

J1000/J5000 Reference Manual

Document Number 10003497.00 Rev.5
Product Code MAN-J1000REF/ MAN-J5000REF
Issued April 2000

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Introduction

About this manual

This manual, together with the:

- J1000/J5000 Installation Manual
- OmniVision User Manual
- OmniVision Help

make up the Jtec J1000/J5000 product documentation.

Refer to *this* manual for configuration information, including:

- definitions of the parameters available for each module and facility
- information about overall system configuration
- a description of, and configuration information for, features such as linesets and Virtual Lines
- information about the product's facilities such as Calling Line Identification and Verification Enhancement (CLIVE)
- a list of the default settings for each module.

The *J1000/J5000 Installation Manual* provides installation, operation, maintenance and troubleshooting information.

The *OmniVision User Manual* and *OmniVision Help* describe the Jtec Network Management System and provide detailed instructions for its use.

Audience

This manual is designed to meet the needs of installation and maintenance personnel and network managers who are responsible for the configuration of the J1000/J5000.

Experience in the use of PCs running Microsoft Windows 95, 98 or NT and communications equipment is assumed, as is a high level of technical knowledge. It is recommended that you attend a Jtec introductory course, including an introduction to OmniVision.

Conventions

Except where otherwise indicated, the information provided in this manual is valid for operation in all countries.

Certain modules and features detailed in this manual are only available to you when your chassis is fitted with a Resource Manager (RM). The following markers are used throughout the manual to indicate the features and modules supported by the Control Module, the Resource Manager, or both:

RM Feature or module supported by the Resource Manager.

CM Feature or module supported by the Control Module.

Using this manual

Each section of this manual provides a different type of information. Refer to the appropriate section for the information you require.

Parameters lists and defines the parameters for each module.

System Configuration contains information about Resource Manager and Control Module default settings and a description of the various features and facilities available. The configuration of these can have implications for the entire system.

Module Configuration lists the default settings for each module as well as providing information about possible changes to the default settings, and giving configuration examples.

Applications provides a number of examples that use the modules and features of this equipment to create various networking solutions.

Overview

J1000/J5000 product family

This manual provides the information you need to configure a J1000/J5000 for the first time or to change an existing configuration. The information applies to all members of the Jtec J1000/J5000 product family. It does not apply to the Terminal Adaptor which is dealt with in the *J1200 Terminal Adaptor User Manual*.

Access Controller (J1000 series only)

Each member of the J1000 series is available as an Access Controller. This performs all the normal Access Controller functions as well as providing the optional facilities. All modules supported by the Control Module can be used in an Access Controller.

Virtual eXchange

Each member of the J1000/J5000 product family is available as a Virtual eXchange (VX). As a Virtual eXchange, the J1000/J5000 can simulate an ISDN exchange and provide basic exchange functions such as call routing, charging and service tones. This enables you to configure a private network of devices using 2 Mbit/s E1 links. You can connect the private network to the public network to allow calls to be made between the two.

All modules can be used in a Virtual eXchange. A VX can also provide all the optional facilities described in Section 3 — *System configuration*.

Voice and data information is processed by the J1000/J5000, including conversion into a digital format if necessary, and integrated onto digital carriers. By this description, the capabilities of the J1000/J5000 are technically defined as those of an intelligent digital multiplexer.

Terminal equipment connects to the J1000/J5000 family using communications cabling and interface protocols, just as if it's connected directly to an exchange network. The equipment provides the interface appropriate to the exchange network you select.

Members of the J1000/J5000 product family can provide a national network access solution for voice and data communications requirements. The equipment can interwork with new and old terminal equipment and exchange networks. Likewise, they can interwork with other network access equipment.

J5000 series

In addition to the features mentioned above, the J5000 also delivers bandwidth efficiency, flexibility and quality of service via Jtec's TimeFrame technology. TimeFrame is the ability to dynamically allocate bandwidth to applications by supporting two modes of trunk operation:

- **Time Mode**
- **TimeFrame Mode.**

These are explained in more detail below.

Time Mode

In Time Mode, both voice and data are carried in separate Time Division Multiplexing (TDM) time slots. Time slots are allocated to applications on a per call basis. The Time Mode is a commonly used access method. Its strength lies in delay-sensitive applications such as voice. This mode is fully compatible with the Jtec J1000 series of products.

TimeFrame Mode

In TimeFrame Mode voice is carried in TDM time slots, ensuring high quality and predictable delivery. These time slots are allocated on a per call basis. Data, management information and signalling are carried in the remaining trunk bandwidth over Frame Relay, taking advantage of the bandwidth efficiency this technology has to offer.

As soon as bandwidth is released by a voice application, it is reclaimed by the data applications. Thus bandwidth is not only dynamically allocated by Frame Relay to data applications, but also between TDM time slots for voice, and Frame Relay for data.

OmniVision

OmniVision is a Microsoft Windows application which runs on a PC. Refer to the *OmniVision User Manual* and *OmniVision Help* for all information about OmniVision operations and features, including screens, parameters, navigation, alarms, polling, logging, status reporting and hardware requirements.

You can establish an OmniVision session and configure:

- a local device, via a direct connection over a serial port or through Ethernet
- a remote device, via a modem
- a remote device, via the ISDN.

This means you can configure a network of J1000/J5000s from one location.

OmniVision enables you to change an existing configuration either online or offline. For example, you can set up a new configuration offline, save it as a database file and then download it at the appropriate time.

The configuration details are stored in both OmniVision files and in non-volatile memory in the Resource Manager or the Control Module.

ISDN concepts

Configuring parameters

The information you enter when configuring device parameters is used differently by different Information Elements (IEs).

Some Information Elements, such as Low Layer Compatibility and High Layer Compatibility, use the information directly, including it in outgoing messages or matching it with IEs in incoming messages. In other circumstances the information is used to make up an IE. For example, the line number is used as part of the calling party IE.

Other configuration information is used to control the device's actions and has no relevance to IEs.

Information Element (IE)

Information about the type of call being made and its destination is conveyed in messages to the network exchange by the user equipment. These messages control the call in its establishment, active and clearing stages. Within each type of message there are Information Elements, some mandatory and others optional. This varies depending upon the type of message and the protocol implemented by the particular country's carriers (refer to your carrier's ISDN protocol specifications).

Information Elements comprise fields which are named according to their function and are constructed of information in octets (8 bit characters) with variable length fields (a minimum of one bit, up to multiple octets, as shown below).

		8	7	6	5	4	3	2	1
Octet		*	*	*	*	*	*	*	*
Binary Values	1 or 0								
Field	one bit								1
	five bits			1	0	1	0	1	
	one octet	0	0	0	0	0	0	1	0
	two octets	1	0	0	1	1	1	0	0
		0	1	0	1	1	0	0	0

Some Information Elements (IEs) are not interpreted by the ISDN, but are passed through to the destination transparently. Others are examined by the network for compliance with protocol, routing, feature invocation and network facility criteria.

Some of these latter IEs may be altered by the network before being presented to the destination ISDN user. Others are interpreted by the network and not passed on. Therefore, IEs may be considered to be of local significance, end-to-end significance or no significance, with respect to the way in which the network uses them.

Jtec's implementation of the ISDN protocols aims to set, as defaults, as many of the IE field values as possible. This minimises the configuration task. This manual identifies possible variations as well as those settings which are necessary to customise the J1000/J5000 to your applications.

The ISDN protocol is only used by the ISDN network's end users. Where the ISDN interfaces with another type of network, for example PSTN or switched packet data, the ISDN provides a protocol conversion between the two networks and allows only suitable types of calls to be routed to them.

Parameters

Introduction

This section lists and defines the parameters available to you when configuring a device. Refer to this section for a description of the parameters for all modules.

Refer to Section 3 — *System Configuration* for the default values for the Control Module and Resource Manager as well as other configuration information for features and facilities that can affect the entire system.

Refer to Section 4 — *Module Configuration* for the default values for each parameter for the modules, and configuration examples.

Definitions

Base Number

ISDN terminal equipment modules are connected to ISDN exchanges by physical bearers. They are identified and addressed by the exchange using a numbering scheme in which each bearer has a unique number, or range of numbers, allocated to it. The number, or the lowest number in the range, allocated to the bearer is the Base Number.

The following examples show the effect on incoming and outgoing calls where the base number is and is not defined.

Incoming call — base number not defined

Number	
Called Party	450 7008
Base	0
Line	450 7008

Note *The Called Party Number minus the Base Number equals the Line Number.*

Incoming call — base number defined

Number	
Called Party	450 7008
Base	450 7000
Line	8

Note *The Called Party Number minus the Base Number equals the Line Number.*

Outgoing call — base number not defined

Number	
Line	450 7008
Base	0
Area Code	2
Calling Party	02 4507008

Note *The Calling Party Number equals the Line Number plus the Base Number and the Area Code.*

Outgoing call — base number defined

Number

Line	8
Base	450 7000
Area Code	2
Calling Party	02 4507008

Notes

- *The Calling Party Number equals the Line Number plus the Base Number and the Area Code.*
- *Sometimes you may be required to enter the Base Number in National Significant Number Format. If so, you must omit the Area Code.*

For further information on Base Numbering see *Appendix H*.

Line configuration

- Line Number** A line number is the unique number associated with a port, an NT channel (time slot) or a Virtual Line. You must enter a line number if the line is to receive incoming calls. However, if the line is a member of either a lineset or a Virtual Line, this is not mandatory.
- Lines can be numbered using OmniVision. Refer to *OmniVision Help* for further information.
- A line number is added to the base number of the incoming Terminal Equipment (TE) module connected to the ISDN when matching the called party Information Element (IE) on incoming calls.
- A line number is added to the base number of the outgoing TE module. It, and the result, is prefixed with the area code to produce the calling party IE on outgoing calls.
- The line number must conform with the number ranges of the ISDN TE modules.
- Subaddress** The subaddress is an additional component of a line or a Hotline number which is transferred end-to-end via the exchange.
- If a subaddress is configured and a called party subaddress Information Element is received in an incoming call, it is matched to allocate the call to a line. If no subaddress is configured, the incoming called party subaddress Information Element is ignored.
- Port ID** The Port Identity (ID) is a number you can assign to uniquely identify a line. The Port ID is not related to the Port Name.
- Port Name** A name can be allocated to a port to make it easier to identify its use or associations. This is particularly useful when creating linesets and Virtual Lines.

ISDN Interface — Bearer Capability

Bearer Capability is a standard ISDN Information Element (IE). It describes to the exchange the type of connection required to make a call. It is a mandatory component in any call setup message and therefore it cannot be disabled.

The setting of a Bearer Capability parameter does not cause the line to be configured according to the setting. The settings simply configure the Information Element. While all parameters can be configured, only a few settings are accepted by a particular exchange.

The parameters you configure do not affect the acceptance of an incoming call to the line, but if an invalid or unimplemented bearer service is specified, outgoing calls may be rejected by the exchange.

The exchange generally delivers the Bearer Capability unaltered to the called party. Some types of user equipment use the Bearer Capability to accept or reject calls. The configuration must therefore satisfy both the exchange and the called user equipment.

Many of the parameters within the Bearer Capability Information Element do not require configuration. These parameters are not identified, described or defined. For further information about these parameters, refer to your carrier's protocol specifications, for example:

- ITU-T (CCITT) Q.931
- Telstra [AOTC] TPH 1856
- Telstra TPH1962
- Telstra TPH 2001
- British Telecom BTNR 190
- British Telecom BTNR 191

The parameters that may require configuration are described below.

Checkboxes

Disable Bearer Checking	When this is enabled, no bearer matching is performed on incoming calls.
Strict Bearer Checking	When this is enabled, a binary match is performed between the Bearer Capability Information Element contained in the incoming Call Establishment Message and the Bearer Capability configured on the line.
Use as LLC if none sent	When this is enabled, a check is made for the presence of a Bearer Capability Information Element in the incoming Call Establishment Message. If none is found, the Low Layer Compatibility Information Element is used to check against the Bearer Capability configured on the line.

Octet 3 Transfer

This information describes the type of information to be transferred.

- Unrest. Digital** Unrestricted Digital. Any information can be transferred. However, an end-to-end digital path is required.
- This information is used by the exchange for service charging. Its provision may result in a different tariff being charged than for analog settings.
- It is also used to enable access to international switched services.
- Speech** Only analog signals can be transferred. If no digital bearers are available, the call may be routed by the carrier via an analog exchange. The exchange provides audible service tones such as ring tone, dial tone, busy tone and Recorded Voice Announcements (RVA).

ISDN Interface — Low Layer Compatibility

Low Layer Compatibility (LLC) provides a means by which a called party can check for compatibility with the calling party. The information is transferred transparently by the exchange to the called user.

Note *The exchange does not check Low Layer Compatibility. It is passed from end-to-end in association with the call setup.*

Low Layer Compatibility settings can be applied to either incoming or outgoing calls or both.

Some modules (usually Digital Line Modules) use parameters in the Low Layer Compatibility Information Element to configure their line characteristics. This is noted in the appropriate line descriptions. Some types of ISDN equipment use the LLC to accept or reject calls. The configuration must therefore comply with the calling line characteristics and also the called party's user equipment.

Apply to

- outgoing calls** This includes LLC settings as an IE in outgoing calls so that the called party can check for compatibility prior to connecting. It is rarely used.
- incoming calls** This compares LLC settings to the IE of incoming calls so that the Digital Line Module can check for compatibility prior to connecting. This is usually enabled.

Note *If LLC is not applied to incoming calls, all calls are treated as if there is no LLC present in their configuration, and they are accepted. If LLC is applied to incoming calls, they are only accepted if LLC is present in their configuration and it matches the configured LLC. If no LLC is present in an incoming call, the call is accepted regardless of whether LLC is applied to incoming calls or not.*

Octet 5 checkbox

Octet 5 describes how data is coded at its lowest level, that is, Layer 1. Enable the checkbox to access the octet 5a - 5d options.

Octet 5 Layer 1

This information describes the method of rate adaption.

CCITT Rate Adaption	ITU-T (CCITT) rate adaption as outlined in recommendations V.110 and X.30. This implies the presence of octet 5a and optionally, octets 5b, 5c and 5d.
A-law	This method is used for speech and audio transmission only. It is the encoding standard used in Europe and Australia.
μ-law	This method is used for speech and audio transmission only. It is the encoding standard used in the U.S.A.
non-CCITT Rate Adaption	This signifies a non-standard method. It is used to convert the user data rate to the 64 kbit/s B-channel rate. This implies the presence of octet 5a and optionally, octets 5b, 5c and 5d.

Octet 5a-d checkbox

This is enabled automatically if the octet 5a - 5d parameters are changed. If the checkbox is not enabled, the parameters are not applied.

Octet 5a-d button

This provides access to the octet 5a - 5d parameters.

Octet 5a checkbox

This provides access to the User Rate and Synchronous lists.

Octet 5a User Rate

Use this to select the user rate of the data.

Inband	Standard 64 kbit/s.
50 - 56000 bit/s	Various data speeds in the range from 50 to 56000 bit/s.

Octet 5b checkbox

This provides access to the Intermediate, NIC Tx and NIC Rx lists. It can only be used if the Octet 5a checkbox has been enabled, and V.110 and X.30 rate adaption are being used.

Octet 5b Intermed list

The intermediate rate is used when adapting the user rate to the bearer rate. This option should be used in conjunction with the user data rate, and must be set to a value greater than or equal to the user rate.

Note *The user rate is set in bit/s and the intermediate rate is set in kbit/s.*

None	Select this option if you selected a user data rate 32 kbit/s or less.
8 kbit/s	8 kilobit/s.
16 kbit/s	16 kilobit/s.
32 kbit/s	32 kilobit/s.

Octet 5c checkbox

This option describes the format of the user data. It is used mainly for asynchronous transmission. It can only be used when Octets 5a and 5b have been enabled.

Octet 5c Stop Bits

This sets the number of stop bits used in a data transmission.

Blank	No stop bits.
1, 1.5, 2	Number of stop bits.

Octet 5c Data Bits

This sets the number of data bits used in a transmission. It should be used in conjunction with the parity bits.

Blank	No number of data bits specified.
7	This is commonly used when a parity bit is to be included.
8	This is the most common number of data bits used in a transmission. If selected, the parity must be set to None .

Note *When V.110 or X.30 rate adaption are used, the number of data bits must be set to 7.*

Octet 5c Parity

A parity bit can be used in a transmission for error detection.

Odd	The parity bit is set so that the total number of bits with value one is odd.
Even	The parity bit is set so that the total number of bits with value one is even.
None	Select this if no parity is used.
Force to 1	The parity bit is set to one.
Force to 0	The parity bit is set to zero.

Octet 5d checkbox

This provides access to the Duplex and Modem parameters. You can only use it when Octets 5a - 5c are enabled.

Octet 5d Duplex

This describes the method of transmission: alternate or simultaneous.

Half	This should be used to indicate two-way, alternate transmission. One end transmits at a time.
Full	This should be used to indicate a two-way simultaneous transmission. If selected, both ends can transmit and receive simultaneously.

ISDN Interface — High Layer Compatibility

The High Layer Compatibility (HLC) Information Element settings provide information which allows the called party to check that the high layer protocols at each end of the call are matched.

For each line, an HLC value is configured for use in outgoing call setup messages as well as up to three values for incoming setup messages. If an HLC value is defined for outgoing calls from a given line it is always sent, otherwise no HLC Information Element is sent.

Note *The exchange does not check High Layer Compatibility. It is passed from end-to-end in association with the call setup.*

Some types of ISDN equipment use the HLC to accept or reject calls. The configuration must, therefore, comply with the calling line characteristics and also the called user equipment.

For all of the following parameters, refer to your exchange's ISDN protocol specifications.

Standards checkboxes

The Telecom Standards checkbox allows you to set the Telstra Australia National Standard definitions.

The CCITT checkbox allows you to set ITU-T (CCITT) defined values for HLC.

Call Direction checkboxes

This option sets the call direction to which the HLC settings apply.

Incoming This compares HLC settings to the IE of incoming calls so that the called party can be checked for compatibility prior to connecting. You can set up to three Incoming HLCs. Any incoming call with an HLC IE present will be checked against the HLC options configured for the called party.

Note *Incoming calls are always checked for HLC. If HLC is present in the configuration of an incoming call, it is checked for verification against the options selected. If no options are selected, or the options do not match, the call is rejected. If no HLC is present in the configuration of an incoming call, it is always accepted.*

Outgoing This includes HLC settings as an Information Element (IE) in outgoing calls so that the called party can check for compatibility prior to connecting. You can set one Outgoing HLC.

ISDN Interface — Call Establishment Modes

The available Call Establishment Modes and their functions vary according to the type of line. They may also depend upon the type of equipment connected.

Some modules have line types that require special Call Establishment Modes not relevant to other line types. The following description covers both the common Call Establishment Modes that apply to all line types and those specific to certain types of lines.

Common Call Establishment Modes

Semipermanent (where available)

A semipermanent connection (SPC) is the equivalent of a tie line or leased line between two parties through the ISDN. It is set up by the ISDN provider but it can be de-activated to free the B-channel for other traffic. Refer to the *OmniVision User Manual* or *OmniVision Help* for further information.

Enable Enables an SPC.

Label This is the semipermanent circuit identifier supplied by the ISDN provider. It can consist of up to nine alphanumeric characters and it must be present for the semipermanent to connect. For semipermanent connections the label performs a function similar to the Line Number.

In some Virtual Line configurations you may need to specify a dummy semipermanent label to enable the semipermanent configuration.

Note *The group number of the Incoming ISDN terminal equipment module must also be configured for the semipermanent connection to proceed.*

Hotline

A Hotline number enables rapid connection of the line to a called party because the called party number is dialled immediately a call establishment request signal is received from the terminal equipment. A Hotline call can be initiated:

- by a line sequence, such as *off-hook*
- manually through OmniVision
- via the SPC Backup call establishment (where available).

A partial Hotline can also be entered to enable pre-dialling.

Note *A Hotline is only relevant to outgoing calls, not calls received from the ISDN.*

Enable	Enables the Hotline feature.
Number	Enter the number to be dialled. It can be up to 20 digits.
XSPC	This enables a Switched Semipermanent Connection (XSPC). An XSPC automatically attempts to establish a Hotline switched call when it is configured, and when an existing XSPC clears. If an attempt to establish a call fails, it is retried at timed intervals, until it is successful or the XSPC is disabled. The XSPC effectively creates a leased line, or tie line across the ISDN, using switched calls.

Note *For DLMs, calls are retried at one second intervals. For ALEMs, ALEM-2s, ALPMs, ALPM-2s, EMMs, EMM-2s and E1Ms, calls are retried within a three to seven second period from when they were cleared.*

Subaddress	Sets the called party Subaddress Information Element in the Hotline call. Refer to the appropriate ISDN protocol specifications for the characteristics of this parameter.
Auto Answer	A connection is made immediately an incoming call is received, without the intervention of connected equipment. A call charge is incurred even if the equipment connected to the called party is busy or unavailable. This option is available for all lines.

Line type dependent Call Establishment Modes

Semipermanent (where available)

(DLMs only)

Backup	This option sets up a normal circuit-switched connection to the Hotline number if the SPC fails. An SPC failure can only be detected by DLMs when using CCITT (ITU-T) V.110 or X.30 rate adaption at 48 kbit/s or lower user data rates. If you select this option, you must enter and enable a Hotline number at the end where the SPC Backup is set. Auto Answer must be enabled at the other end.
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DDI (Indial)

(E1M, ALPM, ALPM-2, EMM, EMM-2)

Enable	Activates DDI. This is used to pass part or all of the called party number Information Element received with an incoming call, to connected terminal equipment for direct dial-in purposes. A line with DDI enabled is usually part of a lineset with a range of numbers.
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Ignore first... digits

(E1M, ALPM, ALPM-2, EMM, EMM-2)

This option allows you to specify the number of digits received from the exchange to be ignored before direct dialling to the connected terminal equipment begins.

For example, if you specify that the first three digits of the number 450 7100 are ignored, 7100 will be dialled. You can specify the maximum number of digits in the Called Party Number IE to be ignored.

Active Call Facilities

(ALPM and ALPM-2 only)

Release call if Busy tone detected This option is only used in tie line applications. It specifies the action to be taken when a PABX line is busy. If enabled, and the line is busy, the call is released.

Use all cadence values This option enables the use of tone detection for all specified cadence values. If it is not enabled only the first two cadence values are used.

(DLMs and COMBO)

Don't release call if V110 frame loss detected This option prevents the call being released if V.110 frame loss associated with CCITT (ITU-T) rate adaption is detected.

Control Leads

(DLMs and COMBO)

This allows you to select the method by which the state of the control leads is transmitted when rate adaption is used.

Local Only The state of the control leads is not transmitted inband.

Transferred Inband Automatically transmits indication (inband) of the real state of the control leads. This is available for DLMs with data rates of 19.2 kbit/s or less.

Signalling

(ALEM, ALEM-2, ALPM, ALPM-2, COMBO ALIM, EMM, EMM-2, E1M)

This selects the method of signalling.

National Significant Remote Number Enter the national significant group number of the remote device. This is only needed when the line is a member of a semipermanent Virtual Line and the semipermanent call is transit-switched to its destination via one or more intermediate devices.

End-to-End Out-of-band signals are transmitted from one end of a connection to the other end. This option is normally used for tie-line applications such as:

- XSPCs
- SPCs (where available).

(EMM, EMM-2 only)

Interpreted Selects the type of proprietary signalling from an EMM. If selected, it enables communications with all analog modules using a standard internal protocol. If not selected, communication is only possible between EMMs using an uninterpreted protocol. This is useful when the EMM cannot recognise the signalling protocol that it is required to transfer.

(EMM, EMM-2 only)

Unacknowledged This option prevents proprietary signalling from implementing a message acknowledgement, thus reducing the amount of signalling. Available for EMMs in uninterpreted mode only.

(E1M only)

E2E answer xfer End-to-end answer transfer. This option provides an end-to-end answering mechanism to support some inter-PABX operations. It provides for the answering of an ISDN switched call followed by the end-to-end transfer of an Answer signal from the called terminal equipment back to the calling line.

Call Control

Describes the way in which calls are sent and received.

(SDLM V.24, ADLM V.24 QDLM and COMBO)

DTR Call (V.24) Calls are controlled by the DTR control lead on V.24 DLMS. If this option is selected a Hotline number is also required.

(ADLM V.24 and QDLM)

AT This option enables AT modem emulation on an ADLM V.24 and a QDLM V.24 operating in asynchronous mode. Calls are setup and cleared using AT commands. The V24 DLMS do not actually become modems because they still implement V.110 rate adaption on an unrestricted digital bearer.

(SDLM X.21, QDLM and COMBO)

X.21 This option enables an X.21 interface between the terminal equipment and the SDLM X.21 for synchronous operation on public data exchanges.

C Signal (X.21) This option enables and disables X.21 call control protocol.

(SDLM V.35 only)

RS (V.35) This option is defined as ITU-T (CCITT) Interchange Circuit No 105. Calls are controlled by the RS/RTS control lead on V.35 lines. A Hotline number is also required if this option is selected.

(QDLM and COMBO)

V.25 bis This option enables Cisco V.25 bis (software release V9.1) compatible line signalling capability when operating over X.21, V.35 and V.24 synchronous line interfaces. It supports both V.25 bis signalling and online data interchange at 64 kbit/s user rate only.

(E1M only)

Connect after indialling digits Similar to Auto Answer except a connection is made after the digits have been dialled. This is used in conjunction with DDI (Indial). It enables call progress tones, such as ringing or busy, to be passed from the terminal equipment (usually a PABX) to the calling party.

PABX Recall

(ALEM, ALEM-2 and COMBO ALIM)

This option enables a timed loop break generated by the connected equipment to be detected and signalled to the other end of the tie line for forwarding to the called terminal equipment. It is only used with tie line applications and requires an ALPM or ALPM-2 at the called end to regenerate the timed loop break.

Enable Enables the timed loop break detection for PABX recall facilities.

Back Busy

(EMM, EMM-2 only)

Provides a busy indication to the connected equipment if the tie line fails:

- XSPC
- SPC

Enable Enables back busy.

If channel is Subrate it's (E1M only)

This option allows you to specify whether a Subrate Virtual Line should apply a compander or a Subrate switch to the information transmitted from or received by the line.

- Voice** For a line that transmits analog signals. A compander for either a VCM or LDCM is assigned.
- Data** For a line that transmits data. It assigns a Subrate switch.

RM TimeFrame Trunk Name (TimeFrame Virtual Line only)

Specifies the trunk name associated with the Packet Channel of a TimeFrame VL.

See *Configuring the TimeFrame VL packet channel* on page 3-58.

RM TimeFrame minimum data rate (TimeFrame Virtual Line only)

Specifies the minimum portion of the TimeFrame VL bandwidth allocated to data. That is, voice is not inserted into this bandwidth.

RM TimeFrame inactivity timeout (TimeFrame Virtual Line only)

Specifies the time of no data activity on the TimeFrame VL after which the call is dropped.

ISDN Interface — ISDN Line Controls

This option allows you to define the numbering schemes and supplementary services to be applied to a selected line.

Line Numbering

The selected line numbering scheme is applied to the line. For outgoing calls it is used to set a parameter in the Calling Party IE. The numbering scheme supplied by the calling party is not checked for incoming calls.

The line numbering scheme values you set are used by the ISDN to determine if the correct line numbering scheme is present for access to itself and other exchanges. You would not usually change them. Refer to your exchange's protocol specifications for further information.

AOC (where available)

Note *The following supplementary service has been defined by Telstra for use in Australia.*

Advice of Charge (AOC) is a supplementary service available from your ISDN provider.

Switched call charge information provided by the ISDN is stored in the Call Logging application within OmniVision and is not sent to any line modules (except IPMN and BRMN in which case it is transparent and no configuration is required).

If you subscribe to it the following options are available.

None	No AOC information received.
Continuous	Incremental costs are received during the call.
End of Call	The total cost is received at the end of the call only.



Don't Increment Hotline No. (TimeFrame Virtual Line only)

This specifies that the Hotline Number should not be automatically incremented. The RM normally increments each of the multiple lines when creating a TimeFrame Virtual Line with a bandwidth greater than 64 kbit/s. If this is enabled, you must manually set the Hotline Numbers for each line.

Outgoing CLI

Calling Line Identification Presentation (CLIP) and Calling Line Identification Restriction (CLIR) are supplementary services available from your ISDN provider. If you subscribe to them the following options are available.

Note *The exchange rejects a call if an inappropriate CLI option is selected.*

CLIP This is the default. Do not change it.

Note *ISDN service providers in Australia do not currently support customer selectable CLIP options.*

CLIR If you subscribe to a permit transfer CLIP option, and enable this, the exchange withholds your number from the called party when you make a call.

Note *This option is a service provider facility. It is not related to the Calling Line Identification and Verification Enhancement (CLIVE) which is described in the System Configuration section of this manual.*

ISDN Interface — Teleservice (where available)

Teleservice is a telecommunications service definition which indicates the application or service being used.

Note *Teleservices are defined by Telstra to enable charging options. Currently only Timelink is used. The Bearer Capability settings are also used for charging purposes.*

Teleservice settings are not matched on incoming calls and have no end-to-end relevance. High Layer Compatibility settings should always be used for end-to-end compatibility definition.

None Normal teleservice setting.

Note *Other Teleservice types exist, but we recommend that you only use them to enable calls to be made to other ISDN equipment that requires them to be present.*

- Timelink** This is a switched call tariff option provided by Telstra Australia on subscription. It has a high flag fall charge with small increments and is mainly used for data calls of medium duration. The tariffs charged depend upon the distance of the call.
- This service means that data calls may become cheaper after approximately two hours when compared with switched calls that are not using Timelink teleservice.

ISDN Internal Control — Restriction

- This option allows you to restrict the calls sent to and received from a selected line.
- None** No restrictions. Receives and makes calls.
- In only** Receives calls only (Outgoing calls are barred).
- Out only** Makes calls only (Incoming calls are not accepted).
- Disable** No calls made or received.
This option can also be used to set manual back busy or blocking for tie line applications on EMMs and E1Ms.

ISDN Internal Control — Dial Type

- This option allows you to select the type of dialling to be used by a device which is connected to an ALEM, ALEM-2, ALPM, ALPM-2 or COMBO ALIM port.
- Pulse** This is used for rotary dial telephone sets.
- DTMF** Dual Tone Multifrequency (DTMF) dialling is the basis for operation of pushbutton telephone sets.
- Both** Enables Pulse or DTMF signalling depending on which is detected first for an outgoing call (ALEM and ALEM-2 only).
- None** Selects no dialling.

ISDN Internal Control — Codec

A codec is a coding/decoding device. The coder is used to convert analog signals to a digital format for transmission over the ISDN exchange. The decoder is used to convert the digital signals back to the original analog signals.

This option allows you to set the relative level and hybrid balance of the codec. It is applicable to the ALEM, ALEM-2, ALPM, ALPM-2, COMBO ALIM, EMM and EMM-2.

Relative level (dBr)

(ALPM, ALPM-2, ALEM, ALEM-2, COMBO ALIM, EMM and EMM-2)

This sets the relative analog signal level or volume of the incoming and outgoing calls.

Note *The dBr reference point is the ISDN.*

(ALEM, ALEM-2, ALPM, ALPM-2, COMBO ALIM, EMM and EMM-2)

- Send Level Out Of This Card** This sets the relative analog signal level being sent from this module.

(ALEM, ALEM-2, ALPM, ALPM-2, COMBO ALIM, EMM and EMM-2)

Receive Level Into This Card This sets the relative analog signal level received into this module.

Hybrid Balance

The hybrid balance settings adjust the line length and impedance values to match the ISDN module to your terminal equipment.

Note *These settings are not usually changed. They vary depending upon the terminal equipment used and should only be changed when used in conjunction with analog transmission measurement equipment.*

ISDN Internal Control — PABX

This option is used in association with tie line applications. It allows you to set the maximum duration of a timed loop break (or hookflash) before the ALEM, ALEM-2 or COMBO-ALIM terminates the call. For example, if you set the recall time to one second, a signal indicating a hookflash is sent to the other end of the tie line if the telephone receiver is on hook for less than one second. If the receiver is on hook for more than one second a signal indicating that the call is to be cleared is sent.

Max PABX Recall Time

Short	250 ms.
Long	750 ms.
Other	User defined, but not greater than 1280 ms.

Note *The maximum recall time allowed is 1280 ms. It is set in stepped values up to 1280 msec. This parameter should be set in conjunction with information provided by your PABX service personnel.*

ISDN Internal Control — Service Tones

This option allows you to define the cadence of the ring and busy tones detected by an ALPM or ALPM-2 line entity and the cadence of the ring voltage generated by an ALEM or ALEM-2 line entity.

Ring	Defines the cadence of the ring tone. This enables you to set or edit the ring cadence. Select the top of one of the bars to edit the current sound period or the baseline between bars to edit the current silence period.
Busy	Defines the length of the busy tone. This enables you to set or edit the busy cadence. Select the top of one of the bars to edit the current sound period or the baseline between bars to edit the current silence period.
Period	Defines the length of the on and off cycles for the ring/busy tone in integer multiples of 100 ms.

Note *A typical Telco ring cadence is 4, 2, 4, 2. Busy is 4, 4, 4, 4 (actually 375 ms). The tolerance for both is +/- 30%.*

ISDN Internal Control — Outgoing TE

- This option allows you to select the TE module (the connection to the ISDN) through which outgoing calls are to be routed when there are multiple TE modules in the chassis. (The currently available TE modules are the IPMT and BRMT.)
- SPID (USA only)** The Service Profile Identifier is a unique number, from nine to twenty digits long, that is used by the BRI NI1 protocol to identify a set of subscribed capabilities.
- Any** This is the default. Calls are sent to any available TE module. This means that outgoing calls are distributed across all TE modules. They are placed on the TE module with the most free B-channels.
- Preferred** Calls are sent to the preferred TE module. If the module is unavailable the call is sent to the TE module with the most free B-channels.
- Exclusive** Calls are sent exclusively to the selected TE module. If all B-channels of the nominated TE module are busy or the TE module or line has failed, the call fails.
- Use TE in slot** If **Preferred** or **Exclusive** is selected you must define a TE module by specifying its card slot within the device.

System Configuration

Introduction

This section provides details of the overall system configuration and the operation of optional system facilities which may have implications for the entire system. As well as the default settings for the Control Module, Resource Manager and the Virtual eXchange (VX), it contains a description of, and configuration information for, the following features common to all products:

- Linesets
- Virtual Lines
- ISDN Interworking
- MicropleX Switching
- Inband Signalling
- Calling Line Identification and Verification Enhancement (CLIVE)
- Serial Alarm Interface Options
- Phone Home.

This section also provides information on the following features that are available only to chassis fitted with the Resource Manager:

- Wideband
- Flash management
- Packet Switching
- Bridge
- IP Router
- HDLC handling
- Frame Relay PVC management
- TimeFrame

The parameters for Control Module configuration are shown on page 3-3.
The parameters for Resource Manager configuration are shown on page 3-5.
TimeFrame configuration is covered on page 3-19. For information about, and configuration instructions for, linesets refer to page 3-46.

Refer to page 3-50 for information about Virtual Lines and instructions for their creation.

Refer to page 3-57 for information about ISDN Interworking.

Refer to page 3-67 for details of MicropleX Switching.

Refer to page 3-74 for information on Inband Signalling.

Refer to page 3-80 for an explanation of, and configuration information for, CLIVE.

Refer to page 3-86 for information on Serial Alarm Interface.

Refer to the *OmniVision User Manual* and *OmniVision Help* for step-by-step instructions on how to use OmniVision.

Control Module (CM)

The Control Module (CM) controls the switching of all data through the system and, therefore, must be present in any device. It has a non-volatile memory to store configuration information for itself and all other line modules present in the subrack. It also maintains a log of all calls made and received, changes in the status of modules and any errors detected.

Note *If the CM is moved to a different slot in the chassis, the configuration details are deleted from that slot when the CM is restarted. The CM will require reconfiguring in its new slot position.*

Configuration

Manager Access

Number If a Manager number is defined, it is used in the same way as a line number.

In Australia, if a number is not defined, all incoming B-channel and D-channel calls which have the correct password are used to establish an NMS session. This allows for the remote configuration of a newly installed device.

In the UK, if a number is not defined, all incoming calls are directed to the Control Module Manager access. This allows for the remote configuration of a newly installed device.

If you are using a Basic Rate connection with Telstra Australia's OnRamp Xpress service, certain limitations apply. Refer to *Appendix S* for details.

The Manager number is often, by convention, the penultimate number in a block. For example, in a block of 100 lines (0-99) it would be 98. However it can be any number within the range allocated to the BRA or PRA.

Note *You should take care when configuring the Manager number from a remote site. If you choose a number outside the range for the PRA and BRA, further remote access will not be possible.*

In Australia, the Manager number is mandatory. End-to-end signalling also uses D-channel calls to transfer information via the ISDN. If a Manager number is configured the Control Module searches other modules to find an end point for the call.

In the UK, the Manager number is mandatory and should be configured first.

Check Incoming CLI This enables checking of incoming remote configuration calls by the Calling Line Identification and Verification Enhancement (CLIVE). Refer to page 3-80 for further information about, and configuration instructions for, CLIVE.

Test Line

Number	<p>This number is used to assist in the diagnosis of equipment and signalling faults. It is not mandatory.</p> <p>When the test line number is dialled a transmit-receive loopback is established to the calling party. The number used is often, by convention, the last number in a block. For example, in a block of 100 lines it would be 99.</p> <p>The number causes all incoming calls (with this number as their called party number) to be automatically answered regardless of all other call control criteria.</p> <p>In Australia, if the incoming call is a D-channel packet call (Demand Transfer), user to user packets are also looped back to the user.</p> <p>Up to 32 simultaneous calls can be answered. This number should be uniquely allocated to the Control Module and no other module.</p>
Subaddress	<p>The subaddress is an additional component of the above number which is transferred end-to-end via the network. If it is used, it frees the available range of numbers for in-dial functions. It is not mandatory.</p>

International inband signalling

This will normally be required when connecting between a node in the U.S.A. and a node in another country. All devices in the network need this enabled. Its not valid to have one end enabled and the other end disabled. This option will also work locally and nationally.

National escape prefix

This should be set to **1** for the U.S.A. and **0** for all other countries.

Line Number Matching

Some authorities are changing network numbers from seven to eight digits. There is a transition period when callers can use either the seven or eight digit number to reach the destination line. Use this facility to enable the device to match either of the numbers dialled by the caller.

The eighth digit is suffixed by the existing seven digit number, for example, the number 3640091 may become 93640091. The matching is achieved by ignoring a specified number of digits from the front of either the called number, the line number or lineset range numbers. Digits are ignored only if the difference between the length of the called number and the line number or lineset range numbers is equal to the number of digits to ignore. When the low number and high number of a lineset number range have different lengths, the called number length is compared with the longer of the range numbers.

For the changeover, set 'Ignore N digits when matching numbers' to 1.

Resource Manager (RM)

The Resource Manager (RM) controls the switching of all the calls and data through the system and, therefore, must be present in any chassis. It has a non-volatile memory to store configuration information for itself and all other line modules present in the subrack. It also maintains a log of all calls made and received, changes in the status of modules and any errors detected.

Note *If the RM is moved to a different slot in the chassis, the configuration details are deleted from that slot when the RM is restarted. The RM will require reconfiguring in its new slot position.*

The Resource Manager also has an internal IP Router and Bridge capable of routing/bridging data between the RM's Ethernet port and TimeFrame trunks. See *Appendix L* for information on the protocol conformance of the RM.

An integral Frame Relay Packet Switch allows switching of Packet information contained in the TimeFrame trunks.

Additional support for HDLC-based protocols such as X.25, Frame Relay and SNA is provided by the on-board HDLC Data Handler.

Flash memory allows configuration information and software images to be stored for all the system modules, including the RM.

Note *When updating the RM software from Network Release B and lower, to Network Release C and above, the new software automatically carries out a conversion process on the configured TimeFrame settings. The conversion allows TimeFrame configured in earlier RM software to operate in the Wideband TimeFrame mode. Wideband TimeFrame operates on trunks greater than 64 kbit/s.*

*Further information on the RM configuration conversion can be found in the **J1000/J5000 Upgrade User Note**.*

Configuration

Manager Access

Number If a Manager number is defined, it is used in the same way as a line number.

Note *If you are using an IP connection from OmniVision, ensure that the **Access Number** entered for **Node Access** in OmniVision is an exact match of the **Manager Access Number**. This should not include area code prefixes. Alternatively, set the **Access Number** to **0** (zero).*

Where D-channel calls are available

If a number is not defined, all incoming B-channel and D-channel calls which have the correct password are used to establish an NMS session. This allows for the remote configuration of a newly installed device.

Where D-channel calls are not available

If a number is not defined, all incoming calls are directed to the Resource Manager access number. This allows for the remote configuration of a newly installed device.

The Manager number is often, by convention, the penultimate number in a block. For example, in a block of 100 lines (0-99) it would be 98. However it can be any number within the range allocated to the BRA or PRA.

Note *You should take care when configuring the Manager number from a remote site. If you choose a number outside the range for the PRA and BRA, further remote access will not be possible.*

In the UK, the Manager number is mandatory and should be configured first.

If you are using a Basic Rate connection with Telstra Australia's OnRamp Xpress service, certain limitations apply. Refer to *Appendix S* for details.

In some markets, end-to-end signalling also uses D-channel calls to transfer information via the ISDN. If a Manager number is configured, the Resource Manager searches other modules to find an end point for the call.

Management IP Address and Subnet Mask

The Management IP address is used by the NMS to access the configuration and management aspects of the RM related to its data processing units. These units may be the IP Router, Bridge, HDLC data handler, Packet Switch and so on.

The entry must be a valid IP address that is accessible via the LAN to which the RM is connected. An IP address takes the form of a dotted decimal notation, for example, 192.195.112.42

The Subnet Mask also conforms to the same decimal notation.

Note *The RM does not support an IP address which contains a .0 (specified in RFC950). For example, the address 172.16.0.80 with subnet mask 255.255.255.0 is not supported.*

Check Incoming CLI

This enables checking of incoming remote configuration calls by the Calling Line Identification and Verification Enhancement (CLIVE). Refer to page 3-80 for further information about, and configuration instructions for, CLIVE.

Test Line**Number**

This number is used to assist in the diagnosis of equipment and signalling faults. It is not mandatory.

When the test line number is dialled a transmit-receive loopback is established to the calling party. The number used is often, by convention, the last number in a block. For example, in a block of 100 lines it would be 99.

The number causes all incoming calls (with this number as their called party number) to be automatically answered regardless of all other call control criteria.

Note *If the incoming call is a D-channel packet call (Demand Transfer), user to user packets are also looped back to the user.*

Up to 32 simultaneous calls can be answered. This number should be uniquely allocated to the Resource Manager and no other module.

Subaddress The subaddress is an additional component of the above number which is transferred end-to-end via the network. If it is used, it frees the available range of numbers for in-dial functions. It is not mandatory.

International inband signalling

This will normally be required when connecting between a node in the U.S.A. and a node in another country. All devices in the network need this enabled. Its not valid to have one end enabled and the other end disabled. This option will also work locally and nationally.

Setting this option also allows you to apply a Number Checking List to the inband signalling router. The Called Party Number can be modified to facilitate inband signalling calls across international boundaries.

National escape prefix

This should be set to **1** for the U.S.A. and **0** for all other countries.

PCM Coding

Set this to A-law or μ -law to set the default PCM coding method that is to be used throughout the device.

Line Number Matching

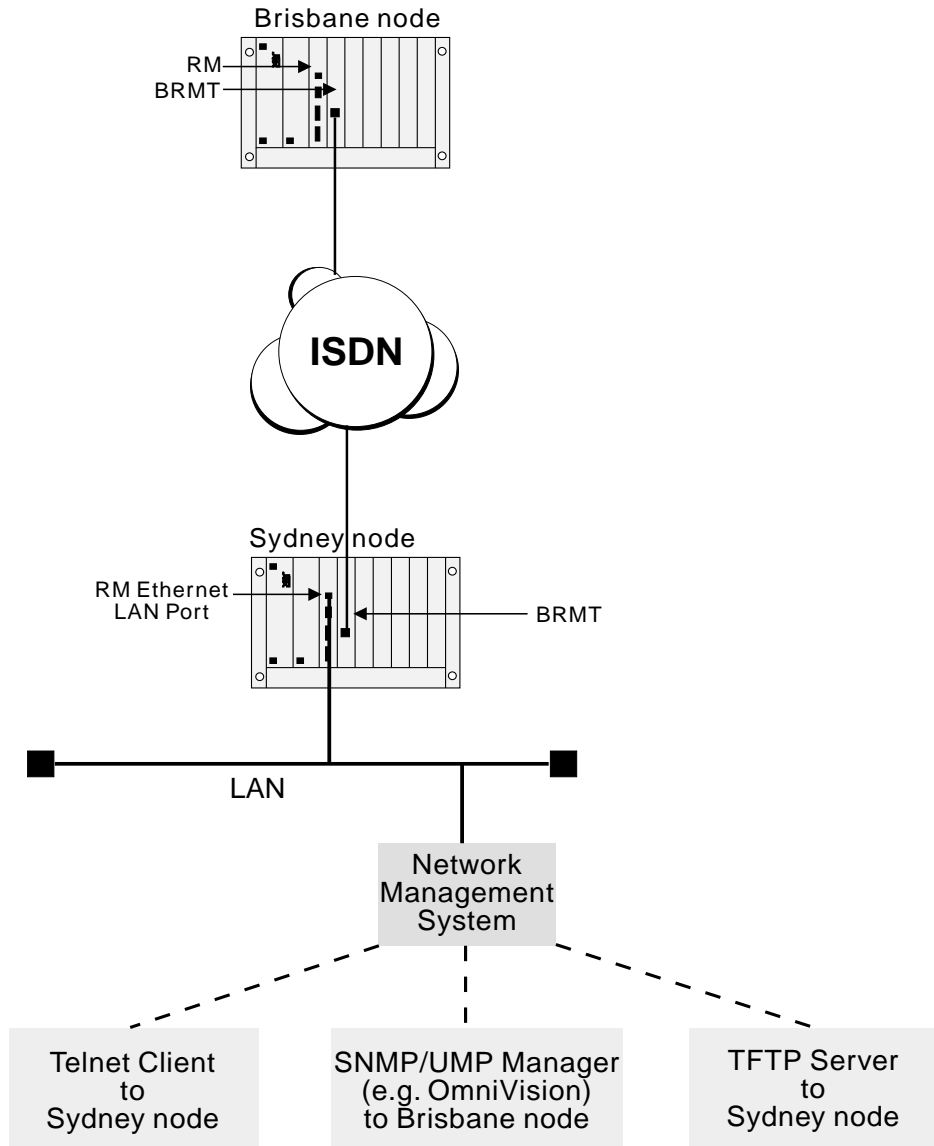
Some authorities are changing network numbers from seven to eight digits. There is a transition period when callers can use either the seven or eight digit number to reach the destination line. Use this facility to enable the device to match either of the numbers dialled by the caller.

The eighth digit is suffixed by the existing seven digit number, for example, the number 3640091 may become 93640091. The matching is achieved by ignoring a specified number of digits from the front of either the called number, the line number or lineset range numbers. Digits are ignored only if the difference between the length of the called number and the line number or lineset range numbers is equal to the number of digits to ignore. When the low number and high number of a lineset number range have different lengths, the called number length is compared with the longer of the range numbers.

For the changeover, set **Ignore N digits when matching numbers** to 1.

Management

The Resource Manager supports local management via the RM Ethernet LAN port and Utilities (DB9) port. It also supports remote management using IP over Frame Relay within a TimeFrame Virtual Line, and UMP (User Management Protocol) over a D- or B-channel. Up to 30 simultaneous OmniVision IP sessions can be active. Standard management protocols such as SNMP (Simple Network Management Protocol), TFTP (Trivial File Transfer Protocol) and Telnet can be used for management along with Jtec's proprietary UMP. A diagram of a typical management scenario is shown below.



Configuring the Management IP Address

To use IP-based management, either via the Ethernet LAN port or remotely, a Management IP Address must be associated with each RM. The Management IP Address is the same IP address that is given to the Ethernet LAN port on the RM. The Management IP Address and hence the LAN Port IP Address can be set in the following ways:

- Using Bootp
- Using RM Module configuration.

Setting Management IP Addresses using Bootp

The RM supports the Bootstrap Protocol (Bootp). The RM uses this to determine the IP Address and Subnet Mask associated with the Ethernet LAN Port. To use this feature a Bootp Server must be installed on the LAN. This must be capable of accepting the Bootp requests from the RM and returning a suitable reply containing the IP address and subnet mask.

In order to use Bootp, the RM Ethernet MAC Address must be associated with an IP Address and Subnet Mask in the Bootp Server's database file. When the Bootp Server receives the Bootp request, it matches the received MAC Address with the one in its database and returns a Bootp reply with the IP address and subnet mask to the RM.

Note *The Ethernet MAC Address of the RM is unique to each board. A label containing the address is located on the back of the board (side without components). It is 12 digits long and starts with the digits 0020BC.*

OmniVision provides a Bootp Server that the RM can use to obtain its IP address. Refer to the *OmniVision User Manual* for details on using the Bootp Server.

This is typically used for an unconfigured locally-managed device to determine the initial IP address associated with the Ethernet LAN port. If a Management IP Address is set in the RM Module Configuration then no Bootp request will be sent by the RM.

The RM does not store the IP Address, obtained from the Bootp server, in non-volatile configuration. Each time the RM is restarted the RM requests an IP Address. To store the IP Address in non-volatile configuration, enter it in the RM Module Configuration dialog.

Setting the LAN IP Address using the RM Module configuration

The RM Module Configuration dialog contains a setting for the Management IP Address and the Subnet Mask. These values are used for the Ethernet LAN port on the RM. They are entered as dotted decimal notation. See page 3-6 for more details.

This is typically used for configuring the Management IP Address for an unconfigured remote node, as well as setting a permanent IP address to avoid using Bootp for a locally managed device. This method cannot be used on an unconfigured device managed locally via the Ethernet LAN port. In this case you must use the Bootp configuration method.

Telnet

The RM has an on-board Telnet Server that allows either a remote Telnet Client or the OmniVision Virtual Terminal to access configuration and status information.

To connect to the RM Telnet Server using a Telnet client on a PC or Workstation, a Management IP Address must be configured. For more information, see *Configuring the Management IP Address* on page 3-9. A Telnet Client should then connect to the Management IP Address of the RM.

To connect to the RM Telnet Server using the OmniVision Virtual Terminal, start an OmniVision management session to the RM. Select the RM with right mouse button, and choose **Virtual Terminal**.

A default logon of username 'admin' and no password exists for the first time connection. To get help on the list of commands supported by the RM Telnet Server enter ? at the `rm:telnet>` prompt and press enter.

Note *A username and password is not required when accessing the RM Telnet Server via the OmniVision Virtual Terminal.*

The RM Telnet Server provides a range of commands for managing the Flash File System, Packet Switching, IP Routing and Bridge of the Resource Manager. See the appropriate section for details on how those commands are used.

To exit the Telnet Server, type `quit`, `exit` or `logout` at the `rm:telnet>` prompt.

A complete set of Telnet commands is given in *Appendix M*.

Flash management

The RM has 4 Mbytes of onboard Flash Memory that is used for keeping non-volatile information such as the RM configuration information, and the software image that is executed by the RM.

The Flash Memory is set up as a file system that can contain up to 3.84 Mbytes of data arranged as files. The configuration information and software images are each stored as files. These files can then be exchanged between the RM and a Network Management System using TFTP (Trivial File Transfer Protocol).

Alternatively, you can use the OmniVision Flash File Manager to examine the contents of the RM Flash memory and transfer files via TFTP. See the *OmniVision User Manual* for details.

You can examine the contents of the RM Flash File System by establishing a session to the RM's Telnet Server. After logging on under Telnet, a list of commands is available for manipulating files, including renaming a file, listing available files, deleting a file and typing a file.

The following commands can be used to manage files contained in the RM Flash File System. These commands can be entered at the `rm:telnet>` prompt:

- `ls` list files in the Flash File System
- `dir` as for `ls`
- `cat [filename]` type a file onto the Telnet screen
- `type [filename]` as for `cat`
- `mv [currentfilename][newfilename]` rename current file to new filename
- `ren [currentfilename][newfilename]` as for `mv`
- `del [filename]` delete a file from flash
- `rm [filename]` as for `del`.

All of the above file commands operate on files within the Flash File System. For information on copying files into and out of the Flash File System refer to *TFTP* on page 3-13.

Selecting the RM software image to run

The RM is capable of storing multiple versions of RM software images, limited only by the amount of available space in the Flash Memory. Each version of RM software is distinguished by a unique version number stored within the software image file header. On startup, the RM Image Loader executes the latest version of the software image file in the file system.

It is also possible to select an image to run in the case where multiple versions of software are maintained on the system. To do this:

- 1 Connect an RM Telnet Server session to the RM. A complete set of Telnet commands is shown in *Appendix M*.
- 2 Enter `run` with no parameters to see which image is currently active.
- 3 Enter `run <filename>` to select a specific software image file in Flash to execute. `<filename>` must be a valid filename that exists in the RM flash file.

If this file exists on startup it will be run regardless of what other versions of RM software images are present in the system. If it doesn't exist, the RM Image Loader finds the latest version available.

Note After entering the name of the file you should wait for 15 seconds before resetting the RM to give the RM a chance to save the name of the file to Flash. Failing to wait for this time will mean that the RM will not save the change to Flash and will restart running the same version of software that it was previously running.

- 4 To clear the selected image and revert to using the default image, enter `run clear` at the `rm:telnet>` prompt
- 5 Enter `run ?` to get help on the run command.

Adding and deleting users

The RM Telnet Server provides commands to add and delete users of Telnet sessions established with the RM. The default user name is `admin` and there is no password. Up to eight users can be specified for the RM Telnet Server.

To display a list of currently configured users, enter the command `user` at the `rm:telnet>` prompt. This command also displays the help for adding and deleting users.

To add a new user, enter the command `user add [name][password]` at the `rm:telnet>` prompt, where `[name]` is the user name and `[password]` is the password used for the connections.

To delete an existing user, enter the command `user delete [name]` at the `rm:telnet>` prompt, where `[name]` is the user name to delete from the list.

Monitor application

The monitor application is used for analysing the operation of the RM software. To start the Monitor application, type `monitor` at the `rm:telnet>` prompt.

To see a list of commands supported by the Monitor application, type `?` at the `rm:monitor>` prompt.

To exit the Monitor application, type `quit` or `exit` at the `rm:monitor>` prompt.

Router application

The router application is used for displaying statistics associated with the internal IP Router and Bridge in the RM. To start the Router application, type `router` at the `rm:telnet>` prompt.

The following commands are supported in the router application and these must be entered at the `rm:router>` prompt. Type `?` at the `rm:router>` prompt to get help on these commands.

- `show stack ip route`
- `show stack ip cache`
- `show stack ip protstats`
- `show stack ip portstats`
- `show stack ip arptable`
- `show stack ip rarptable`

To exit the router application, type `quit` or `exit` at the `rm:router>` prompt.

Rebooting the RM

To reboot the Resource Manager, enter the command `reboot now` at the `rm:telnet>` prompt. The RM will shut down any current activity and reboot after a period of five seconds.

System, Line, Trunk and WAN port status

To view the status of the RM and the availability of Lines, Trunks and WAN Ports, type `system` at the `rm:telnet>` prompt.

TFTP

The RM has a TFTP Client that is used to transfer files between the RM Flash File System and an external file storage source such as the Network Management Station disk. Typical uses for this including transferring updated RM software image files to the Resource Manager and transferring RM configuration files from the Resource Manager to the NMS.

To access the TFTP client, you must establish a Telnet session to the RM. Enter `tftp` at the `rm:telnet>` prompt. A `?` can be used to display the list of commands available within the RM TFTP Client. These commands include:

<code>connect</code>	connect to a TFTP server (e.g. <code>connect 192.195.112.15</code>)
<code>put</code>	send a file to the TFTP server (e.g. <code>put user.ini</code>)
<code>get</code>	receive a file from the TFTP server (e.g. <code>get rm.bin</code>)
<code>status</code>	display the state of connection to a Telnet Server
<code>quit</code>	quit the TFTP Client application

Note *When using the `get` command to download a file, the RM deletes the file of the same name (if it exists) from its Flash Memory before proceeding with the `get` operation. If the `get` operation fails, the RM Flash memory will no longer contain the file of the same name.*

Note that the server that you wish to connect to can also be entered when starting the TFTP client This is done by entering, for example, `tftp 192.195.112.15`.

A TFTP Client must connect to a TFTP Server associated with your NMS. OmniVision supplies a TFTP Server for use with the RM. The TFTP Server for OmniVision is added to the startup menu when it is installed. The server starts automatically when the NMS workstation is started.

To use the OmniVision TFTP server you must set the root directory from which files will be sent and received. Use the **Set Root** menu option to set this path. This must be a valid path accessible from your NMS workstation.

Note *Ensure that the **Retry Timeout** in the OmniVision TFTP Server is set to at least 5 seconds. If you do not do this, a 'Disk full' error is reported during the transfer. This is an inaccurate error message.*

In this release, the process has been automated using OmniVision's Flash File Manager.

Serial File Transfer

The RM has an in-built XMODEM file transfer capability that you can access via the DB9 serial port on the RM front panel. Refer to the RM section of the *J1000/J5000 Installation Manual* for information on connecting a PC serial port either directly using a cable, or remotely via a dial-up modem connection.

To use the Serial File Transfer capability you will need a PC Terminal Program that supports 9600 bit/s and the XMODEM file transfer protocol. Examples of these programs include HyperTerminal and PROCOMM.

Starting a connection

The DB9 serial port on the RM is used for two purposes:

- OmniVision management using the proprietary JASYNC protocol, and
- Serial File Transfer.

When first connecting a Terminal program, you may need to change the port from operating in a JASYNC mode back to File Transfer mode. To do this, enter at least ten (10) carriage return characters by holding down the <Enter> or <Return> key on your keyboard until the prompt `Monitor>` appears on the screen.

Note *OmniVision automatically switches the protocol on this port back to JASYNC if it is connected after a Serial File Transfer is established.*

Commands

You can enter the following commands at the `Monitor>` prompt for the Serial File transfer operation:

dir	get a listing of files in the RM Flash File System. Usage: <code>dir</code>
del	delete a file from the RM Flash File System. Usage: <code>del <filename></code>
xrecv	set up the RM to receive a file via the XMODEM protocol and store it in the Flash File System. Usage: <code>xrecv <filename></code>
download	download a file from the RM Flash File System to another module in the chassis. Usage: <code>download <filename> <cardslot> <interface> <line></code> Refer to the <i>J1000/J5000 Upgrade User Note</i> for details on downloading files to specific types of modules in a J5000 chassis.

Transferring a file

Proceed with the file transfer using the following steps:

- 1 Run the Terminal program on your PC and ensure that the chosen COM port settings are correct (9600 bit/s, no parity, 8 data bits, 1 stop bit).
- 2 If you are remotely establishing the connection via a modem, dial the number of the modem connected to the RM DB9 port and wait until the CONNECT message displays on the screen.
- 3 Set the RM into Serial File Transfer mode by pressing <Enter> until the `Monitor>` prompt appears.
- 4 Ensure that the file you are transferring to the RM does not already exist in the Flash by entering the command `dir`. If the file does exist, delete it by using the `delete` command.
- 5 Set up the RM to receive the file by typing the following command at the `Monitor>` prompt:

```
xrcv <filename>
```

where <filename> is the name of the file you want to store in the RM Flash File System. Do not enter any path information into the <filename>. For example, if the location you are transferring from is `C:\OmniVision\Bootimag\rm.gz`, **do not** enter that full path as the `xrcv <filename>` - just enter `rm.gz`.

Note You have 90 seconds to complete step 6. If the RM times-out, repeat step 5.

- 6 Set up your PC to upload the image using the XMODEM protocol. Typically, you do this by selecting XMODEM as the protocol and type in the path and filename of the file to transfer. Adapt this instruction to suit the Terminal program you are using.
- 7 Start the upload process and monitor the progress of the transfer by watching the statistics presented on the terminal screen.
- 8 When the transfer is complete, ensure that the file is in the RM. Do this by typing `dir` and confirm that the file is listed, and its status is OK. Repeat steps 4 to 8 for any subsequent files that you need to transfer.
- 9 Once the files are transferred, and if appropriate, proceed to download them to other modules in the chassis. If the transferred file is a new RM version, then reboot the RM.



Virtual eXchange

The Virtual eXchange (VX) provides a switching system suitable for a private ISDN network. It also provides call routing, call charging and service tones within the network. The private network can be connected to the public network, allowing calls to be made between users of the two networks.

Configuring a private network

This section describes the steps to configure a private network and details the parameters concerned. Some aspects of the configuration are optional, but are shown here for completeness. The optional parameters are:

- call charging
- base number
- service tone configuration
- endless loop prevention.

For VX application examples see page 5-30 in the *Applications* section.

Virtual eXchange configuration

Selecting the VX option on the Control Module or Resource Manager opens the Virtual eXchange dialog. The parameters set in this dialog affect all private network calls on the VX.

Enable VX Select this checkbox to enable the VX. All outgoing calls will be sent initially via the private network unless the line making the call is configured to use a 'Preferred' or 'Exclusive' TE. If the called party cannot be connected through the private network, the call is resubmitted to the public network via a TE mode interface.

Capacity Select the number of calls to be handled by the VX. The call capacity should be set to allow for the anticipated peak load of the VX.

Note *The Control Module or Resource Manager line resource is shared between the Virtual eXchange and the Virtual lines. Care should be taken in choosing the call capacity when the VX is used in conjunction with Virtual lines.*

Send dial tone for en-bloc call setup of ? digits or less Enter the number of digits, received in a call establishment message, for which you wish to provide a dialling tone. The default value is zero.

Configuring VX service tones

The VX provides service tones that are used to indicate the progress of private analog calls. The default cadences are based on Telstra Australia Specification 1529. To configure the tones select the **Tones** button from the VX configuration dialog.

Period Two on and two off periods of the cadence can be adjusted in increments of 200 ms.

Configuring the VX operation

To configure the VX operation select the **Operation** button in the VX configuration dialog.

Numbering

You must enter a valid range of numbers and Area Code for the VX to function correctly.

Base number	Enter the base number of the VX. This is particularly useful if you have more than one public connection.
Minimum number in this device	Enter the lowest valid number in the range.
Maximum number in this device	Enter the highest valid number in the range.
Area code (no escape prefix)	Enter the local area code without its leading zero.

Private network charging

Private network charging is optional. The VX can be configured to provide a combination of fixed cost per call and time charge.

Flag fall charge	Specify a fixed cost per call in multiples of charge units. The maximum flag fall is 65535 units.
Charge period	Specify a charge period from 1 to 65535 seconds. This parameter is multiplied by the setting in the Each unit costs parameter.
Each unit costs	Enter the cost of a unit in multiples of 0.1 cent up to 25.5 cents (1 to 255).

Endless loop prevention

Specify the maximum number of nodes that a call is allowed to pass through before reaching its destination. The default value is 10. You must set this parameter to a value greater than, or equal to, the length of the longest path within the private network.

Trunk module configuration

This section details the configuration settings for trunk modules used in linking private networks and those providing access to public networks.

Private network modules

Access to private network links between VX nodes is provided by IPMNs, or DTMs in NT mode.

Configure IPMN

Refer to page 4-6 of the *Module configuration* section for information on configuring the IPMN.

Configure DTM NT

Refer to page 4-79 of the *Module configuration* section for information on configuring the DTM NT.

Public network modules

Primary Rate access to the public ISDN is provided by IPMTs. Basic Rate access is provided by BRMTs.

Configure IPMTs

Refer to page 4-3 of the *Module configuration* section for information on configuring the IPMT.

Configure BRMT

Due to the characteristics of the Basic Rate specification, the link will deactivate when there are no active calls established across the link. To prevent link deactivation, we recommend that the **Retain Layer 2 establishment** and the **Disable T309** checkboxes in the **Basic Rate TE Configuration** dialog for the respective BRMT are enabled.

Refer to page 4-17 of the *Module configuration* section for information on configuring the BRMT.

Lineset route configuration

Configuring linesets with multiple routes allow you to develop private networks that incorporate redundant paths and can accommodate peak traffic overflow. Information on configuring linesets can be found on page 3-46.

Linesets used to define routes in a VX network must be configured as **Non-local**. Selecting **Non-local** sets the following parameters:

- Linear Hunting (keep list static)
- Return cause of 'No circuit/channel available' to allow fallback TE & NT resubmission.

Reverse IPMN order should be selected. Since the IPMN does not provide channel negotiation, it is possible that, if the linesets are in the same order, calls received simultaneously from an E1 link and the VX will collide. Reversing the order lessens the chance of call collision. Reversal should only be at one end of the private link.

Control Module/Resource Manager configuration

The following parameters are associated with the configuration of a private network:

Manager access number	This must be configured. If it is left blank, all calls will be connected to the UMPE (User Management Protocol End), the end point for OmniVision connections.
Allow resubmission to TE	Enable to allow calls to be sent to the public network when VX is enabled.
Allow resubmission to NT	Enabled.
Log resubmitted calls	Disabled.

TimeFrame

Overview

A TimeFrame Virtual Line is designed to provide optimum bandwidth efficiency for data while still providing high quality of service for voice in a voice and data network. It does this by the dynamic addition and removal of TDM slots for voice within a packet stream containing all of the data between two sites.

A TimeFrame Virtual Line is a channel existing between two devices that contains two distinct sections:

- a packet channel for carrying data and;
- a TDM section for carrying voice.

A single TimeFrame Virtual Line can operate up to 1472 kbit/s. Refer to page 3-22 for restrictions on TimeFrame bandwidth allocation.

The voice circuits within a TimeFrame VL are compressed using advanced voice compression techniques to ensure that they occupy minimum TDM bandwidth even when active. The remainder of the TimeFrame VL channel is allocated to a Packet Channel.

There are several aspects to configuring TimeFrame. The following list summarises those aspects and provides cross references to the information:

- Refer to *Virtual Lines* on page 3-50 for information on configuring TimeFrame Virtual Line call handling aspects such as line numbers, call establishment methods and data rates.
- Refer to *Packet Ports* on page 3-34 for information on configuring a Packet Port to provide access to the packet capabilities of a TimeFrame Virtual Line. This includes both internal and external data devices running HDLC based protocols such as PPP, X.25 and Frame Relay.
- Refer to *Switching Packet Data* on page 3-21 for information on configuring the packet switch to switch data from the RM's internal IP Router/Bridge and Packet Ports onto a TimeFrame Virtual Line.
- Refer to *Frame Relay PVC management* on page 3-42 for details on configuring a Packet Port to accept data from a connected device that supports the user-side Frame Relay PVC management protocol.
- Refer to *Wideband Operation* on page 3-38 for information on using TimeFrame Virtual Lines and Packet Ports in a Wideband mode (>64 kbit/s).

See the *TimeFrame Virtual Line applications* on page 5-3 for examples of configuring various types of TimeFrame networks.

TimeFrame limitations

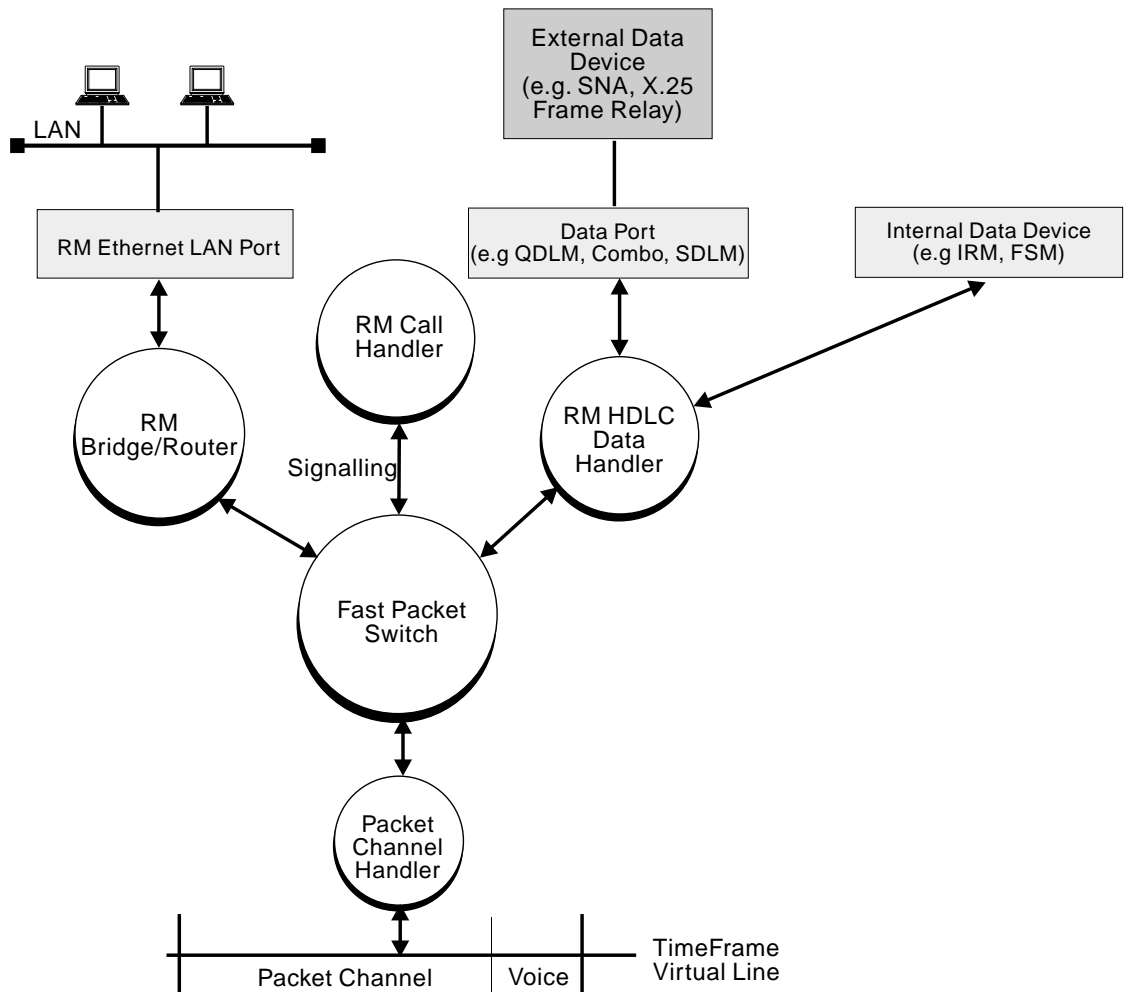
The following limitations apply when using TimeFrame:

- For voice calls, a 64kbit/s TimeFrame Bearer always reserves a minimum of 8kbit/s for data and end-to-end signalling. For example, a 128 kbit/s TimeFrame Virtual Line (two bearers) reserves a minimum of 16 kbit/s, thus supporting up to fourteen 8 kbit/s compressed voice calls. If the two bearers are used only for data, the full 128 kbit/s bandwidth is available.
- The RM can handle a combined bandwidth of 23 x 64kbit/s (1472 kbit/s) channels for all Category 1 Packet Ports and TimeFrame Virtual Lines. For example, an RM cannot have an additional Category 1 Packet Port or TimeFrame Virtual Line if there is already a 704kbit/s Category 1 Packet Port and a 768kbit/s TimeFrame Virtual Line configured. Refer to page 3-34 for information on *Category 1 Packet Ports*.
- A Subrate Virtual Line can have at most one Packet Port or at most one TimeFrame Virtual Line as its member. A Subrate Virtual Line cannot simultaneously have a Packet Port and a TimeFrame Virtual Line as its members. Refer to *Virtual Lines* on page 3-50 for more information on Subrate Virtual Lines.
- A Packet Port and a TimeFrame Virtual Line can be members of the same Wideband Virtual Line, as long as they do not share the same Subrate Virtual Line. Refer to *Virtual Lines* on page 3-50 for more information on Subrate Virtual Lines.

Switching Packet Data

This section defines how the RM takes data from multiple sources and multiplexes that over the Packet Channel within the TimeFrame Virtual Line.

The switching of packet data within the TimeFrame VL Packet Channel is controlled by a Fast Packet Switch running on the Resource Manager. There are multiple sources of data that can be fed into the Fast Packet Switch (FPS) from within a system. Each of these sources of data is shown in the diagram below.



The RM Bridge/Router bridges and routes data between the locally connected Ethernet LAN port and each of the TimeFrame VLs that support a connection to the Bridge/Router.

The RM HDLC Data Handler takes HDLC-based data such as X.25, SNA and Frame Relay from an externally connected device via one of the available data ports (for example, QDLM, COMBO or SDLM). It also accepts data from internally connected devices such as an IRM or FSM. The data from both sources is passed to the Fast Packet Switch for switching to the appropriate destination.

To allow support for a wide range of HDLC-based protocols, the Data Handler operates in two modes:

- encapsulation
- transparent.

In encapsulation mode, also referred to as 7E Data Handling, the incoming HDLC packet is encapsulated in a Frame Relay frame by prepending a Frame Relay header and an RFC1490 header. This allows the FPS to recognise it as a FR frame and switch it accordingly.

In transparent mode, the Data Handler does not alter the incoming HDLC frame and passes it directly to the FPS. As the frame is passed directly, it must have a Frame Relay header to allow it to be switched.

The only protocol supported in transparent mode is Frame Relay. All other protocols must be set for encapsulation within a Frame Relay frame.

Each of the above data sources within the TimeFrame VL are framed using the Frame Relay protocol to allow Fast Packet Switching of all of the data within the Packet Switch. All data is RFC1490 encapsulated to allow demultiplexing of each data source within the TimeFrame VL.

To further enhance the bandwidth usage of the TimeFrame VL Packet Channel, data compression can be selectively applied to each of the data sources discussed in this section.

If data compression is enabled, the RM will try to compress all packets on the Frame Relay route. If a compressed packet is larger than the original packet, due to it already being compressed, the original packet is forwarded across the link rather than the larger compressed packet. This makes the valuable trunk bandwidth more efficient.

The following sections contain more detail on the operation and configuration of each of the components in the data path described here.

Backplane bandwidth allowance

TimeFrame Virtual Lines use the RM internal timeslots, rather than the backplane timeslots. TimeFrame bits are Subrate-switched directly from the internal timeslots to the trunk module's timeslots. However, if a TimeFrame Virtual Line is connected through a BCAM-ISO, it will require two backplane timeslots to switch data into the BCAM-ISO.

For example, a 128kbit/s Timeframe Virtual Line, connected through a BCAM-ISO to an IPMT, requires two backplane timeslots for the IPMT and four (2x2) additional backplane timeslots for the TimeFrame Virtual Line and the BCAM-ISO. The additional backplane timeslots are allocated dynamically.

Refer to *Appendix J – System Limits* for more information on backplane timeslot allocation

Subrate switch usage

No Subrate switch is required to map a TimeFrame Virtual Line to a Subrate Virtual Line, as the RM can switch Subrate bits from RM internal timeslots to the destination timeslots.

Configuring the TimeFrame VL packet channel

This section describes how to configure a TimeFrame Virtual Line to send data packets from devices connected to a J1000/J5000. These data packets can originate from:

- externally connected devices such as an SNA terminal or a router via a physical data port on a COMBO or DLM, or;
- internally generated by the RM Bridge/Router from a connected LAN, or;
- internally connected modules such as the IRM or FSM, or;
- end-to-end signalling messages from the RM call handler.

All data packets sent over a TimeFrame VL must transit through the Fast Packet Switch (FPS). The FPS is responsible for switching a data packet from a source data service to a destination data service. A data service refers to an interface that handles data packets. These include:

- HDLC Data Handler (Data from COMBO or DLM data ports)
- Bridge/Router Handler (Data from the RM Internal Bridge/Router)
- Packet Channel Handler (Data from a TimeFrame VL Packet Channel).

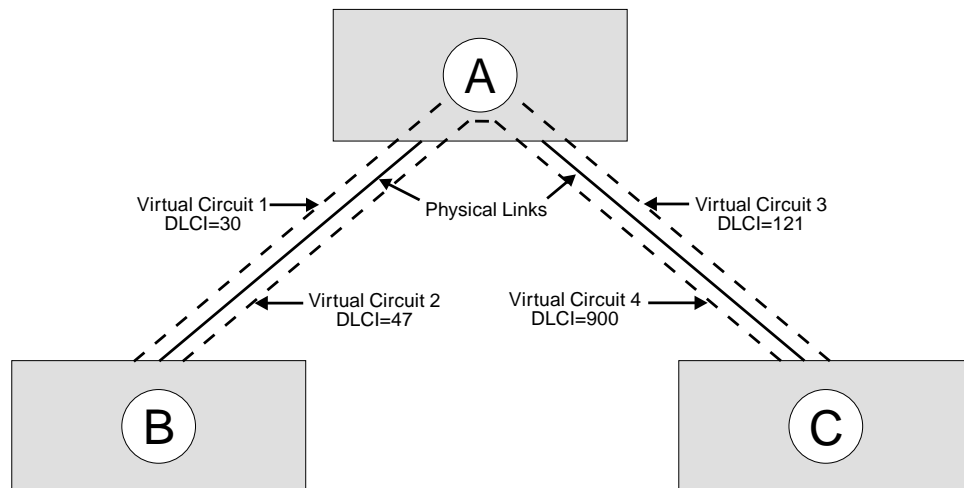
To allow the FPS to correctly switch from source to destination each data packet must have information within the packet to allow this switching to occur. The FPS in the RM uses Frame Relay as the switching protocol as it natively provides a number of features which are beneficial to fast switching of frames.

Each Frame Relay frame has a standard 2 byte header which contains information about the frame. The primary bit of information within the header is an address that is commonly referred to as the DLCI (Data Link Circuit Identifier). In a standard Frame Relay frame the DLCI can contain values between 0 and 1023, allowing up to 1024 unique addresses to be assigned to frames.

Note *In the RM, only DLCIs 30-991 are available for use on TimeFrame Virtual Lines. All other DLCIs are reserved.*

A DLCI is used as an address for a connection between two devices within a network. This connection is commonly referred to as a Virtual Circuit (VC). As a result of this, each physical Frame Relay interface can theoretically support up to 1024 Virtual Circuits going to either the same destination or switched within an external network to different destinations. This scheme allows multiple sources of data to be multiplexed over a single link for delivery to one or more remote destinations. The RM can support up to 32 Virtual Circuits on each TimeFrame Virtual Line.

The following diagram shows a three node network with four Virtual Circuits connecting the sites over two physical links. Note that the Virtual Circuit Number is only a label to identify the VC within the network diagram. It does not have to correspond to the DLCI for the link. It is the DLCI that is used by the Packet Switch to switch data packets.



All data services that communicate with the Fast Packet Switch in the RM do so through a Frame Relay Interface. It is this Frame Relay Interface that defines the DLCI and Virtual Circuit parameters associated with the data service, and allows data packets to be switched within the Fast Packet Switch from source to destination.

In the RM, the Data Handler, Bridge/Router and Packet Channel Handler all have Frame Relay interfaces associated with them.

The steps involved in configuring the TimeFrame VL to accept data consist of:

- Configuring a Frame Relay Interface and its Virtual Circuits for each data service connected to the Packet Switch
- Associating that Virtual Circuit with one or more data service types
- Configuring the Fast Packet Switch Routing Table to switch the data packets between source and destination data services.

Configuring a Data Handler

The Data Handler is used for connecting external or internal data devices to a TimeFrame data service, for example, an SNA terminal or a router. These devices are connected through either the data ports provided by modules such as a COMBO or DLM, or internal ports such as IRM WAN ports. There are thirteen Frame Relay Interfaces associated with the Data Handler. These interfaces are called Packet Ports. In the configuration they are referred to as Packet Port 1 to Packet Port 13.

Note *To optimise the performance of the RM we recommend that you select Category 1 Packet Ports in preference to Category 2 Packet Ports where possible. See Packet Ports on page 3-34 for a description of the differences between the two port categories.*

OmniVision contains a Data Handler application to allow you to set up your devices. See the *OmniVision User Manual* for details. The following information is provided for setting up the data handler using a Telnet or Virtual Terminal session. It must be used in conjunction with OmniVision to complete the configuration.

The following steps apply to configuring a Data Handler:

- 1 Configure the data port to which the device is connected, for example, COMBO, DLM or IRM. Refer to the particular module sections for more details.
- 2 Run the Telnet application and connect to the RM to configure the Frame Relay Interface and Virtual Circuits. Initiate the configure mode by entering `configure` at the `rm:telnet>` prompt. A complete set of Telnet commands is shown in *Appendix M*.
- 3 View the current settings of the Data Handler being configured by showing the Frame Relay Interface for the corresponding Packet Port using the command

```
show fr-int <interface>
```

where `<interface>` takes on values `pktport1` to `pktport13`.

The Virtual Circuit determines in which mode, encapsulation or transparent, the Packet Port, and hence the Data Handler operates. Each Packet Port only supports one Virtual Circuit when operating in encapsulation mode. Up to 64 Virtual Circuits are supported on each Packet Port when operating in transparent mode with Frame Relay PVC Management enabled.

Note *The Packet Port interfaces are preconfigured to operate as 7E data handlers. Because of this, they operate in encapsulation mode and prepend a Frame Relay header on all received frames. A DLCI has been configured for the Virtual Circuit. This is only required to be changed if the setting is not suitable for use, for example, if a data type other than 7E encapsulation is required, or the port is to operate in transparent mode.*

- 4 To add a new Virtual Circuit to the Packet Port, the original one must first be deleted. This is done using the command:

```
delete vc <interface> <vc name>
```

where:

<interface> takes on values pktport1 to pktport13 and is the interface from which the Virtual Circuit is being deleted;
and <vc name> is the name of the Virtual Circuit to be deleted from that interface.

- 5 A new Virtual Circuit is added using the command:

```
add vc <interface>
```

where <interface> takes on values pktport1 to pktport13 and is the interface to which the Virtual Circuit is being added.

- 6 Enter the name of the Virtual Circuit.
- 7 Enter the DLCI for the Virtual Circuit.

Note *This value is not used if the data type is set to **fr** for Frame Relay, and Frame Relay PVC Management is disabled.*

- 8 Enter the Data Type associated with the Virtual Circuit. A value of **fr** selects transparent operation. All other values select the type of RFC1490 encapsulation used on the packet. 7e is the standard proprietary encapsulation.

Note *Only the High Priority data type field is used for the Packet Port interfaces. No value should be entered for Medium and Low priority.*

- 9 Establish a manual connection between the data port to which the device is connected, and the Data Handler on the RM, configured in the previous steps. Use OmniVision to establish this manual connection. The manual connection is connected from the data port, to which the device is connected, to the corresponding Packet Port on the RM. For example, if **pktport1** was configured in steps 2 to 8 above, a manual connection would be established from the data port to Packet Port 1. If the Packet Port contains multiple lines, each line must be manually connected.
- 10 Repeat this procedure for all data ports to be connected to a Data Handler.
- 11 Configure the remaining required data services for the device and then refer to *Configuring the Fast Packet Switch* on page 3-32 for connecting data services together.

Configuring the Bridge/Router

The RM contains a Transparent Bridge and an IP Router to allow traffic received on the LAN to be routed or bridged across a TimeFrame Virtual Line. There is one Frame Relay Interface associated with the Bridge/Router and it supports eight Virtual Circuits, allowing up to eight Bridge/Router ports to be switched onto TimeFrame Virtual Lines. The Bridge Filtering Database entries are aged out after five minutes. To configure the Bridge/Router:

- 1 Run the Telnet application and connect to the RM to configure the LAN and WAN ports of the Bridge/Router. Initiate the configure mode by entering `configure` at the `rm:telnet>` prompt. A complete set of Telnet commands is shown in *Appendix M*.
- 2 Configure the Ethernet LAN port to suit the network. The RM contains a `show lan` command to enable you to view the settings of the Ethernet LAN port and a `set lan` command to alter these settings. This command configures both the bridging and the IP routing parameters associated with the port. To accept the current setting of a parameter, press `<enter>` at the prompt for the new parameter value.
- 3 Configure the settings on each of the WAN ports to suit the network. The RM contains `show wan <port>` and `set wan <port>` commands where `<port>` has the value of 1 to 8. These commands allow you to view and alter the settings of the eight RM Bridge/Router WAN ports that can be connected to the TimeFrame trunks.

Note *When setting a WAN port, one of the parameters determines whether the link is unnumbered or not. If the link is configured as unnumbered, a further prompt for a Remote Address is made. The Remote Address is the network address of the remote end of the unnumbered link. This value can be entered here as an alternative to creating a static route. If RIP is enabled on this port or a static route is entered into the static route table, the Remote IP Address can remain at 0.0.0.0.*

- 4 The Transparent Bridge and the Spanning Tree Algorithm can be globally enabled or disabled by using the global bridge settings. To view the settings use the command `show bridge`. To alter the settings use the command `set bridge`.
- 5 IP Routing and RIP can be globally disabled or enabled using the global IP settings. To view the settings use the command `show ip`. To alter these settings use the command `set ip`.
- 6 To set a default IP route for the RM IP Router use the `set defaultroute` command. The current setting for the default route can be viewed using the `show defaultroute` command. The default route can be deleted using the `delete default route` command.

Note *If the default route is on an unnumbered link then the Port Number of the WAN port to which the traffic is sent must be entered. This takes on a value of 1 to 8 for WAN ports 1 to 8 of the RM Bridge/Router. If the default route is not unnumbered then enter the IP address of the device to which the traffic should be sent.*

- 7 If RIP is not used on one or more ports, static routes may be required to allow IP traffic to be routed from the RM IP Router. To view a list of configured static routes use the command `show staticroute`. To add a static route to the routing table enter the command `add staticroute <entry>` where `<entry>` is a free entry number within the static routing table. The `<entry>` value can be determined from the `show staticroute` command.

To delete a static route from the static routing table, enter the command `delete staticroute <entry>` where `<entry>` is the entry number of the static route to delete.

Note *If you are using the RM to interconnect sub-networks of the same network over WAN Ports, you must use **Numbered WAN Ports**. Using **Unnumbered WAN Ports** prevents the inclusion of subnet routes in the RIP updates, and may cause improper operation of the network.*

- 8 Each of the eight WAN ports must be connected to a TimeFrame Virtual Line via the Packet Switch routing table. The following table shows the mapping between the WAN port number, the Port ID and the Port DLCI. The WAN Port number is configured using the `set wan <port>` command. This information is used in the Packet Switch routing table that connects the Bridge/Router to that WAN port.

WAN Port number	Port 1 ID	Port 1 DLCI
1	router	51
2	router	52
3	router	53
4	router	54
5	router	55
6	router	56
7	router	57
8	router	58

For example, to connect WAN port 2 from the Bridge/Router to Packet Channel Handler trunk2, an entry must be added to the Packet Switch Routing Table which has the following settings:

- Port 1 ID set to **router**
- Port 1 DLCI set to **52**
- Port 2 ID set to **trunk2**
- Port 2 DLCI set to correspond to a valid Virtual Circuit within the trunk2 Frame

Relay Interface

- The Data Type set to **all**.

Note *The values for Port 1 DLCI are fixed and cannot be changed. This DLCI value can however be mapped to an alternative value if required by switching this entry to a Virtual Circuit within Packet Channel Handler that has the required DLCI.*

- 9** Configure the remaining required data services for the device and then refer to *Configuring the Fast Packet Switch* on page 3-32 for connecting data services together.

Note *You must restart the RM if you change any of the parameters for the IP Router or the Bridge. This includes the use of any of the following commands:*

```
set lan
set wan <port number>
add staticroute
delete staticroute
set defaultroute
delete defaultroute
set ip
set bridge
```

Configuring a Packet Channel Handler

The Packet Channel Handler (PCH) is responsible for prioritising and queuing data for the TimeFrame trunks. There are eight Frame Relay Interfaces associated with the PCH for accessing the eight TimeFrame trunks. In the configuration they are referred to as trunk1 to trunk8. The following steps apply to configuring a Packet Channel Handler:

- 1 Configure the TimeFrame Virtual Line as detailed on page 3-50.

The association between a PCH Frame Relay Interface and a TimeFrame VL is made through the Call Establishment dialog in OmniVision when setting up a TimeFrame Virtual Line. When configuring a TimeFrame VL, a TimeFrame Trunk Name of trunk1 through to trunk8 is automatically assigned by OmniVision. Use that trunk name when configuring the Packet Channel Handler (PCH) that is associated with that TimeFrame VL. This means that there is a one-to-one relationship between a TimeFrame VL and a PCH.

- 2 Run the Telnet application and connect to the RM to configure the Frame Relay Interface and Virtual Circuits. Initiate the configure mode by entering `configure` at the `rm:telnet>` prompt. A complete set of Telnet commands is shown in *Appendix M*.
- 3 View the current settings of the Packet Channel Handler being configured by showing the Frame Relay Interface for the corresponding trunk. To do this, use the command:

```
show fr-int <interface>
```

where `<interface>` takes on values trunk1 to trunk8.

Note *The trunk interfaces are preconfigured with two Virtual Circuits to pass all data across a trunk. Each trunk can support up to 32 Virtual Circuits.*

- 4 To delete a Virtual Circuit from a trunk use the command:

```
delete vc <interface> <vc name>
```

where:

`<interface>` takes on values trunk1 to trunk8 and is the interface from which the Virtual Circuit is being deleted;

and `<vc name>` is the name of the Virtual Circuit to be deleted from that interface.

- 5 To add a new Virtual Circuit to the trunk use the command:

```
add vc <interface>
```

where <interface> takes on values trunk1 to trunk8 and is the interface to which the Virtual Circuit is being added.

- 7 Enter the name of the Virtual Circuit.
- 8 Enter the DLCI for the Virtual Circuit.
- 9 Enter the Data Types and priorities of the data types associated with the Virtual Circuit.

Note *If data priority is a concern, you should use the first configured Virtual Circuit on a trunk (i.e. DLCI 100) to set up data priority. For instance, high priority=**e2e**, medium priority=**7e** and low priority=**all**. Avoid setting up too many Virtual Circuits in a trunk as the priority scheme is not implemented across multiple Virtual Circuits. End-to-end signalling always travels on DLCI 100.*

If all data is to be passed at equal priority, enter **all** in the High Priority Data Type field and nothing for the Medium and Low Priority. If the Virtual Circuit is to contain a number of different data types that need to be prioritised, enter the data types for High Priority, Medium Priority and Low Priority respectively.

Note *Only Data Types specified in the High, Medium and Low Priority fields for a Virtual Circuit will be passed across that Virtual Circuit. Be sure that, if you split up the data types, you cover all data types travelling across the Virtual Circuit.*

- 10 Repeat the appropriate parts of this procedure for creating additional TimeFrame Virtual Line to Packet Channel Handler mappings. Also use this procedure for adding or deleting Virtual Circuits from a Packet Channel Handler.
- 11 Configure any remaining data services required for the device and then refer to *Configuring the Fast Packet Switch* on page 3-32 for connecting data services together.

Configuring the Fast Packet Switch

The Fast Packet Switch (FPS) is used to switch data packets between any number of configured Frame Relay Interfaces. Each entry in the Fast Packet Switch table that describes how to switch a packet from a DLCI on one interface to another DLCI on a different interface is called a route. The Resource Manager's FPS can support up to 512 routes in its route table. To configure the Fast Packet Switch:

- 1 Establish a Telnet session to the RM. A complete set of Telnet commands is shown in *Appendix M*.
- 2 Enter `configure` at the `rm:telnet>` prompt.
- 3 To show a list of all of the configured routes in the FPS enter the command `show fr-route`.
- 4 To show details about a specific entry in the FPS route table enter the command `show fr-route <entry>` where `<entry>` is the number of the entry in the FPS route table.
- 5 To add an entry into the FPS route table enter the command `add fr-route <entry>` where `<entry>` is an unused entry number in the routing table. Use the `show fr-route` command as described above to find an unused entry in the FPS route table.

Note *To replace an existing FR route entry, enter the <entry> number corresponding to the route that you want to replace.*

When adding a FPS route entry, enter the information as prompted. Note that each FPS route entry entered is bi-directional so packets will be switched both ways between the Port 1 and Port 2 values entered.

- 6 Enter the Port 1 Frame Relay Interface ID using the identifier (for example, `pktport1`) from the appropriate interface entered for the data services. Enter the DLCI corresponding to the Virtual Circuit on the Frame Relay Interface to switch.
- 7 Enter the Frame Relay Interface ID identifier (for example, `trunk1`) and DLCI for the Virtual Circuit for Port 2.
- 8 Enable or disable compression for this FPS route entry as appropriate.

Typically, you would disable compression for an FPS route entry corresponding to a switch between two Packet Channel Handler data services. This is because the FPS is only acting as a transit node between the two connected devices.

Compression would typically be enabled, however, for a Data Handler or RM Bridge/Router data service connected to a Packet Channel data service on a TimeFrame VL. This compresses the data coming from these services and hence reduces the amount of data actually transmitted on the TimeFrame VL.

Note *When enabling compression, the order of the Port 1 and Port 2 values as entered above is very important. Data packets coming from Port 1 and switched to Port 2 are compressed while data packet coming from Port 2 and switched to Port 1 are decompressed. This means Port 1 must correspond to either a Data Handler or RM Bridge/Router data service and Port 2 must correspond to a Packet Channel Handler data service.*

- 9** Enter the Data Types switched in this FPS route entry. This field allows the Data Type of the packet to be used to demultiplex and switch to the correct interface when a DLCI is used to carry more than one type of data.

For example, if a Data Handler and RM Bridge/Router are switched onto Virtual Circuits within a Packet Channel Handler, then two route entries would be required in the FPS. This is described in *Configuration of Packet Channel Handler Service* on page 3-30.

The first route entry would have:

- Port 1 set to the FR Interface ID and DLCI of the Data Handler
- Port 2 set to the FR Interface ID and first DLCI of the Packet Channel Handler
- Data Type for this entry would be **7E Data**.

Note *If the Data Handler is configured to operate transparently, that is data type of fr, the Data Type for this entry must be set to all.*

The second route entry would have:

- Port 1 set to FR Interface ID and DLCI of the RM Bridge/Router
- Port 2 set to the FR Interface ID and second DLCI of the Packet Channel Handler
- Data Type for this entry would be **all**.

Note *The data originating from the RM Bridge/Router and a Data Handler must go over different Virtual Circuits on a TimeFrame Virtual Line.*

- 10** To delete an entry in the FPS route table enter the command `delete fr-route <entry>`.

Packet Port

Note *In earlier releases Packet Ports were referred to as Packet Lines. Packet Lines supported up to 64 kbit/s, Packet Ports support greater than 64kbit/s.*

A Packet Port is an interface between a Resource Manager HDLC data handler and an HDLC data device. The interface can either be internal or external to the chassis. An external data device is usually connected to a Packet Port through a digital data port such as a COMBO data port. The following modules and line-types are supported by Packet Ports:

- COMBO Digital Line Interfaces
- All Digital Line Modules (DLMs) except the ADLM V.24
- All ISDN Primary Rate Modules (IPMs)
- Digital Trunk Module (DTM)
- BCAM-ISO
- IRM
- FSM
- Subrate, Non-Subrate and TimeFrame Virtual Lines.

Packet Ports only support Manual Connection.

There are 13 Packet Ports in each RM, and they can be categorised into two groups based on their functions:

- **Category 1:** Packet Port 1 to 7. These support non-CCITT rate adaption, idle or flag fill, NRZ line encoding and greater than 64kbit/s operation.
- **Category 2:** Packet Port 8 to 13. These support CCITT or non-CCITT rate adaption, idle fill, NRZ, NRZI mark or NRZI space line encoding and up to 64kbit/s operation.

Note *The RM can handle a combined maximum bandwidth of 23 x 64kbit/s channels for all Category 1 Packet Ports and TimeFrame Virtual Lines. For example, an RM cannot handle an additional Category 1 Packet Port or TimeFrame Virtual Line if a 704kbit/s Category 1 Packet Port and a 768kbit/s TimeFrame Virtual Line are already configured.*

A Packet Port can be manually connected to another data port such as an HSDM port. In the case of a Wideband (>64 kbit/s) connection, each Packet Port line should be manually connected to the corresponding data port line. For example, the first Packet Port line must be connected to the first HSDM line, the second to the second, and so on. Packet Ports do not support switch call or XSPC operations.

A Packet Port can be a member of a Wideband (non-Subrate) or a Subrate (<=64kbit/s) Virtual Line.

Notes

A Subrate Virtual Line can have, at most, one Packet Port or one TimeFrame Virtual Line as its member. A Subrate Virtual Line cannot simultaneously have a Packet Port and a TimeFrame Virtual Line as its members.

A Packet Port and a TimeFrame Virtual Line can be members of the same Wideband Virtual Line, as long as they do not share the same Subrate Virtual Line.

In the case where a Packet Port is a member of Subrate Virtual Line, calls cannot be made or cleared from the Packet Port as it does not support switch call operations.

Backplane bandwidth allowance

A Packet Port uses internal RM timeslots that cannot be accessed by other modules. Furthermore, a Category 2 Packet Port does not possess Subrate switch functions, so it cannot switch Subrate bits onto a backplane timeslot. The RM sets up temporary backplane timeslots that can be accessed by other modules, and allows Subrate switching from the temporary timeslots to the destination timeslots. The following table lists the number of temporary backplane timeslots required by each type of connection.

Packet Port Category	Manual connection (except BCAM-ISO)	Manual connection to BCAM-ISO	Member of a Virtual Line	Member of a Subrate Virtual Line	Member of a Subrate Virtual Line through a BCAM-ISO
1	0	1	0	0	See Note 1
2	0	See Note 2	0	1	See Note 1

Notes

1 *The RM does not support the case where a Packet Port is a member of a Subrate Virtual Line, which in turn is connected through a BCAM-ISO.*

2 *Since Category 2 Packet Ports only operate up to 64 kbit/s, there is no need to use a BCAM-ISO.*

Packet Ports use backplane bandwidth dynamically. This means that backplane bandwidth is allocated as required. For example, a 128 kbit/s Packet Port does not use backplane bandwidth unless it is connected.

If the RM fails to find an available backplane timeslot, it will send an alarm to the NMS.

If a backplane timeslot, which has been allocated to a Packet Port, has been de-allocated due to an insertion of another module, the RM will locate another timeslot for the Packet Port. If the RM fails to find an available backplane timeslot, it will send an alarm to the NMS.

Subrate switch usage

A Subrate switch is **not** required to map a Category 1 Packet Port to a Subrate Virtual Line, as the RM can switch Subrate bits from internal timeslots to destination timeslots.

A Subrate switch is also **not** required if a Category 2 Packet Port is manually connected to another port.

A Subrate switch **is** required if a Category 2 Packet Port is a member of a Subrate Virtual Line.

Configuring a Packet Port

Port Configuration for Packet Port

Port ID	Enter the Port ID . This is a number you can assign to uniquely identify a line and is not related to the Port Name.
Port Name	Enter the Port Name . This can be allocated to a port for easy identification. It is useful for linesets and Virtual Lines.
Category 1 Port Setup:	
Rate Adaption	Only Non-CCITT is available for Category 1 ports. See page 2-8 for details.
Line Fill	Set to either Idle or Flag .
Line Coding	Only NRZ can be selected for Category 1 ports
Max Port Speed	Set the Max Port Speed up to 1472k . The maximum port speed displayed is governed by the difference between the aggregate Category 1 Packet Ports and TimeFrame Virtual Line speed (i.e. 1472K), and the bandwidth already allocated to configured Category 1 Packet Ports and TimeFrame VLS. The number of Packet Port and TimeFrame bearer channels in use are displayed above this field.
Category 2 Port Setup:	
Rate Adaption	Select Non-CCITT or CCITT . See page 2-8 for details.
Line Fill	Only Idle is available for Category 2 ports.
Line Coding	Set to either NRZ , Mark NRZI or Space NRZI .
Max Port Speed	Set the Max Port Speed up to 64k .

Call Establishment Modes

Semipermanent Select **Enable** and enter the semipermanent label to set up a semipermanent call.

Hotline Enable to set up a Hotline call. In a Packet Port or Virtual Line configuration where multiple (64kbit/s) lines are used, each line requires a Hotline number. If a subaddress is not used, the Hotline number for subsequent (64kbit/s) lines is automatically incremented, based on the Hotline number configured. The automatic Hotline number increment can be disabled by checking the **Don't increment Hotline No.** checkbox.

Select **XSPC** to enable a Switched semipermanent connection. This is only available with a Hotline call.

Call options Select **Auto Answer** to connect the incoming call to the line as soon as it is received, irrespective of the signalling state of the connected equipment.

Advanced Settings

Select the **Advanced Settings** button to go to the Line configuration for the Packet Port. See page 2-5 for further details.

Note *Flag Fill must be used when connecting a Packet Port to an IRM. This means that a Category 1 Packet Port must be used for this purpose.*

Wideband operation

A Wideband port is a data port or Virtual Line that is capable of operating at greater than 64kbit/s. Narrowband ports operate at 64 kbit/s or less. The following ports and Virtual Lines support Wideband rates:

- Category 1 Packet Ports
- TimeFrame Virtual Lines
- COMBO module data ports (version 5.00 onwards)
- HSDM data ports.

All Wideband ports exhibit similar characteristics in way they use Subrate switches and the adaptive speed feature. Unless otherwise stated, all Wideband ports follow the guidelines specified below.

Subrate switch usage

When a data port is configured to be less than 64kbit/s and is a member of a Subrate Virtual Line, it requires a Subrate switch to map its Subrate bits onto a Subrate Virtual Line.

When a Wideband data port is configured to be greater than 64kbit/s and one of its lines is a member of a Subrate Virtual Line, all data port lines need constant delay Subrate switches to switch all bits onto the Subrate Virtual Line. Constant delay Subrate switches are needed to avoid differential delay between the lines. For instance, a 120kbit/s data port that is a member of a Wideband Virtual Line requires two Subrate switches.

Note *Subrate switches are not required for switched calls between data ports.*

Constant delay Subrate switches can be found in the following modules:

- MFCM V.5.01 or later
- COMBO V5.00 or later
- HSDM.

Appendix Q lists the number of constant- and variable-delay Subrate switches for each module.

Note *An insufficient number of Subrate switches will result in corrupted data during transmission. An alarm is sent to the NMS if there are not enough Subrate switches to handle the call.*

If a module that has been assigned to provide Subrate switches for a data port is disconnected, the RM locates other Subrate switches in the chassis for that data port. If the RM fails to locate available Subrate switches, an alarm is sent to the NMS.

Subrate switches are assigned dynamically. This means that they are only assigned to a port when the port is connected.

The exception to the Subrate switch usage rule are Category 1 Packet Ports and TimeFrame Virtual Lines. These can switch Subrate bits from internal RM timeslots to destination timeslots.

Adaptive speed feature

Under normal circumstances, a Wideband port uses its configured speed to connect to another port or trunk. If the speed of the connected device is lower than the speed of the Wideband port, the port adapts to the speed of the connected device.

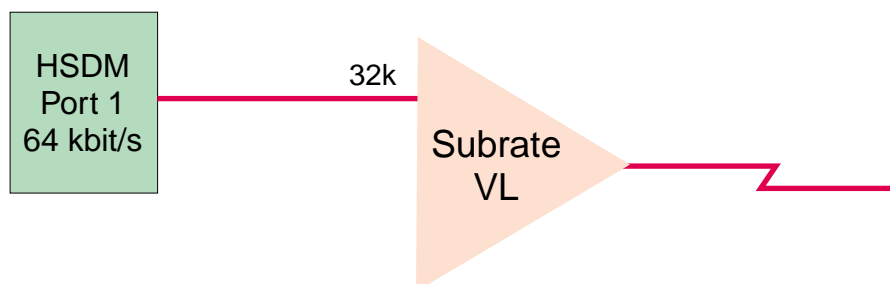
The adaptive speed feature is supported in a connection:

- between a Wideband port and a trunk, or
- between a Wideband port and a Virtual Line (Subrate or Wideband).

This feature is not supported in a connection between two Wideband ports, such as between a Packet Port and a HSDM port.

A Wideband port adapts to the speed of the connected device only if the Wideband port is operating in non-CCITT rate adaption mode, and its configured speed is higher than the speed of the connected entity. In the case where the configured Wideband port speed is lower than the speed of the connected device, the Wideband port maintains its configured speed.

For example, when an HSDM port is configured to 64 kbit/s, and it is a 32 kbit/s member of a Subrate Virtual Line, it adapts to the 32 kbit/s or to the speed at which it is connected. The following diagram illustrates this.



Since the connected speed may be different from the configured speed, the diagnostic dialog of the Wideband port displays the connected speed of the Wideband port.

Connections through a BCAM-ISO

When making a Wideband connection over a switched ISDN, a BCAM-ISO is required to compensate for differential delays between the individual ISDN lines.

The BCAM-ISO can only aggregate up to 16 B-channels. This limits the bandwidth of a Wideband connection through BCAM-ISO to 1024 kbit/s.

Note *The BCAM-ISO only supports mode 0, 1 and 3 operation when connecting to Wideband ports other than a Time Frame Virtual Line. A TimeFrame Virtual Line supports only modes 1 and 3.*

There are two methods of connecting a Wideband port through a BCAM-ISO:

- Manually connect the Wideband port to the BCAM-ISO. The BCAM-ISO is then configured to make the connection to the remote device.
- Switch calls from the Wideband port directly to its destination through the BCAM-ISO.

Resynchronisation

Due to the resynchronisation performed by the network, the BCAM-ISO can experience synchronisation errors. BCAM-ISO mode 3 is capable of detecting and automatically resynchronises if a network resynchronisation happens.

Mode 0 and 1, however, rely on the end-device to detect and handle network resynchronisation. If a network resynchronisation is detected, the end-device should disconnect and, if applicable, reconnect the call.

When operating in modes 0 and 1, end-devices should provide an end-to-end protocol that monitors the link sanity. The end-to-end protocol provides a means of detecting and handling network resynchronisation.

Manual connection to BCAM-ISO

The following point should be noted when configuring manual connections between Wideband ports and BCAM-ISOs:

- All manual connections, except for Packet Ports and TimeFrame, must be made as 'Internal' connections, not external.

Switch calls through BCAM-ISO

The following points should be noted when configuring Wideband ports to make switched calls via BCAM-ISOs:

- Set Wideband port's **Outgoing TE** to **BCAM-ISO**
- Receiving device Lineset setup:
 - Create a lineset on the destination device for the destination Wideband port
 - Add all the BCAM-ISO interface 0 lines to this lineset
 - Also add all of the lines from the Wideband port to the same lineset.

Notes *The Wideband port lines must come **after** the BCAM-ISO lines in the lineset.*

Frame Relay PVC management

Frame Relay PVC management provides a means to verify the integrity of a Frame Relay link by allowing you to detect changes in the state of each of the link's Permanent Virtual Circuits (PVC). It also allows you to detect the failure of a PVC and recover it. Frame Relay PVC management is only implemented on the RM's Packet Ports.

The PVC management procedures include:

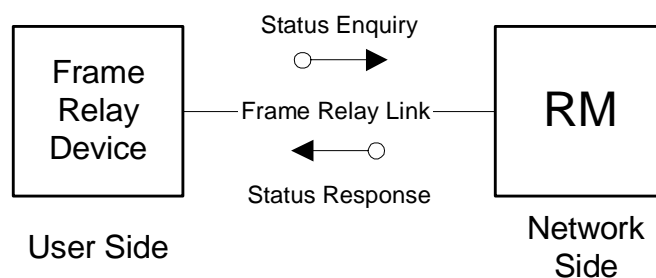
- notification of the addition of a PVC
- detection of the deletion of a PVC
- notification of the availability (active) or unavailability (inactive) state of a configured PVC. *Inactive* means that the PVC is configured but is not available to be used. *Active* means that the PVC is available to be used
- link integrity verification.

The RM supports network-side PVC management procedures with the ITU-T Q.933, Annex A and ANSI T1.617a, Annex D protocols on Packet Ports that have been configured to carry data in transparent (Frame Relay) mode. These procedures allow the RM to connect to frame relay devices that operate user-side PVC management procedures.

Note *Network-side PVC management procedures must be enabled on a Packet Port when - and only when - user-side management procedures are enabled on a connecting Frame Relay device. Some Frame Relay-capable devices that operate user-side procedures, such as the IRM, do not allow them to be disabled. Consequently, the RM must always have network-side procedures enabled while connected via a Frame Relay link to such a device.*

Polling

During PVC management, the user side periodically requests the network side to verify the integrity of the link and to report on the status of each of the link's PVCs; the network side responds in turn. This cycle of request and response is called the polling cycle, and the interval between subsequent status requests is called the polling interval. The following diagram illustrates the polling feature.



The tables below define the parameters that control the polling cycles. Table 1 defines the parameters in the user-side equipment, and Table 2 defines the network-side parameters that may be configured for a transparent-mode Packet Port. The default parameter values given should be suitable for most circumstances. Varying these parameters will alter the sensitivity of the link to errors and changes in state.

Parameter	Range	Default	Description
Polling interval	5-30	10	Specifies the interval, in seconds, between successive status requests
Full status polling counter	1-255	6	Specifies how often, in terms of polling intervals, a full report of the status of all link PVCs is requested. For example, with the default value a full status report will be requested on every 6th status request.
Error threshold	1-10	3	The number of errors detected before declaring the interface down. Note that this should be less than or equal to the monitored events count.
Monitored events count	1-10	4	The number of status polling intervals over which the error threshold is counted. Note that if this is set to a value much less than the full status polling counter then the link could go in and out of error condition without the user equipment or network being notified.

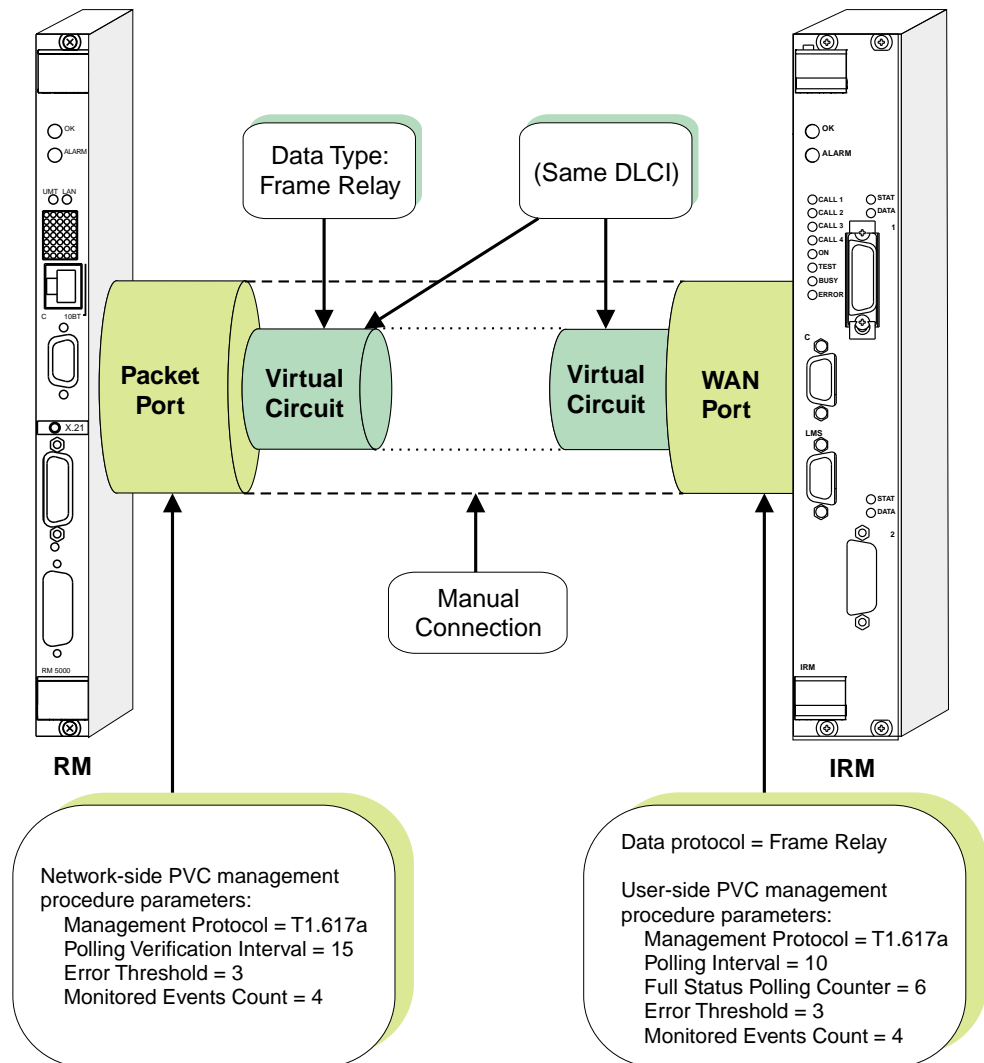
Table 1 – User-Side PVC Management Parameters

Parameter	Range	Default	Description
Polling verification interval	5-30	15	Specifies the interval, in seconds, to wait for a status enquiry message before recording an error. Note: this should be greater than the polling interval.
Error threshold	1-10	3	The number of errors detected before declaring the interface down. Note that this should be less than or equal to the monitored events count.
Monitored events count	1-10	4	The number of status polling intervals over which the error threshold is counted. Note that if this is set to a value much less than the full status polling counter then the link could go in and out of error condition without the user equipment or network being notified.

Table 2 – RM (Network-Side) PVC Management Parameters

Example – Linking from an IRM to an RM Packet Port

This example describes the configuration of a Frame Relay connection between an RM and an IRM in the same chassis. The following diagram summarises this connection.



Configuration

The configuration process has the following steps:

- 1 Configure an IRM WAN port to use the Frame Relay protocol. Set the protocol parameters listed in Table 1 for the user-side PVC management procedures. Use the T1.617a protocol and the default parameter values listed in the table.
- 2 Add a Virtual Circuit (also referred to as a 'channel') to the chosen IRM WAN port.
- 3 Configure a Packet Port on the RM. Specify the same speed as the chosen IRM WAN port. Enable network-side PVC management procedures using the T1.617a protocol, and the default parameter values listed in Table 2.

-
- 4 Configure a Virtual Circuit on the chosen Packet Port, with a DLCI to match the Virtual Circuit configured on the IRM WAN port in step 2. Set the following Virtual Circuit data priorities:

High:Frame Relay ('FR')

Medium:(none)

Low:(none)

- 5 Create a manual connection between the chosen IRM WAN port and the Packet Port.

The link between the IRM WAN port and the RM Packet Port should now be ready to carry Frame Relay-encapsulated data. Procedures for establishing data routes to the RM Packet Port are covered on page 3-25. Configuration of IRM WAN ports is dealt with in the IRM module configuration on page 4-215.

Repeat the above steps for each Virtual Circuit to added between the IRM and the Packet Port.

Note *When enabling PVC management on a Packet Port, each of the port's PVCs must be configured to carry only Frame Relay-encapsulated data by setting the data type of fr.*

Linesets

A lineset is one or more lines which share a number or range of numbers. These lines are referred to as members of a lineset. A lineset can consist of up to 220 lines and any combination of line types.

You can configure multiple linesets with an overall total of up to 1000 members per device.

Note *The total number of linesets available for the RM and CM is shown in Appendix I.*

Linesets are used to direct incoming calls: they have no effect on outgoing calls. They are particularly useful for directing incoming calls into a number of PABX ports which share a common function.

To create a lineset, you define one or more lines as members. This means that a single number, or a range of numbers apply to all the lines in the set. You can attach:

- a single number to several lines
- a range of numbers to several lines
- a range of numbers to a single line.

It is possible, but not very useful, to define a lineset with a single number and a single line. A single line can also be a member of more than one lineset. This is useful if you need to attach two or more different ranges of numbers to the same lines.

Lines in a lineset may have individual line numbers as well as the lineset numbers, so that they can also be called directly.

Lineset members may be identical, so that if one is busy another can be chosen for an incoming call. Alternatively, they may all be of different types, with a single number. This means that different types of calls are accepted by the appropriate type of line. For example, a fax call is accepted on a fax line and a telephone call is accepted on a telephone line.

Linesets may include TimeFrame Virtual Lines, but **not** Subrate, non-Subrate or broadcast Virtual Lines. For more information see *Virtual Lines* on page 3-50.

Matching linesets

When a call establishment message is received from the ISDN, any linesets which have been defined are examined for a match first, before individual lines. If a lineset which matches the called party number is found, individual lines are not examined, even if the call is not accepted by the lineset.

If a lineset is found which matches the number in the call establishment message, the individual members of the lineset are examined until one which meets all the call acceptance criteria is found. When a match is found, the call proceeds as if it were specifically for that line.

If no suitable line is found in the lineset, a message is sent to the network indicating that the call has been rejected. This will reflect the actual cause of the rejection as closely as possible. For example, a lineset could consist of a fax line and a telephone line. If a setup for a fax call is received while the fax is busy and the telephone is idle, the call will be rejected. The cause message sent would be **USER BUSY**, not **INCOMPATIBLE DESTINATION**.

The lines in the lineset are examined in the same order in which they were entered during configuration. The search can be conducted by:

- **Linear hunting**

The search always starts at the top of the Associated Lines List (the Associated Lines List contains all lines which are members of a lineset). The call is directed to the first idle member on the list. If no idle members are found, the call is rejected with a **BUSY** indication.

- **Cyclic hunting**

The search always starts one member further down the Associated Lines list than the previous call, regardless of whether that call was successful. The call is directed to the first idle member encountered. Once the call has been serviced, the list is rotated so that the next member is placed at the top of the list. This ensures that calls are distributed as evenly as possible amongst members, and that lines that are faulty but not blocked will not cause the service to fail continuously. If no idle members are found, the call is released with a **BUSY** indication.

Notes

- *A new lineset is automatically created when you configure the minimum and maximum numbers for an IPMN or a BRMN. A lineset for an IPMN contains 30 B-channels and one D-channel while a lineset for an BRMN contains two B-channels and one D-channel.*
 - *A new lineset is automatically created when you enter the line number for a BCAM or BCAM-ISO. This lineset always contains the eight channels of the BCAM, or sixteen channels of a BCAM-ISO, regardless of the line speed used.*
 - *In both the above cases, the lineset mechanism is used by the Control Module or Resource Manager to effect the B-channel aggregation of the ISDN NT function. You should not change these linesets.*
-

Configuration

To configure a lineset, choose **Linesets** from the Edit Menu in OmniVision.

Parameters

Minimum	Enter the minimum number in the range, or the only number.
Maximum	Enter the maximum number in the range.

Notes

- *The lineset line number format is a full local number, as received in the called party identification on incoming calls.*
 - *The base number is not used in the calculation of lineset numbers. This is different from the line numbers of individual lines. Refer to **Line Numbers** on page 2-15 of the Parameters section for further information.*
-

Reverse IPMN order	Select this if you want IPMN lines to be listed in reverse order. See <i>Configuring IPMN</i> on page 4-6 for more information.
---------------------------	---

Attributes for destination type

Select from **Forced TE resubmission on match**, **Local**, **Non-local** or **Custom**. The custom selection allows you to edit the following parameters:

Call delivery and line member attributes

Select from the following attributes:

Linear Hunting	Select this option to direct all calls to the first idle member on the Associated Lines list. See above for a complete explanation of this option. This is the default setting for Non-local attributes.
Cyclic Hunting	Select this option to evenly distribute calls amongst members of the Associated Lines list. See above for a complete explanation of this option. This is the default setting for the Local attributes.

Allow call to be sent back on originating interface if it is a member of this lineset

This is automatically enabled when you choose **Local** attributes.

Match on exact length	Enable if you require the number in an incoming call to be matched exactly to the line member rather than to the lineset.
------------------------------	---

Action when call cannot be delivered to any line of the lineset

Select from the following attributes:

Allow TE resubmission when call not originating from this device node fails

The default setting is 'Disabled'.

Return cause of "No circuit/channel available" to allow fallback TE &NT resubmission

This is automatically enabled when you choose **Non-local** attributes.

Do not allow TE resubmission for any cause value

This is automatically enabled when you choose **Local** attributes.



Reference Master Priorities

The Reference Master provides a 2 Mbit/s clock to the backplane which is sourced from the network. Modules that can be chosen as the Reference Master include ISDN PRA TE, ISDN PRA NT, E1TE, E1NT and ISDN BRA TE.

Default priorities

You can change the priority for each of the modules, set each one to the default individually, or set all modules to the defaults shown below.

Australian defaults

- 1 ISDN PRA TE
- 2 ISDN PRA NT
- 3 E1TE
- 4 ISDN BRA TE
- 5 E1NT/DTM
- 6 HSDM
- 7 COMBO V.5.00 and above

ROW defaults

- 1 ISDN PRA TE
- 2 ISDN BRA TE
- 3 ISDN PRA NT
- 4 E1TE
- 5 E1NT/DTM
- 6 HSDM
- 7 COMBO V.5.00 and above

Virtual Lines

A Virtual Line (VL) is a named group of lines which can be treated as one entity for call control purposes. It can have the same call control parameters, such as semipermanent label, line number, Hotline number, and so on, as a physical line.

A Virtual Line has no physical attributes itself: it is defined by its members. However, its member's call control parameters are not used by the Virtual Line when initiating or receiving calls.

The four different types of Virtual Line are:

- TimeFrame Virtual Line
- Subrate Multiplexed Virtual Line
- Non Subrate Multiplexed Virtual Line
- Broadcast Virtual Line.

The parameters available to you when creating Virtual Lines are described on page 3-56. For detailed instructions on setting up Virtual Lines refer to the *OmniVision User Manual* or *OmniVision Help*. For some typical configuration settings refer to the *Subrate Virtual Line applications* on page 5-24.

Note *The total number of lines involved in Virtual Lines cannot exceed 400 per device.*

TimeFrame Virtual Line

TimeFrame Virtual Lines allow a packet data stream and compressed voice calls to share a single bearer channel. The packet data bandwidth is dynamically adjusted, depending on the number of voice calls using the bearer. Voice calls are compressed using 8kbit/s MP-MLQ voice compression. When a voice call is added, the packet data bandwidth is reduced by 8kbit/s. Similarly, when a voice call is removed, the data bandwidth is increased by 8kbit/s. This dynamic adjustment of the packet data bandwidth maximises the use of the bearer circuit.

Note *TimeFrame Virtual Lines do not support 16 kbit/s MP-MLQ voice compression. They do support compression on the LDCM and MFCM. Refer to page 4-189 and page 4-193 for details on using TimeFrame with LDCMs and MFCMs respectively.*

Call establishment methods

A TimeFrame Virtual Line is the entity that connects to the bearer channel. It can be configured to:

- establish a switched call on demand;
- establish a long-held switched call (XSPC), or;
- use a permanent connection (manual connection or semipermanent connection).

In the case of demand-established calls, the TimeFrame Virtual Line maintains the call whenever there is packet data present, or voice calls are using the bearer. An activity timer shuts down a call if there is no data activity for a configured period of time. The number dialled for demand-established calls or XSPC is determined by the TimeFrame Virtual Line Hotline number.

For XSPC calls, the calling end must have XSPC enabled in Call Establishment modes. Similarly, the called end must have **Auto Answer** enabled.

Note *For demand-established TimeFrame Virtual Lines, do not enable **Auto Answer** at either the called or calling end. Enabling **Auto Answer** will prevent the **Activity Timer** from operating correctly and therefore the TimeFrame Virtual Line will not be shut down automatically.*

When the TimeFrame Virtual Line contains more than one line (its data rate is greater than 64 kbit/s) there are several points to note about the use of the **Port Number**, **Subaddress**, **Hotline Number** and **Hotline Subaddress** fields.

Port Number with no Subaddress

If you enter a number in the **Port Number**, and the **Subaddress** has no number, the RM automatically increments the Port Number for each line. For example, on a 192 kbit/s TimeFrame Virtual Line with a Port Number of 93640080, the Port Number for Line 0 is 93640080, for Line 1 is 93640081, and for Line 2 is 93640082. This ensures that each line has a unique number.

Port Number with a Subaddress

If you enter a number in the **Port Number** and **Subaddress** fields, the RM does not increment the Port Number for each Line. Instead, it increments the Subaddress for each line. For example, on a 192 kbit/s TimeFrame Virtual Line with a Port Number of 93640080 and Subaddress 10, each line has the following values:

Line	Port Number	Subaddress
0	93640080	10
1	93640080	11
2	93640080	12

This ensures that each line has a unique number.

Hotline Number with a Hotline Subaddress

If you enter a **Hotline Number** and **Hotline Subaddress**, the RM automatically copies the Hotline Number to each line in the TimeFrame Virtual Line, and increments the Hotline Subaddress for each line. For example, on a 192 kbit/s TimeFrame Virtual Line with a Hotline Number of 93900080 and Subaddress 10, each line has the following values:

Line	Hotline Number	Hotline Subaddress
0	93900080	10
1	93900080	11
2	93900080	12

This ensures that each line dials a unique number when establishing a call.

Hotline Number with no Hotline Subaddress

If you enter a **Hotline Number** and no **Subaddress**, you must use the **Don't Increment Hotline No.** option to control the operation of the automatic incrementing of the Hotline Number.

If the option is disabled, the Hotline Number is incremented for each line in the TimeFrame Virtual Line. If the option is enabled, the Hotline Number is **not** incremented, but is instead copied to each line.

The following table shows how the Hotline Number of a 192 kbit/s TimeFrame Virtual Line is affected by enabling or disabling the **Don't Increment Hotline No.** option:

Line	Enable Don't Increment Hotline Number	Disable Don't Increment Hotline Number
0	93900080	93900080
1	93900080	93900081
2	93900080	93900081

When configuring a TimeFrame Virtual Line with multiple lines, the two best methods for setting the numbering are:

- 1 Enter the **Port Numbers** and **Subaddress** with matching **Hotline Numbers** and **Subaddresses** at the other end
- 2 Enter a **Port Number** and no **Subaddress**, and then enter a matching **Hotline Number** at the other end with **Don't Increment Hotline No.** disabled.

For an example of how this is used with BCAM-ISO, refer to *TimeFrame Applications* on page 5-3.

TimeFrame resources

TimeFrame Resources are the end points used to terminate and originate voice calls that are to be switched onto or off a TimeFrame Virtual Line. They require no configuration except that they are added to linesets to receive voice calls onto a TimeFrame Virtual Line. TimeFrame Resources are allocated individually. There must be at least one TimeFrame Resource allocated for voice to operate over a TimeFrame Virtual Line.

Refer to *Appendix I* for details on the maximum number of TimeFrame Resources in a system.

Note *The number of TimeFrame Resources allocated need only correspond to the total number required across all TimeFrame VLS. For example, if four TimeFrame VLS are configured and each was to carry a maximum of three voice calls, then only 12 TimeFrame Resources need be allocated. Doing this will reduce the number of lines used within the device, thereby allowing them to be used in other applications.*

Packet Channel

The packet data stream consists of a combination of end-to-end signalling data for the voice calls, IP router data and HDLC data. This data is switched through the Fast Packet Switch to the portion of the bearer channel not being used for voice calls.

Set the maximum data rate of the TimeFrame Virtual Line using the Max Port Speed parameter. This can be up to 1472 kbit/s. Refer to *TimeFrame limitations* on page 3-20 for limitations on data rates.

You can configure the minimum data bandwidth for a TimeFrame Virtual Line to ensure that there is always a guaranteed portion allocated for sending data. Refer to *Call Establishment Modes* on page 2-11.

If a demand-established TimeFrame Virtual line is initiated by data passing through the Packet Switch, and the call fails, the call is retried at intervals of 15 to 20 seconds. During these intervals, the Packet Switch discards any data destined for the channel that is being established.

Matching a voice call to a TimeFrame Virtual Line

The TimeFrame Virtual Line selected for a particular voice call is determined by matching the called party number with the Hotline number of a TimeFrame Virtual Line. In the case where there is a match with more than one TimeFrame Virtual Line, the bearer that has the minimum free capacity to receive voice calls is selected. Since the numbers dialled for voice calls are typically PABX extension numbers, the numbers must be translated to select the required TimeFrame Virtual Line. The number translation is achieved using Number Checking to modify the called party number to that of the Hotline Number of the TimeFrame Virtual Line on which the call is to be sent. In the case of a Wideband TimeFrame Virtual Line, the called party number is translated, using Number Checking, to the Hotline Number of the first line of the TimeFrame Virtual Line on which the call is sent. The dialled number is passed to the remote end in the end-to-end signalling data for the call, and presented as the called party number at the remote end.

SAM channels

A SAM channel can be manually connected to an RM's Packet Port, making it a member of a TimeFrame Virtual Line. This is made possible by using the RM's 7E data handling. All on-board RM SAM channels can be used in this way. The SAM channel cannot be a member of a demand-established TimeFrame VL, as it sends data continuously to ensure the link remains active. The Packet Port would interpret this data as the link being busy, and would not disconnect the TimeFrame VL.

If a SAM channel is made a member of a TimeFrame VL, the Virtual Line must be an XSPC, SPC or manually connected call.

For more information refer to *TimeFrame* on page 3-19. For examples of configuring TimeFrame Virtual Lines see page 5-3.

Transit switching

TimeFrame Virtual Lines support the transit switching of voice calls onto other TimeFrame Virtual Lines and Subrate Virtual Lines. VCMs and SRMMs cannot be used in the node that is carrying out the transit switching as they do not support TimeFrame. The switching is however supported over Subrate switches in modules like the LDCM, MFCM, COMBO and HSDM.

Subrate Multiplexed Virtual Line



A Subrate Multiplexed Virtual Line consists of several lines (compressed voice, Subrate data or a mixture of both) that are subchannel multiplexed in increments of 8 kbit/s into one 64 kbit/s channel. For example, a Subrate Multiplexed Virtual Line could consist of four EMM lines each compressed to 16 kbit/s. Refer to *Appendix A — Data rates for Subrate multiplexing* for details of data rates that can be multiplexed.

A Subrate Multiplexed Virtual Line is used to optimise the capacity of the required network bandwidth while maintaining end-to-end functionality. It is used in point-to-point applications where multiple services are required between the two ends.

A Subrate Multiplexed Virtual Line has most of the properties of a normal line. For example, it has a line number, can make and receive calls and can be configured. Manual, Hotline and semipermanent connections (where available) are possible, providing both ends of the connection have the same subchannel allocation for members. For example, a 16-16-32 kbit/s Virtual Line is not compatible with a 32-16-16 or a 16-32-16 kbit/s Virtual Line.

Virtual line members cannot make individual calls: they can only participate in a call made to or from the Virtual Line. However, single members may cause the Virtual Line to initiate a switched call in the case of demand-established Virtual Lines.

Note *In the case of semipermanent connections, the corresponding members at either end are connected as if each had a dedicated semipermanent connection.*

Non Subrate Virtual Line



A Non Subrate Virtual Line is an association of one or more lines per device. These can be any type of line, including other Virtual Lines.

Notes *A Non Subrate Virtual Line can have another Non Subrate Virtual Line as a member. This is a useful feature, but you should ensure that a Non Subrate Virtual Line is not defined as a member of itself. OmniVision does not check for this.*

Each Non Subrate Virtual Line can have a maximum of 20 members. If you require more than 20 members, make a second Non Subrate Virtual Line containing the extra members, and make it a member of the first Subrate Virtual Line.

A Non Subrate Virtual Line does not, in itself, have any line characteristics such as bearer capability, line number and so on, as it is not used to independently initiate or terminate calls. Its purpose is to group together other lines for the forwarding of call control bits from OmniVision.

You allocate lines to a Non Subrate Virtual Line in the same way as you would allocate them to a Subrate Multiplexed Virtual Line. However, more than eight members are allowed and you do not need to allocate Subrate bits as they are not required by a Non Subrate Virtual Line.

Membership of a Non Subrate Virtual Line has no effect on the ability of a line to make and receive calls. Calls can be made and received by a member of a Non Subrate Virtual Line without any effect on the other members of the Non Subrate Virtual Line. Therefore, you configure members of a Non Subrate Virtual Line in the same way as you would if their association did not exist.

When a call request is made to the Non Subrate Virtual Line by one of its members, it establishes multiple calls associated with all of its members. This means that multiple calls and connections can be executed with a single request.

Wideband Virtual Line



A Wideband Virtual Line exhibits all the same characteristics as a Non Subrate Virtual Line, and is configured in the same way. The main difference is that a Wideband Virtual Line can additionally support lines that are members of a Wideband Packet Port and TimeFrame Virtual Line. See *Packet Port* on page 3-34 and *TimeFrame Virtual Line* on page 3-50 for more information.

Note *The RM does not support Wideband Virtual Line calls using an ISDN that requires the BCAM-ISO to provide channel aggregation.*



Broadcast Virtual Line

A Broadcast Virtual Line is similar to a Subrate Virtual Line in that it can make and receive calls, but it has only one member.

The function of a Broadcast Virtual Line is to generate and/or receive calls on behalf of the physical line. When a Broadcast Virtual Line makes or receives a call, the connection is always half duplex and towards the network.

A Broadcast Virtual Line can only maintain one active call at a time. However, a physical line can be defined as a member of a number of Broadcast Virtual Lines, each of which can make a call to the destination represented by its Hotline number. To initiate a true Broadcast call, the individual Broadcast Virtual Lines should be defined as members of a Non Subrate Virtual Line. A number of calls can then be generated from a single physical line with a single command.

Note *A Broadcast call provides unidirectional data transmission from the originating Virtual Line to the various destinations.*

Parameters

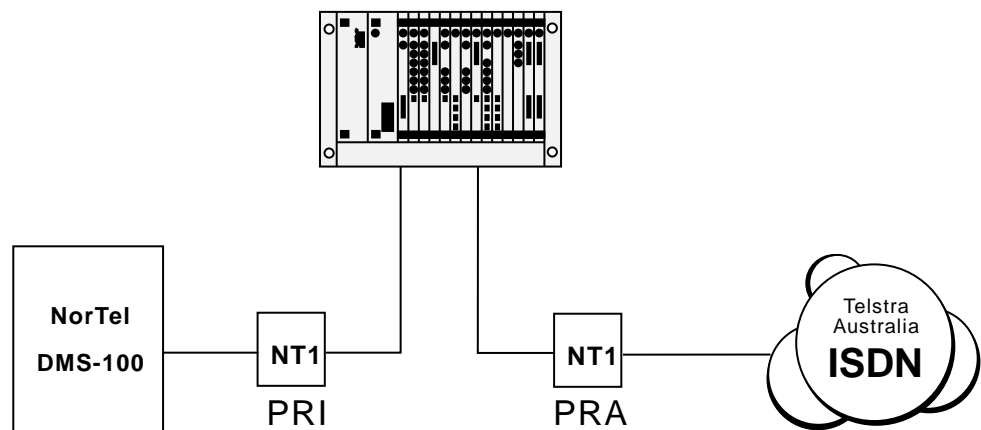
Virtual Line Name	Enter a unique descriptive name, up to eight characters in length.
Number	Displays the Virtual Line number.
Type	Displays the type of Virtual Line, that is TimeFrame, Subrate, non-Subrate or broadcast.
Members	Displays the members and where appropriate, Subrate channel allocation of the selected Virtual Line.
Available Lines list	Displays the available lines. Select the lines you want to add to the Virtual Line.
Virtual Lines list	Displays names of all Virtual Lines set up.
Virtual Line Members	Lists the members of the Virtual Line and, in the case of a Subrate Virtual Line, the Subrate channel allocation.
Config	Select to configure the selected member, line or port.
Delete	Select to delete the selected line from the Virtual Line members list.
Add	Select to add the selected line to the Virtual Line members list.
Setup	Select this button to configure the line characteristics of the Subrate Virtual Line
Line	Select this button to configure the line characteristics of the Subrate Virtual Line, directly (without entering the member screen).
Port	Select to configure the TimeFrame Virtual Line.
New VL	Select to create a new Virtual Line.
Subrate	Select to create a Subrate Virtual Line.
Non-Subrate	Select to create a non Subrate Virtual Line.
Broadcast	Select to create a broadcast Virtual Line.
TimeFrame	Select to create a TimeFrame Virtual Line.
Remove VL	Allows you to delete the selected Virtual Line.
TimeFrame Capacity	Select the number of TimeFrame Resources required for the system.

ISDN Interworking

Introduction

ISDN Interworking provides a method of linking two ISDNs which are presented to a device via NT1 primary rate access points. Calls originating in one ISDN can be shipped through the device via an Interworking Unit to the other ISDN.

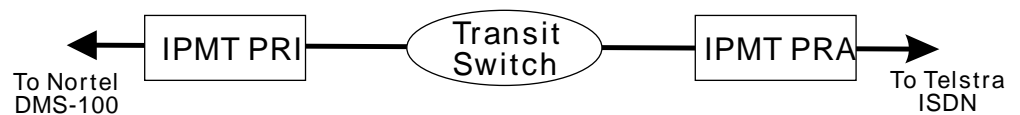
The following example assigns one ISDN as being provided by a NorTel DMS-100 exchange and the other ISDN as being provided by Telstra Australia. For the purposes of discussion, the NorTel ISDN access point is denoted as Primary Rate Interface (PRI) and the Telstra ISDN access point is denoted as Primary Rate Access (PRA).



PRA-PRI Interworking

To achieve this, the device provides an Interworking Unit through which calls are routed. The Interworking Unit can either:

- take a call from an IPMT with PRI protocol and connect it to a call on an IPMT with PRA protocol, or



- take a call from an IPMT with PRA protocol and connect it to a call on an IPMT with PRI protocol.

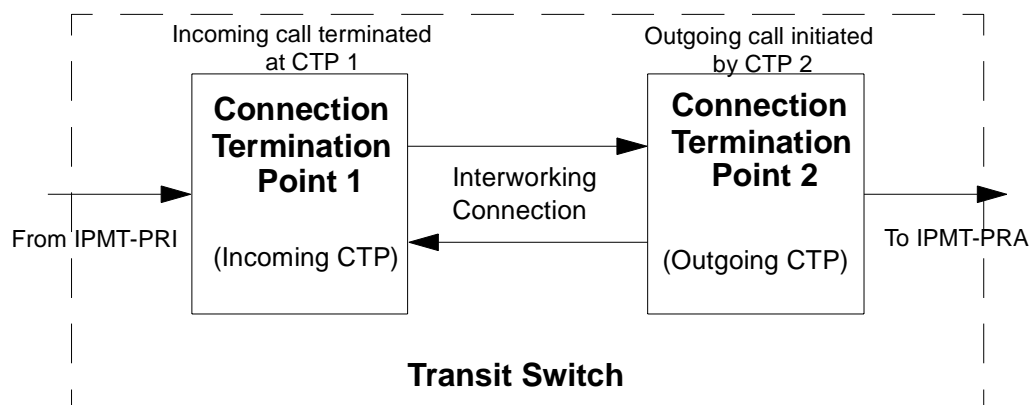
The calls are routed through an Interworking Unit, as shown in the diagram on the following page.

Call connected through an Interworking Unit

Each call made through the Interworking Unit consists of two calls:

- an incoming call from one network into the device and
- a corresponding outgoing call from the device into the other network

as shown in the following diagram.



Interworking Unit

The interworking connection between Interworking Unit lines is established as a non-volatile, bi-directional, manual connection which supports the interconnection of B-channels and signalling channels. See *Configuration* below for instructions on configuring the connection between Interworking Unit lines.

As Telstra Australia ISDN supports a wider range of call types than DMS-100, the Interworking Unit must interconnect different call types. For example, the Interworking Unit can route a switched call from DMS-100 to its destination via a Telstra Australia semipermanent connection.

Each call is independent as far as B-channel and Call Reference is concerned. Two Interworking Unit lines are required to form a connection from one ISDN interface to another. These calls only exist between the ISDN Interface and the Connection Termination Point (CTP).

Interworking Unit line

This is an element in the Control Module or Resource Manager which you can configure to terminate a connection from an ISDN interface. Each Interworking Unit line can:

- have a direct line number, or
- be part of a lineset with a range of numbers
- be used for bi-directional calls (incoming and outgoing).

To make Interworking transparent, place all Interworking Unit lines in a lineset with the number range of 0000000-9999999.

Clocking

The DMS-100 must have the same master reference clock as the Telstra Australia ISDN. You can configure a device to derive its clock from either the PRA IPMT or the PRI IPMT.

If you configure a device to derive its clock from the PRA, the DMS-100 can derive its clock from the PRI. If there is no valid clock source the device sends an alarm to OmniVision.

The device attempts to obtain its reference clock from the most reliable source available. The possible sources are the ISDN or the leased line provider.

If no reference clock source is available, the modules operate off a locally generated clock according to the following priority list:

- ISDN PRA TE modules
- ISDN PRA NT and E1M modules
- ISDN BRA TE modules.

The reference clock master changes if the current master fails to sense network clock or if a module with a higher priority senses network clock. See *Reference Master Priorities* on page 3-49 for more information.

You can exclude modules from selection as reference clock providers through OmniVision.

For more examples of the Interworking facility see page 5-34 of the *Applications* section.

Operation

Incoming calls

An incoming call is directed to the incoming Interworking Unit line which matches the called number or semipermanent label. The incoming Interworking Unit line passes the complete number (with indial digits suppressed if required) as well as the Calling Line Identification to the outgoing Interworking Unit line.

For a switched call, the dialled number is passed en-bloc across the interworking connection. For a semipermanent connection with end-to-end signalling, the digits are passed as they are received to enable overlap dialling.

Outgoing Calls

When the outgoing Interworking Unit line is notified of a call by the incoming Interworking Unit line, it makes an outgoing call based on its Call Establishment Modes settings. The call can be made to either:

- a Hotline number, if one is configured
- the number passed to the outgoing Interworking Unit line (with any initial modifications) from the incoming Interworking Unit line, or
- a combination of both, where the Hotline number is sent en-bloc followed by the initial digits.

Call types

ISDN Interworking supports the following call types.

On both the PRA and PRI sides:

- Switched call
- Long held switched call (Timelink — Australia only)
- SPC (where available)
- SPC with end-to-end signalling (IDAS) (where available)
- XSPC
- XSPC with end-to-end signalling (IDAS).

Between PRA and PRI:

A range of combinations of the above call types are possible.

Voice Compression

You can use voice compression on the PRA side as long as calls are made to, or received from, another Jtec device.

Supplementary services

The following supplementary services are available.

On the PRA side:

- Calling Line Identification.

On the PRI side:

- Calling Line Identification.

Between PRA and PRI:

- Calling Line Identification is carried transparently across the Interworking Unit.

Configuration

Introduction

The following configuration instructions are based on the application diagram on page 3-57. It assumes that the PRI is physically connected to one IPMT and the PRA to another. As a convention (which can be adapted to suit the application's physical implementation) the IPMTs are referred to as L (left) and R (right) which describes their location in the subrack and the OmniVision screen display.

The Interworking Unit lines are logically and physically located on the Control Module or Resource Manager but are connected to each of the IPMTs via the internal buses.

Control Module/Resource Manager

ISDN Interwork Unit Configuration

The incoming call's calling party number and called party number, as well as any information elements you set for the incoming Interworking Unit lines, are passed through the interwork connection to the outgoing CTP. The outgoing CTP then sends them out with the call. They take precedence over any existing outgoing call settings for the outgoing Interworking Unit lines.

Selected Interwork Unit Configuration defaults

Select Outgoing TE for this Interwork

Any.

Changing Selected Interwork Unit Configuration defaults

Select Outgoing TE for this Interwork

Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list.

Common Config

This option allows you to select from a subset of Call Establishment Modes and Bearer Capability for all Interwork lines. You can select both the left and right hand sides of the Interwork Unit. These options are explained in the *Parameters* chapter, except for the following parameters which are explained on page 3-63:

- Transparent signalling
- Prog change
- Brick Wall
- DPNSS E2E.

Configure Interwork Lines (L + R)

Line Number	Enter the line number assigned to this Connection Termination Point. For outgoing calls, or if the line is a member of a lineset, this parameter can be left blank.
Subaddress	Enter the subaddress if required. It is not mandatory to enter anything in this parameter.
Port ID	Enter the Port ID if required. It is not mandatory to enter anything in this parameter.
Port Name	Enter the Port Name if required. It is not mandatory to enter anything in this parameter.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. The following table shows the default settings.

Bearer Capability defaults

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults are enabled for incoming calls and disabled for outgoing calls. For further information about the default settings, refer to page 2-7 of the *Parameters* section of this manual.

ISDN Interface — High Layer Compatibility

These parameters are not normally used. Refer to page 2-10 of the *Parameters* section of this manual for further information.

ISDN Interface — Call Establishment Modes

The following table shows the Call Establishment Modes defaults.

Call Establishment Modes defaults

	Parameter	Default
Semipermanent (where available)	Enable	Cleared
	Label	—
Hotline	Enable	Cleared
	Number	—
	XSPC	Cleared
	Subaddress	—
DDI (Indial)	Enable	Selected
	Ignore first	0 digits
If Channel is Subrate, It's	Voice (use compander)	Selected
	Data (use Subrate switch)	Cleared
Signalling	National Significant Remote Number	—
	End-to-End	Cleared
	E2E service tones	Cleared
Call Control	Auto Answer	Cleared
	Callback Orig (RM only)	Cleared
	Callback Dest (RM only)	Cleared
	CBack Hotline (RM only)	Cleared
Advanced	Transparent signalling	Cleared
	Prog change	Cleared
	Brick Wall	Cleared

Changing Call Establishment Modes defaults

The settings for these parameters control the way in which calls are made. You can change the following settings to meet your requirements.

Semipermanent (where available)

Enter the semipermanent number in the **Label** parameter and select **Enable**.

Hotline Enter the Hotline number and the subaddress number, if required, and select **Enable**. Outgoing calls using this outgoing Interworking Unit line are directed to the Hotline number.

You can also select **XSPC** to enable a Switched Semipermanent Connection. This is only available if Hotline is enabled.

Note When a RHS Interworking Unit has **Transparent Signalling** enabled, the number dialled into the IWU is **not** prefixed with the **Hotline** number of the LHS of the Interworking Unit. If you require number prefixing, use the **CLI/Number Checking** feature.

DDI (Indial) Select **Enable** and enter the number of digits to be ignored. For example, if the call is for extension 23 of the telephone number 8096923, the number of digits to ignore is five. The Indial setting for the incoming Interworking Unit line is passed across the interwork connection and used by the outgoing Interworking Unit line for the outgoing call.

If Channel is Subrate, It's Select **data** if the channel is being used for framed data rather than A-law voice.

Signalling Enter the National Significant Remote Number in the number parameter for end-to-end signalling applications, if required.

End-to-End Enable this where the line is to be used in equipment tie line applications.

E2E service tones Enable this to provide service tones from the incoming IWU of this VX. This is useful where the network or the originating device cannot generate tones. When enabled, the incoming IWU detects a call and provides the corresponding service tones.

Auto Answer Select this to connect the incoming call to the line as soon as it is received, irrespective of the signalling state of the connected equipment.

When a tie line has end-to-end enabled, select **Auto Answer** to connect the end-to-end tie line call request rather than returning the connected equipment's signalling indications.

RM

Callback Orig Used for the CallBack feature. You should enable this for the end of the network that is to accept all call charging. For more details see the *CallBack* feature in *Interworking applications* on page 5-39.

RM

Callback Dest Used for the CallBack feature. You should enable this at the end of the network that is not to receive call charging. For more details see the *CallBack* feature in *Interworking applications* on page 5-39.

RM

CBack Hotline Used for the CallBack feature. You can enable this to force the CallBack Orig end to use the number entered in the Hotline Number of this dialog for CallBack. This is instead of using the CLI of the CallBack Dest end. For more details see the *CallBack* feature in *Interworking applications* on page 5-39.

Transparent signalling Select this option to have full Q.931 messages passed end-to-end within the user information packets. Note that 'End to End Signalling' must be selected for this to work.

Prog change Select if you want Alerting messages changed to Progress messages.

This can be used when a standard telephone call is diverted to a mobile network. When the mobile phone is out of the operation area or is turned off, a verbal Alerting Message is sent. This Alerting message is read as a ring tone by the ISDN. When Prog change is enabled, the Alerting message is changed to a verbal Progress message.

Brick Wall Enable this to allow the Bearer Capability, High Layer Compatibility and Low Layer Compatibility information elements of the incoming call to be replaced with those set up in the outgoing IWU. The Called Party Number and subaddress are replaced if they are set on the outgoing IWU. The Calling Party Number and subaddress are not changed.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E164
	Hotline	E164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI lists	—

Changing ISDN Line Controls defaults

AOC (where available) Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via the Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID Do not select **CLIP** (CLI Presentation).

If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

CLIVE List Assignment Select this option to apply a CLI list to this line. See page 3-80 for more information.

Numbering format Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface

ISDN Interface defaults

Teleservice None

ISDN Internal Control

ISDN Internal Control defaults

Restriction None

Changing ISDN Internal Control defaults

Restriction Select **Disable** to put the line into a fault state so that it is not used by the device.

Configuration of IPMT for Interworking

IPMT

Refer to IPMT on page 4-3 of the *Module Configuration* section of this manual for basic configuration instructions. The configuration instructions for the interworking options are shown below.

Options

- Disable T309** Select to disable T309.
- Enable CRC-4** Select to enable CRC-4. You must select this for a PRI IPMT.
- Do not use as reference clock** Select to disable reference clock.
The setting you select for this parameter depends upon the clocking requirements for each application. Refer to *Clocking* on page 3-59 for further information.

MicropleX switching

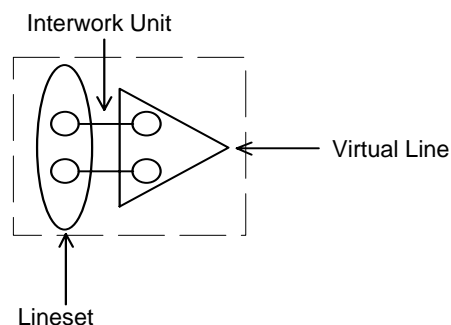
Introduction

The VX can be used to provide static subchannel multiplexing. Since no switching is performed, lines always connect to other lines as defined in a Virtual Line configuration. This configuration can be extended to provide both point-to-point and transit connections – again the connections are static.

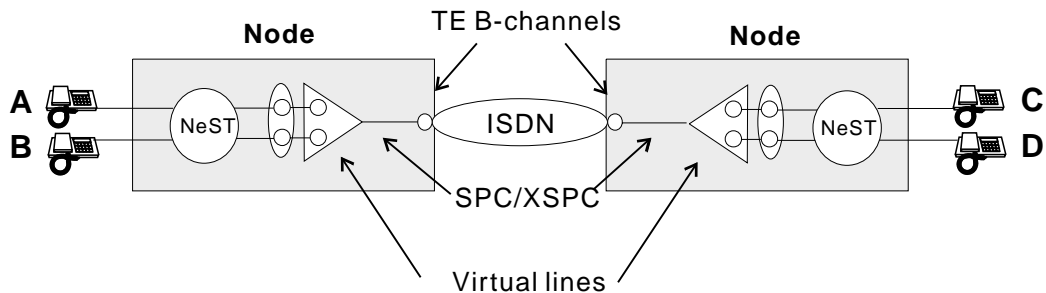
Where switching is required, the calls have to be delivered to separate switching equipment, such as a PABX, or to another device, and then back again to the originating device. The switched application therefore requires additional equipment and, in the case of voice services, adds additional decompression and compression stages.

MicropleX switching allows switching onto Virtual Lines to be performed within a single VX. A new type of subchannel line is configured which is not connected to any physical port. This line can be configured to be Subrate voice or Subrate data. In the case of voice, an LDCM or COMBO compander is used to connect the port. For data, an SRMM Subrate switch is used. This new line can have calls directed to it, using the normal call routing techniques such as matching line numbers and/or linesets.

This different form of Subrate line is obtained by combining the Interwork and the Network Simulation Task (NeST) functions. The NeST performs the number analysis and local switching for the VX. See *ISDN Interworking* on page 3-57 for details of the Interwork function. The basic configuration unit used to implement this application consists of pairs of interwork unit lines, a Subrate Virtual Line and a Lineset. This functional block is represented by the figure below:

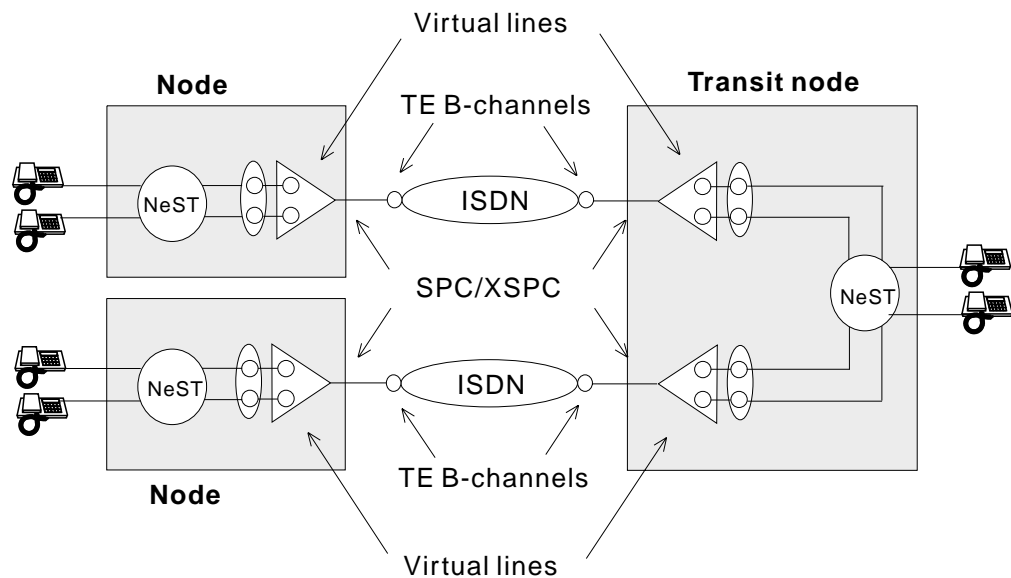


An NT line such as an ALEM, ALEM-2, COMBO ALIM or E1MN can dial a number which matches the Lineset number range of the left side of the interwork unit lines. The call is switched by the NeST. For a point-to-point application, this functional block is configured as illustrated below.



In the switched application, telephones A and B can connect to either telephones C or D.

This switching scheme can be extended past the point-to-point case using a tandem switch configuration in the transit nodes. The tandem configuration is illustrated below:



Calls entering the transit node can be switched to a destination line within the transit node or be switched through the transit node toward another node. In this application any phone can call any other one.

For more information on MicropleX Switching applications see page 5-27 of the *Applications* section.

Configuration

Introduction

The different operational modes of the Subrate circuits each require different configurations on the interwork unit lines, the Virtual Lines and possibly the Linesets. The possible operational modes are:

- tie line using XSPC
- manual connection
- demand established circuits
- tie line using SPC (where available).

Each of these operational modes can use B- or D-channel signalling.

The following sections describe the basic configuration for each of these combinations. In these configurations, it is assumed that the left side of the interwork unit is assigned to a lineset and the right side is assigned to a Virtual Line. This is not essential, but is used here for consistency. Also the configuration for a single interwork unit pair will be given, since the configuration of other interworks which are members of the Virtual Line will be identical.

Additionally, the following items must be configured for correct operation of the device:

- TE modules Group Number and National Significant Area Code
- Control Module or Resource Manager Access Number
- VX 'Minimum number in this device', 'Maximum number in this device' and 'Area code without leading zero'. See page 3-17 for details on these VX settings.

XSPC operation

Lineset configuration

Configure as Non-local linesets with the following default settings:

**Call delivery and
line member list
attributes**

Select **Linear Hunting (keep list static)**.

**Action when call
cannot be
delivered...**

Select **Return cause of 'No circuit/channel available' to allow fallback TE & NT resubmission**.

Refer to page 3-46 for further information on configuring Linesets.

Interworking Unit left side configuration

This remains at the default settings. See *ISDN Interworking* on page 3-57 for further information.

Interworking Unit right side configuration

Call establishment modes	Select SPC/Virtual Line/Manually connected . Select Indial . Select End to end signalling . Select Transparent signalling mode .
Bearer Capability	Select All 64K Digital . See <i>ISDN Interworking</i> on page 3-57 for further information.

Virtual line configuration — B-channel signalling

Hotline	Enable XSPC at one end only, if using a point-to-point connection.
Outgoing TE	Select as Preferred or Exclusive on the Virtual Line with XSPC enabled. This prevents the B-channel call from being delivered to the interwork unit.
Virtual line Hotline number	The number defined for the Virtual Line Hotline number must be entered as the line number in the remote end Virtual Line. Enter the number defined for the Virtual Line Hotline number in the Router table with a SAM B-channel selected as the primary route.
SAM B-channel	Configure as a member of the Virtual Line.
Semipermanent	Enable semipermanent for the SAM B-channel used for the B-channel Signalling.

Virtual line configuration — D-channel signalling (where available)

Hotline	Enable XSPC at one end only, if using a point-to-point connection.
Outgoing TE	Select as Preferred or Exclusive on the Virtual Line with XSPC enabled. This prevents the B-channel call from being delivered to the interwork unit.
Virtual line Hotline number	The number defined for the Virtual line Hotline number must be entered as the line number in the remote end Virtual Line.

Manual Connection operation

Lineset configuration

Configure as Non-local linesets with the following default settings:

Call delivery and line member list attributes	Select Linear Hunting (keep list static) .
Action when call cannot be delivered...	Select Return cause of 'No circuit/channel available' to allow fallback TE & NT resubmission .

Refer to page 3-46 for further information on configuring Linesets.

Interworking Unit left side configuration

This remains at the default settings. See *ISDN Interworking* on page 3-57 for further information.

Interworking Unit right side configuration

- Call establishment modes** Select **SPC/Virtual Line/Manually connected**.
Select **India**.
Select **End to end signalling**.
Select **Transparent signalling mode**.
- Bearer Capability** Select **All 64K Digital**.
See *ISDN Interworking* on page 3-57 for further information.

Virtual line configuration — B-channel signalling

- Interwork lines** Enter a user-defined SPC label for those individual interwork unit lines which are members of the Virtual Line.

Configure the **Signalling—National Significant Remote Number** for the individual interwork unit lines which are members of the Virtual Line to the IPMT or BRMT group number of the remote end TE.

Enter the number defined for the Interworking Unit **Signalling—National Significant Remote Number** in the Router table, with a SAM B-channel selected as the primary route. The Router table configuration is only required at the device with the highest Group number.
- Outgoing TE** Selection for End to End Signalling must be set to **Preferred** or **Exclusive** in the Control Module or Resource Manager dialog.
- SAM B-channel** Configure as a member of the Virtual Line.
- Semipermanent** Enable for the SAM B-channel used for the B-channel Signalling.

Virtual line configuration — D-channel signalling (where available)

- Interwork lines** Enter a user-defined SPC label for those individual interwork unit lines which are members of the Virtual Line.

Configure the **Signalling—National Significant Remote Number** for the individual interwork unit lines which are members of the Virtual Line to the IPMT or BRMT group number of the remote end TE.

Enter the number defined for the Interworking Unit **Signalling—National Significant Remote Number** in the Router table, with a SAM B-channel selected as the primary route. The Router table configuration is only required at the device with the highest Group number.
- Outgoing TE** Selection for End to End Signalling must be set to **Preferred** or **Exclusive** in the Control Module or Resource Manager dialog.

Demand Established operation

Lineset configuration

Configure as Non-local linesets with the following default settings:

- Call delivery and line member list attributes** Select **Linear Hunting (keep list static)**.
- Action when call cannot be delivered...** Select **Return cause of 'No circuit/channel available' to allow fallback TE & NT resubmission**.

Refer to page 3-46 for further information on configuring Linesets.

Interworking Unit left side configuration

This remains at the default settings. See *ISDN Interworking* on page 3-57 for further information.

Interworking Unit right side configuration

Call establishment mode Select **India**.

Bearer capability Select **All 64K Digital**.

Virtual line configuration — B-channel signalling

The following settings apply to both ends of the point-to-point connection.

Outgoing TE Select as **Preferred** or **Exclusive** on the Virtual Line. This prevents the B-channel call from being delivered to the interwork unit.

Virtual line Hotline number Enter this as the line number in the remote end Virtual Line.
Enter in the Router table with a SAM B-channel selected as the primary route.

SAM B-channel Configure as a member of the Virtual Line.

Virtual line configuration — D-channel signalling (where available)

Hotline Enable.

Outgoing TE Select as **Preferred** or **Exclusive** on the Virtual Line. This prevents the B-channel call from being delivered to the interwork unit.

Virtual line Hotline number Enter this as the line number in the remote end Virtual Line.

SPC operation (where available)

Lineset configuration

Configure as Non-local linesets with the following default settings:

Call delivery and line member list attributes Select **Linear Hunting (keep list static)**.

Action when call cannot be delivered... Select **Return cause of 'No circuit/channel available' to allow fallback TE & NT resubmission**.

Refer to page 3-46 for further information on configuring Linesets.

Interworking Unit left side configuration

This remains at the default settings. See *ISDN Interworking* on page 3-57 for further information.

Interworking Unit right side configuration

Call establishment modes	Select SPC/Virtual Line/Manually connected . Select India . Select End to end signalling . Select Transparent signalling mode .
Bearer Capability	Select All 64K Digital . See <i>ISDN Interworking</i> on page 3-57 for further information.

Virtual line configuration — B-channel signalling

Interwork lines	Enter a user-defined SPC label for those individual interwork unit lines which are members of the Virtual Line. Configure the Signalling—National Significant Remote Number for the individual interwork unit lines which are members of the Virtual Line to the IPMT or BRMT group number of the remote end TE. Enter the number defined for the Interworking Unit Signalling—National Significant Remote Number in the Router table, with a SAM B-channel selected as the primary route. The Router table configuration is only required at the device with the highest Group number.
Outgoing TE	Set End to End Signalling to Preferred or Exclusive in the Control Module or Resource Manager dialog.
SAM B-channel Semipermanent	Configure as a member of the Virtual Line. Enable for the SAM B-channel used for the B-channel Signalling.

Virtual line configuration — D-channel signalling

Interwork lines	Enter a user-defined SPC label for those individual interwork unit lines which are members of the Virtual Line. Configure the Signalling—National Significant Remote Number for the individual interwork unit lines which are members of the Virtual Line to the IPMT or BRMT group number of the remote end TE. Enter the number defined for the Interworking Unit Signalling—National Significant Remote Number in the Router table, with a SAM B-channel selected as the primary route. The Router table configuration is only required at the device with the highest Group number.
Outgoing TE	Selection for End to End Signalling must be set to Preferred or Exclusive in the Control Module or Resource Manager dialog.

Inband signalling facility

Introduction

Inband Signalling provides an alternative to the ISDN D-channel user-to-user messaging service (but it does not preclude its use). It is useful where this service is unavailable or uneconomical.

The Inband Signalling function is provided by either the Control Module (CM), Resource Manager (RM) or the Signalling Access Module (SAM), all of which can function as network nodes. Each node contains a router and communications links. The interface on the SAM is synchronous V.24 DTE at rates up to 38.4 kbit/s.

An inband connection consists of one or more communications links and transit routers. As Control Modules or Resource Managers cannot be linked together for networking purposes, you require a DLM with V.24 DCE interface or a SAM with V.24 DCE interface to build a network.

You can use one or more routers and communications links to create a network of VXs to transport network management and end-to-end tieline signalling (UDAS/IDAS) information.

For examples of how to apply Inband signalling in different scenarios see the *Applications* section page 5-43.

Routers

Each Signalling Access Module, Control Module or Resource Manager contains a router. A router provides a routing table which defines:

- an address (phone number) or range of addresses
- the port for the primary route
- the port for the secondary route.

To establish a network management or end-to-end signalling call, a path must be connected through a series of one or more routers. Each one makes a connection with the next until the call reaches its destination.

Routers establish B-channel and D-channel calls as required for each end-to-end connection. You can configure the router ports and tables to define every required path. Each path may involve several links and routers.

Router Ports

Control Module/Resource Manager

For Inband Signalling the CM supports up to 16 D-channel ports.

The RM supports one dedicated 8 kbit/s port on the backplane and up to 16 other ports configured as either 8 kbit/s backplane ports or D-channel ports.

SAM

For Inband Signalling, SAM provides:

- backplane ports which can be configured as either:
 - eight 8 kbit/s ports
 - four 8 kbit/s ports and one 64 kbit/s port
 - two 64 kbit/s ports
- one front panel port which can be either X.21, V.35 or synchronous V.24 and can be configured as either a DCE or DTE.

SDLMs

You can also use a V.24, V.35 or X.21 SDLM to provide extra front panel ports for an CM, RM or SAM router.

Connecting Router Ports

Router Ports are connected to create communications links. You can connect:

- two front panel ports using a cable
- two backplane ports (in the same chassis) using an OmniVision manual connection
- two backplane ports via the ISDN B-channel (either dedicated or Subrate multiplexed)
- a backplane port to a front panel port via an SDLM. (To do this, you connect the backplane port to the backplane of the SDLM via an OmniVision manual connection. Then use a cable to connect the SDLM front panel port to the front panel router port.)
- two D-channel ports via an ISDN D-channel.

When connecting front panel ports you must ensure that one is a DTE and the other a DCE. However, this is not necessary for a connection between two D-channel ports or backplane ports. Connected backplane ports must also be configured for the same data rates.

Call routing and connection

To establish a network management or end-to-end signalling call a path must be connected through one or more routers.

When a router receives a call setup it attempts to match the called party number with an address in its routing table. If the matching port has no physical connection to the next router, an ISDN B-channel or D-channel call is made to the Hotline number configured for the router port.

Sharing an existing connection

After a connection has been established for one call, other calls requiring the same connection can share it. Even if the other connection only requires some of an established connection the common links can be shared thereby reducing communications costs.

Backup routes

If a connection which forms part of the primary route fails, the secondary route is used. If this also fails an alarm is generated.

Signalling end-points

When Inband Signalling is used for network management the end points can be:

- OmniVision — (connected to the front panel of the Control Module or Resource Manager)
- a Network Management System (NMS) connected to any front panel port
- the Control Module or Resource Manager.

When Inband Signalling is used for end-to-end tie line signalling the end-points are ALEM, ALEM-2, ALPM, ALPM-2, COMBO ALIM, EMM or E1M lines. They are all connected to the router in the CM or RM via a router port.

End-point addressing

Every end-point has a unique address (phone number). When a router receives an incoming call setup from another router, it attempts to match the called party number with an end-point in its router table. If a match is found it connects the call to the router port for that end-point.

Networks

You can use routers and communications links to design a management/signalling network. The front panel connectors must be of the same interface type and configured correctly. That is, for each pair, one must be a DTE and the other a DCE.

You can design a number of different types of networks, including:

- Point-to-point
- Two devices connected via the ISDN
- Star.

A Signalling Access Module in one chassis can be connected with up to eight other devices.

For examples of how to apply Inband Signalling in different scenarios see the *Applications* section page 5-43.

Configuration

This section provides some basic guidelines for configuring Signalling Access Modules, the Control Module and Resource Manager for Inband Signalling.

To build a private network, you must first set up routing tables for each router.

Configuring the Control Module or Resource Manager

Signalling Access (CM/RM-SAM)

SAM Line Control

Edit Line Select the required line. Choose either the B or D button to convert one of the 16 configurable SAM channels to B- or D-channel.

Note *The CM only supports D-Channels.*

Configure

Router tables Enables you to set up an Inband Signalling network.

Note *The RM-SAM can contain up to a total of 256 entries in its router table. The number of router table entries that can point to a single SAM B-channel or D-channel is limited only by the maximum number of entries in the router table.*

Inband Signalling Routing Table

Destination Address

Min/only Enter the number or the minimum number for a range.

Max Enter the maximum number for a range.

SAM channel idle holding time

This has a range from 0 to 2550 secs. The default is 30 secs. When enabled, the channel is held open for the specified time even when no activity is detected.

Area code

Enter the Area Code without the Escape Prefix (normally the leading zero).

ISDN Interface — Low Layer Compatibility and Bearer Capability

The default values for Low Layer Compatibility are such that LLC is not applied to outgoing calls, and the Bearer Capability is set to Unrestricted Digital.

Note *If you are downloading a configuration from an earlier release, you must change these parameters to suit the latest defaults.*

ISDN Interface — Call Establishment Modes

The following table shows Call Establishment Modes default settings for each channel.

	Parameter	Default
Semipermanent (where available)	Enable	Cleared
	Label	—
Hotline	Enable	Cleared
	Number	—
	XSPC	Cleared
	Subaddress	—
Call Control	—	—

Changing Call Establishment Modes defaults

The settings for these parameters control the way in which calls are made. You can configure the following call types:

Semipermanent (where available)

Enter the semipermanent number in the **Label** parameter and select **Enable**. You should also clear the Hotline checkbox.

Hotline

Enter the Hotline number for the channel and enter a subaddress, if required. Select **Enable**. You should also clear the Semipermanent checkbox.

Select **XSPC** to enable a switched semipermanent connection if required. You can only select this if a Hotline is enabled.

Configuring a SAM

SAM Line Control

Edit Line Select the required line.

Configure

Router tables Enables you to set up an Inband Signalling network.

Configure Inband Signalling Routing Table

Min/only Enter the number or the minimum number for a range.

Max Enter the maximum number for a range.

ISDN Interface — Call Establishment Modes

The following table shows Call Establishment Modes default settings for each channel.

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	Subaddress	—
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Semipermanent (where available)

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Hotline

Enter the Hotline number for the channel and enter a subaddress, if required. Select **Enable**. You should also clear the Semipermanent checkbox.

Select **XSPC** to enable a switched semipermanent connection if required. You can only select this if a Hotline is enabled.

ISDN Internal Control

Restriction	None.
Operation (Front panel only)	Select DCE or DTE.
Use Q.931 protocol (D-channel only)	This is the default.
Use X.25 protocol (D-channel only)	Select to use the X.25 protocol.
Send Acknowledge message (D-channel only)	When enabled every message sent on this line will be responded to with an acknowledge message. This should normally be disabled to obtain a higher data throughput. Both ends of the link should have the same setting.
Enable protocol discriminator (D-channel only)	When enabled a protocol discriminator with the value of 00 is added to all messages. This is required when calls are made via networks which check the protocol discriminator value. Both ends of the link should have the same setting.
Outgoing TE	Select the required TE. If you select Preferred or Exclusive , you must then select the TE from the displayed list. This setting affects all the module's lines.

Calling Line Identification and Verification Enhancement

Calling Line Identification and Verification Enhancement (CLIVE) enables the device to:

- accept or reject calls based on the CLI/Number
- modify accepted calling and/or called party identification.

CLI/number checking

Calls are accepted or rejected on the basis of the calling or called party's number which is provided by the ISDN with each call. This is checked against CLI/Number Checking lists which you set up.

CLI/Number checking is available for incoming and outgoing calls from the ISDN.

If the call is accepted the number can be modified in a variety of ways. For example, it can be replaced with another number or it can have a prefix or suffix added to it.

CLI/Number checking is always active. Its effect on the operation of the device is mediated by the **Available CLI/Number** list.

Its main features are:

- A maximum of 254 CLI/Number Checking Lists
- A List can be applied to all Lines
- A line can have up to 20 CLI/Number Checking Lists applied to it.

Calling Party Number

The Calling Party Number is provided by the ISDN on incoming calls. As it is a network specific application, its format and provision depends upon the network. In Australia, the Calling Party Number is provided on all ISDN calls unless the caller subscribes to Calling Line Identification Restriction (CLIR) and specifically requests that the number be suppressed.

The number is verified and inserted by the exchange if the caller omits it or provides an incorrect number.

The Calling Party Number is always restricted if the call does not originate in the ISDN, unless the ISDN service is a special provision.

CLI/Number lists

A CLI/Number list contains sets of Calling Party Numbers against which incoming calls are checked.

You can create and manipulate up to 254 CLI/Number lists and apply these lists to selected lines. There is no limit to the number of lines a list can be applied to. You can choose to accept or reject calls from numbers that match those contained in the lists.

Each CLI/Number list contains a particular group of calling party numbers. For example, one list may contain all the Sydney telephone numbers and another list may contain the telephone numbers of members of a special group, such as all the board members of a particular company.

CLI/Number lists may be:

- **global**
This list effects all lines in the device and is always called *Global*. You should take care when creating a global list to ensure that calls are not mistakenly rejected.
- **specific**
These lists can be applied to individual lines, individual modules or linesets. You provide a descriptive name for each list you create.

How CLI/Number checking works

CLI/Number checking is always active. Its effect on the operation of the device is determined by the contents of the global and specific CLI/Number lists and the way in which the specific lists are applied to lines. If no global or specific CLI/Number list is present, all calls are accepted.

If a global CLI/Number list is present, it is always checked for the calling party number first. If the number is found and the call is accepted, any specific CLI/Number lists are then checked, in the order in which they were applied to the line.

If the number is found in a CLI/Number list, the call will be either accepted or rejected, as specified by the list. If none of the specific CLI/Number lists contain the calling party number, the call will be rejected. If the Calling Party Number is not present, the call will be rejected by CLI/Number lists which accept a range.

Warning

You must take particular care when using CLI/Number checking to reject calls. After defining and applying the list(s) of calling party numbers to be rejected, you should define a CLI/Number list which accepts every possible calling party. Add it after the other CLI/Number lists associated with the line. If you do not do this, every call will be rejected, not just those which you have explicitly defined.

A flexible combination of selective number manipulation, selective rejection and 'accept the rest' processing can be applied to calls by sequential application of several list types.

Checking the Calling Party Number in an 'Accept Only' List

When number checking the calling party number in an 'accept only' list, all call matches against a list are accepted, whilst those with no match are checked against each of the next lists in order until all lists are checked. An accepted call will undergo designated calling or called number manipulation before forward call processing. If no match is found to any list the call will be rejected. To be able to accept calls in this case, another number checking list must be created which will accept all calls that fall within the required number range. This catch-all list is often referred to as a 'bucket', with the bucket being the last list applied to a line. The list application order is important in this case as this must be the last list applied.

Checking the Calling Party Number in a 'Reject Any' List

When number checking the Calling Party Number in a 'Reject Any' list, all call matches to a list are rejected immediately, whilst those with no match are checked against each of the next lists in order until all lists are checked. If no match is found to any list the call will be rejected. To be able to accept calls in this case, a "bucket" must be applied as the last list to a line.

Checking the Called Party Number in an 'Accept Only' List

When number checking the called party number in an 'accept only' list, all calls with matched numbers will undergo the designated calling or called number manipulation before further forward call processing. Lists are checked in order until a match is made. If the called party number is not matched to any list, the call will be forward call processed in the normal manner.

Note *A 'bucket' is not required to prevent calls from being rejected following failure to match against an 'Accept Only' list. As called party number matching is used predominantly for number changing rather than number checking, the 'always accept' default for unmatched calls is the most desirable operational characteristic for this type of list.*

Checking the Called Party Number in a 'Reject Any' List

When number checking the called party in a 'Reject Any' list, all call matches to a list are rejected immediately, whilst those with no match are checked against each of the next lists in order until all lists are checked. If no match is found to any list the call will be forward call processed in the normal manner.

Configuration

To configure CLIVE, use **Number Checking** in OmniVision. You can then:

- define CLI/Number lists
- apply them to specific lines as required
- specify any modifications to the called party number.

Parameters

CLI/Number Checking

CLI/Number List Name	Select Global or enter a unique descriptive name for the list.
ID Number	Displays the identification number for the selected CLI/Number list.
Type	Displays the list type for the selected CLI/Number list.
Numbers	Displays the number(s) and/or range(s) for the selected CLI/Number list.

CLI/Number Checking — Configure

Apply	Select to apply the selected CLI/Number list to lines, modules or linesets.
Numbers	Select to enter the number or range of numbers to be checked.
New List	Select to create a new CLI/Number list.
Remove List	Select to remove the selected CLI/Number list.

CLI/Number Checking List — How to Apply This List

Accept Only Numbers in List	This is the default. Accept it to specify that calls from numbers in this list will be accepted.
Reject Any Numbers in List	Select to specify that calls from numbers in this list will be rejected.

CLI/Number Checking List — Apply to

Calling party number	Select to apply the list to the calling party number.
Called party number	Select to apply list to called party number.

CLI/Number Checking List — What To Do With Accepted Calls

Pass on Directly	Select to pass the call on without modification.
Modify Called Number	Select to pass the call on with one of the following modifications. This is the default.
Modify Calling Number	Select to pass the call on with one of the following modifications.
Modifier	Enter the modifier number (up to 10 digits). It can be a complete number, a prefix or a suffix.
Replace With	Select to replace the called party number with the modifier number displayed.
Prefix With	Select to insert the modifier digits before the called party number as a prefix.
Suffix With	Select to insert the modifier digits after the called party number as a suffix.
Add	Select to add, numerically, the modifier to the called party number.
Subtract	Select to subtract, numerically, the modifier from the called party number.
Replace with Number	Select to replace the called party number with the calling party number.
Number Prefixed With	Select to replace the called party number with the calling party number and modifier as a prefix.
Number Suffixed With	Select to replace the called party number with the calling party number and modifier as a suffix.

Number With Added Select to replace the called party number with the calling party number numerically added to the modifier.

Number With Subtracted Select to replace the called party number with the calling party number with the modifier numerically subtracted from it.

CLI Number Checking List - Apply When Line Is

Call Destination Select to apply Number Checking at the Call Destination, that is, check the number after call processing and optionally change the Calling or Called number.

Call Source Select to apply Number Checking at the Call source, that is, check the number before processing and optionally change the Calling or Called number.

Feature Trans. Select when applying public network number mapping to DPNSS or QSIG for supplementary service handling. This prevents a problem on some networks where calls from the DPNSS or QSIG module to the PBX undergo a number change on the way out due to standard number changing.

Release 12 This is the Default. Number Checking is only carried out at the Call Destination. Release 12 and earlier software only supports this form of number checking.

Called Number = <CALLED> Displays the called party number format, resulting from the specified modification.

Min/Only Enter the number or minimum number of the range for this CLI/Number list to check. Both the Min/Only and Max field accept a wildcard character of ?. For example, to check all the numbers in a range 1090 to 1099, you could specify a Min/Only number of **109?** to check all of these numbers.

Max Enter the maximum number of the range for this CLI/Number list to check.

List Name Displays the name of the selected CLI/Number list. You can change it if required.

Add Number Select to add the number in the Min/Only field to the list of numbers checked by this CLI/Number list.

Add Range Select to add the range in the Max and Min fields to the list of numbers checked by this CLI/Number list.

Delete Select to delete the selected number or range from the list of numbers to be checked by this CLI/Number list.

Delete All Select to delete all the numbers or ranges from the list of numbers to be checked by this CLI/Number list.

Import This enables you to produce a CLI/Number list in a database or spreadsheet package and then import it into OmniVision.

Export This enables you to export a CLI/Number list from OmniVision to a database or spreadsheet package.

Accept Call when no CLI provided Select to accept calls when no CLI is provided, as well as those that match the specification.

Which party to apply to

Calling party number Select to apply list to calling party number.

Called party number Select to apply list to called party number.

Applying CLI/Number Checking

Add Select to add the selected module, line or lineset for this CLI/Number list to check.

Delete Select to delete the selected module, line or lineset from the list for this CLI/Number list to check.

Applying CLI/Number Checking — List all

- Modules** Select to display a list of all possible modules for this CLI/Number list to check.
- Lines** Select to display a list of all possible lines for this CLI/Number list to check.
- Linesets** Select to display a list of all possible linesets for this CLI/Number list to check.

CLI/Number Checking

- Add** Select to add the selected CLI/Number list to the ***This line checks*** list.
- Delete** Select to delete the selected CLI/Number list from the ***This line checks*** list.

Serial Alarm Interface

The Serial Alarm Interface provides an interface between Jtec's Network Management System, OmniVision, and another Network Management System, British Telecom's Concert. Concert complies with OSI Network Management Forum standards. It does not manage Jtec products directly but manages them through their own element manager, in Jtec's case, OmniVision.

The Serial Alarm Interface enables OmniVision to:

- report Alarm information to Concert via OmniVision's Serial Alarm Log
- send an information message to Concert to verify that OmniVision is working.

Information message to Concert

OmniVision can send an information message to Concert at regular intervals to indicate that OmniVision is still working. This is sent irrespective of whether there is Alarm information from the device. To enable this feature, follow the configuration instructions below.

Configuration

See your *OmniVision User Manual* for details on accessing these parameters.

Serial Alarms On Select this to enable the interface.

Configuration of 'hello' message

Device	Enter the Device name. It can be up to 32 alphanumeric characters. This name is displayed in the NMS_NODENAME field of the Hello Message.
Manager	Enter the Manager number. It can be up to 20 digits. This number is displayed in the NUMBER field of the Hello Message.
Enable 'hello' messages.	Select this to enable 'hello' messages.
Interval between 'hellos' (5 secs to 24 hrs)	Enter the required time (between 5 seconds and 24 hours) between 'hello' messages. The default is 10 seconds.

Configuration of SPEAC

SPEAC (System Protocol for External Access and Control) is available with the OmniVision NCS product. For more information, refer to the *OmniVision User Manual*.

Phone Home

Note *The Phone Home feature is not available to users of OmniVision NDM and NDM-5 as it requires the use of the OmniVision Map.*

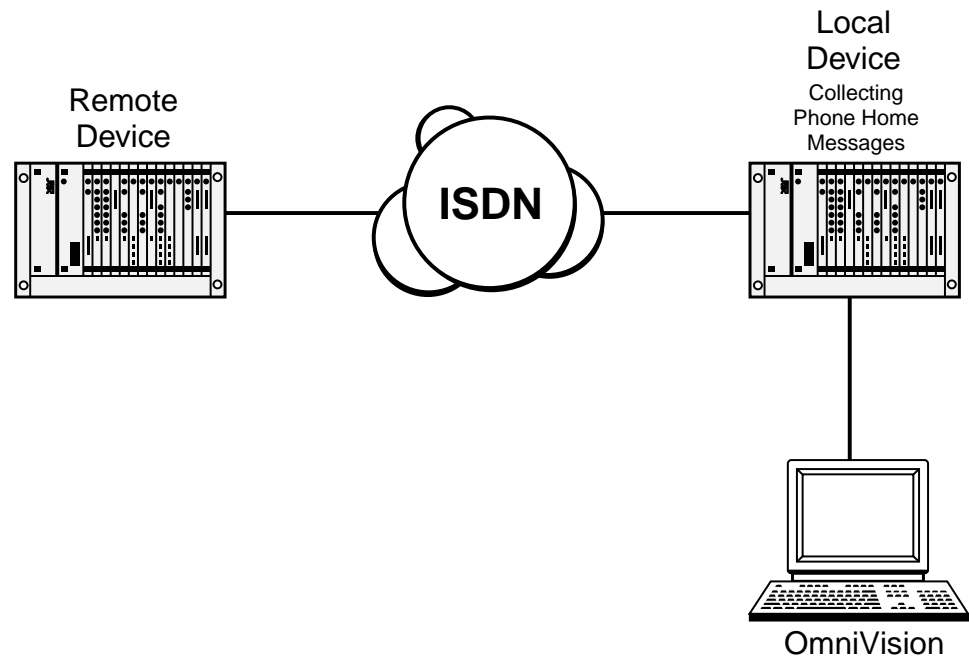
The Phone Home feature is an enhancement to the existing spontaneous alarm emission feature. It allows a central site manager to have confidence in the operational state of a device without having to maintain a management session or poll it. Instead, the device 'phones home' periodically if there are no alarms to report, thus providing a positive indication that it is operating correctly.

If the Phone Home call is made over a network which supports D-channel Message Demand Transfer Mode, it does not need to actually connect, and no flag fall charges are incurred.

Important *If the Phone Home call is made using a B-channel, the call will connect and it will incur a flag fall charge. We recommend that B-channels are not used for the Phone Home feature.*

Operation

The following diagram show how a Phone Home network could be constructed.



OmniVision is configured to poll the Remote Device at a fixed interval. The Remote Device is configured to call the Local Device which is set to collect the Phone Home messages at intervals less than that which OmniVision polls the device. The Local Device sends the collected messages to OmniVision as they are received.

If the Remote Device phones home within the OmniVision poll interval, OmniVision restarts the poll interval timer. If the Remote Device fails to phone home before the OmniVision poll timer expires, OmniVision polls the device. If the polled session fails, an alarm is raised for the device.

A poll interval period of five minutes longer than the Phone Home period is generally adequate to ensure that OmniVision does not have to poll the device when it is not required.

Note *In this release, the Phone Home feature is only supported if OmniVision is connected to the local device by a serial connection. It is **not** supported over a LAN connection.*

Configuration

This section is based around the simple network shown in the diagram above.

Remote device

To set up the Remote Device for Phone Home:

- 1 Select **Enable Alarm Signalling** in the **RM Configuration** dialog.
- 2 Set the **Phone Home Expiry Time**.
- 3 Enter the number of the Local Device in the **To Number** field. This is the device chosen to collect the messages on behalf of the central site manager (OmniVision). An alternative number (**Alt Number**) can be entered if required, as well as the **Number of retries**.

Local device

The Local Device, chosen to collect the phone home messages, requires no special configuration other than the entry of the **Manager Number**.

OmniVision

There are several steps for preparing OmniVision to support Phone Home. The basic steps are detailed below. Refer to the *OmniVision User Manual* for more information.

- 1 Enable **Accept Automatic Alarm Sessions** in **Option Settings**. Enter the details of how the Local Device should be accessed. Provide an **Access Number** if the device must be accessed remotely.
- 2 In **Navigator**, add the entries for the Remote Devices being monitored to the **Master List**. Ensure that the **Poll Time** in **Advanced Settings** is set to an interval larger than the **Phone Home Timer** in each Remote Device. Make the session **Online**.

- 3** Create a network map using OmniVision, and add the devices being monitored to the **OmniVision Map**. When adding these entries ensure that:
- the **Node Name** is the same as the name in the **Master List**
 - the **Net Address** is the IP Address of the OmniVision station
 - the **SNMP Community Names** accessed via the **Comm...** button are set to the **Node Name** as they appear on the **Master List**.

Once the above steps are complete and the objects are functioning correctly, the Map will display them in green. When an alarm occurs, the object changes to red. When the alarm clears, the indicator turns to magenta. You can view the alarms using the Alarm Log.

Module configuration

Introduction

This section provides information about each of the modules. The default settings for the parameters for all modules are listed, together with possible changes and configuration examples.

Note *To allow for full and flexible use of ISDN facilities, each module has many parameters and options within parameters that you can select. However, in most cases, the pre-existing default settings are correct. These are known to work and you should use them unless you have specific requirements and a thorough understanding of the protocols and their configuration.*

Some of the modules detailed in this section are only available to you when your chassis is fitted with a Resource Manager (RM). The following markers are used throughout the manual to indicate the modules supported by the Control Module, the Resource Manager or both:



Feature or module supported by Resource Manager.



Feature or module supported by Control Module.



ISDN Primary Rate TE Module (IPMT)

The ISDN Primary Rate TE Module (IPMT) is a 2.048 Mbit/s Terminal End-point (TE) module. It allows access to 30 B-channels and one D-channel by an NT1 at the T interface point.

The IPMT supports Dual/multiple node access. For further information on this subject, refer to *Appendix H*.

Configuration

Base Number	<p>Enter the base number in local significant format (no area code).</p> <p>The base number is subtracted from an incoming called party number and the call is routed to the resulting number. It is added to the line number and the area code to produce the calling party number on outgoing calls.</p> <p>If a base number is not entered, individual line numbers should be the full local significant number.</p> <p>The base number allows lines to be accessed from different ISDN lines or multiple ISDN modules that have different number ranges.</p>
Group Number	<p>Enter the group number in local significant format (no area code). It is mandatory when a semipermanent connection is used.</p> <p>The group number is the main directory number within the number range allocated by the ISDN provider. It is shown on the label of the NT1 device to which the IPMT is connected.</p>
CPN Prefix (UK only)	<p>Enter the prefix for the called party number. Call control prefixes the received called party number with the number set for the CPN prefix before performing any call routing.</p>
National Significant Area Code	<p>Enter the national significant area code. It is added to the calling line number for outgoing calls and is mandatory for semipermanent connections.</p> <p>The national significant area code is the area code defined by the ISDN provider, without the leading zero.</p>
B-channels available	<p>Accept the default, all 30 B-channels available. You do not need to change it.</p>
Force EnBloc Sending (ETSI only)	<p>When enabled, the IPMT always sends the dialling digits to the network enbloc. The IPMT collects digits until the Timed Dialling value expires, and then sends all the digits. You must enable Select channel if Force EnBloc Sending is enabled.</p>
Timed Dialling (ETSI only)	<p>This sets the maximum interval between dialled digits (interdigit timeout period). When the set value expires, all of the collected digits are sent to the network. This is used in conjunction with Force EnBloc Sending, and must not be set to disabled, when Force EnBloc Sending is enabled.</p>
# is Sending Complete (ETSI only)	<p>If you have enabled Force EnBloc Sending, this option determines the function of the dialled hash (#). If enabled, the number collection stops when a hash is received, and the number is sent enbloc to the network. If this option is disabled, the hash is collected as another digit to be dialled to the network.</p>
Disable T309	<p>T309 is a timer. If you enable it, and a prolonged failure occurs, calls are cleared after 15 seconds. This parameter is not mandatory.</p>

Enable CRC-4	<p>Enable this if your ISDN Terminal Equipment requires a cyclic redundancy check (CRC) to monitor the line. This enables CRC-4 error checking at Layer 1.</p> <p>CRC-4 is always transmitted. If you enable this function, CRC-4 data becomes the Layer 1 line monitoring criteria. If it is disabled, framing errors are interpreted as Layer 1 line errors.</p> <p>CRC Enable is not available for use on DASS2.</p>
Don't use as reference clock	<p>Enable this if you do not want the IPMT to supply the reference clock.</p>
Don't use for D-channel calls	<p>Enable this to prevent the Resource Manager or Control Module from delivering outgoing D-channel calls through the IPMT.</p>
Don't use as bus master	<p>Enable this to prevent the IPMT being selected to provide the backplane bus clock.</p>
Prefix called number (ROW only)	<p>Enable to change Group number to Called Party Number (CPN). See <i>Group number</i> and <i>CPN prefix</i> above.</p>
DMS-100 (idle fill L2)	<p>Enable this if you want to insert idle characters between messages instead of flags. This is required when interfacing with NorTel DMS-100.</p>
Retain Layer 2	<p>Enable this if you want to prevent the network from deactivating the link when it is idle. This is necessary if you want the IPMT to provide constant clock.</p>
Unknown number type	<p>Enable this if you want all calls leaving this port to be set to an unknown number, regardless of the type that was set when the call was generated.</p>
Suppress CLI Octet 3a	<p>Enable this if you want to prevent the sending of octet 3a of the Calling Party Number (CLI) information element. This octet contains information about the presentation and verification of the CLI. Some networks and PABXs will not recognise the calling party number without this option set.</p>
Restrict CLI digits to (UK and ROW only)	<p>Enable this if you want to limit the number of digits sent in the CLI (Calling Party Number). In the UK, the number sent as the CLI must be the same length as the number received from the network in the Called Party Number. The number of digits set (<i>n</i>), is the number of digits from the end of the line number that are sent as the CLI.</p>
Add progress to alerting	<p>Enable to add a progress indicator to the incoming alerting message. This is enabled by default for DASS modules, as DASS exchanges always provide tone, but do not support progress indicators.</p>
Auxiliary working (UK only)	<p>This is a special mode of operation used in the UK in which incoming calls on a particular channel are directed to a particular destination. No called party number information is provided in an incoming SETUP message.</p> <p>If a call arrives with no called party number and the IPMT is in Auxiliary Working mode, the device creates a called party number by adding the channel number to the module's Base Number. The resulting number is used to match either a line number or a lineset range as configured by the user.</p>
Protocol (ETSI only)	<p>Select the required protocol from the drop-down list. The list caters for minor differences in the ETSI protocol implementations of France and Germany.</p>
Convert notify to disconnect	<p>Select if phone conference units cannot clear a call. This allows the units to DISCONNECT a call immediately, instead of waiting the 90 seconds clear time initiated when the NOTIFY is received from the remote hang-up.</p>
Select channel	<p>Enable to force selection of a channel for outgoing calls.</p>
Exclusive/Pref. TE use only	<p>Enabling this removes the module from the TE table. The TE can then only be used for outgoing calls if the NT making the call has Outgoing TE exclusive selected.</p>

Forward Channel selection (UK only)

When channel selection is enabled for outgoing calls, channels are usually selected in reverse order, that is, channel 30 is selected first if it is available, and channel 1 is selected last. The channels used for auxiliary working are usually the higher numbered channels, so to avoid choosing them unless absolutely necessary, the selection order can be changed to begin at channel 1 and increment from there.



ISDN Primary Rate NT Module (IPMN)

The ISDN Primary Rate Module NT (IPMN) is a network interface emulation module, providing 30 B-channels and one D-channel. The module has a single port which provides the physical characteristics of an NT1 at the T interface point. Other ISDN TEs, such as other similar equipment or PABXs, may connect to this module.

Configuration

Maximum	Enter the maximum number in the range for this module.
Minimum	Enter the minimum number in the range for this module. (It may be the same as the maximum number.)

Notes

- *These are complete numbers; the base number is not applied.*
 - *Configuring the maximum and minimum numbers creates a lineset within the device. It contains the 30 B-channels and the D-channel as members.*
-

ISDN PRA NT Line Control defaults

Outgoing TE	Any.
CRC Enable	Enabled.
Don't use as bus master	Disabled.
Don't use as reference clock	Disabled.

You do not normally need to change these parameters.

Changing ISDN PRA NT Line Control defaults

Outgoing TE	Select the required TE for all the module's lines. If you select Preferred or Exclusive , you must then select the TE from the list of available TEs.
CRC Enable	Enable this if your ISDN Terminal Equipment requires a Cyclic Redundancy Check (CRC) to monitor the line. This enables CRC-4 error checking at Layer 1. CRC-4 is always transmitted. If you enable this function, CRC-4 data becomes the Layer 1 line monitoring criteria. If it is disabled, framing errors are interpreted as Layer 1 line errors.
Don't use as bus master	Enable this to prevent the IPMN being selected to provide the backplane bus clock.
Don't use as reference clock	Enable this if you do not want the IPMN to supply the reference clock.

Applications...

Connected to

Select the equipment that you are connecting to from the following available connections:

- VX IPMN
- Enbloc Receiving NT
- Overlap Receiving NT
- PBX TE or IPMT
- Custom (see below).

Custom settings

Suppress VX IE20	Select to strip from the message, any IE20s containing NETWORK IDs and PLANs that equal 0. Information Elements with these settings are specifically internal VXIEs. These are stripped before sending to the peer to avoid clashes of external IE20s (Network Specific Facilities) that may be generated by some PBXs and networks.
Suppress Display IE	Select to strip the DISPLAY IE before sending message to peer.
Convert Setup_Ack->Call Proc	Select to prevent SETUP ACKs messages being sent (this indicates that the connection is to an enbloc receiving peer).
Strip progress indicators	Select to strip progress indicators before sending the message to peer.
Convert User NSF (Network Specific Facilities)	Not applicable (T1 only).
Force Enbloc Sending	Not applicable (T1 only).
Options	
Idle fill Layer 2 (DMS-100)	Enable if you want to insert idle characters between messages instead of flags. This is necessary if you want the IPMN to provide a constant clock.
Channel Negotiation	<p>Enable if you want channel negotiation on incoming calls.</p> <p>When this parameter is enabled, a 'preferred' channel is selected. The connected PABX may however want to use the designated channel to set up its own outgoing call and will require the incoming call from the device to be set up on a different channel. In this case the PABX will attempt to negotiate the incoming call to another channel. This feature helps to prevent calls from being rejected should a call collision occur.</p> <p>When disabled, the call from the device specifies an 'exclusive' channel, and if a collision occurs, the incoming call is rejected.</p>
Extend T303 to 9.5s	Enable if you want to extend Timer T303 from 4 seconds to 9.5 seconds. Timer T303 supervises transmission of the Setup message. You should enable this if the equipment connected to the IPMN cannot respond within 4 seconds.
Add progress to alerting	When enabled, a Progress Indicator No.8 is added to the Alerting message sent from the peer.
Do not send progress for Cause 34	When enabled the Progress Indicator is not sent to the peer. This allows the peer to do a resubmission.

Resubmit on User Busy (ETSI only) When enabled, any call rejected with the disconnect cause 'User Busy' is translated to a cause of 'No Circuit Channel Available'. This allows a resubmission to occur.

Convert progress to alerting When enabled this causes alerting messages with a progress indicator sent from a peer to be converted to a progress message.

Options (ETSI only)

Open B-Channel early Select this to allow early connection of the B channel to the ISDN to prevent speech clipping.

Suppress CM/RM dial tone Select to prevent stray CM/RM dial tone being heard by the caller. Stray or delayed CM/RM dial tone occurs when the CM or RM is operating at high load.

Number type unknown This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Unknown'. This may be a requirement of some PBXs and exchanges.

Number type subscriber This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Subscriber'. This may be a requirement of some PBXs and exchanges.

Indial digits

Limit DDI digits (UK and ROW only) Enable if you want to limit the called party number to the number of digits set in 'DDI digits to send'. Only the last *n* called party number digits are sent to the IPMN.

Protocol (ETSI only)

Select the required protocol from the drop-down list. The list caters for minor differences in the ETSI protocol implementations of France and Germany.

Force EnBloc (ETSI only)

Force EnBloc Sending When enabled, the IPMT always sends the dialling digits to the network enbloc. The IPMN collects digits until the **Timed Dialling** value expires, and then sends all the digits. You must enable **Select channel** if **Force EnBloc Sending** is enabled.

Timed Dialling This sets the maximum interval between dialled digits (interdigit timeout period). When the set value expires, all of the collected digits are sent to the network. This is used in conjunction with **Force EnBloc Sending**, and must **not** be set to **disabled**, when **Force EnBloc Sending** is enabled.

is Sending Complete If you have enabled **Force EnBloc Sending**, this option determines the function of the dialled hash (#). If enabled, the number collection stops when a hash is received, and the number is sent enbloc to the network. If this option is disabled, the hash is collected as another digit to be dialled to the network.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls.

Configure individual lines

The complete line configuration for each of the B-channels and D-channel is available using this dialog; however, the configuration is not normally used, as the IPMN ISDN messages are derived from those coming from the TE.



ISDN Primary Rate NT Module -T1 (IPMN-T1)

The ISDN Primary Rate Module NT- T1 (IPMN-T1) is a network interface emulation module, providing 23 B-channels and one D-channel. The module has a single port which provides the physical characteristics of an NT1 at the T interface point. Other ISDN TEs, such as other similar equipment or PABXs, may connect to this module.

Configuration

Maximum	Enter the maximum number in the range for this module.
Minimum	Enter the minimum number in the range for this module. (It may be the same as the maximum number.)

Notes

- *These are complete numbers; the base number is not applied.*
 - *Configuring the maximum and minimum numbers creates a lineset within the device. It contains the 23 B-channels and the D-channel as members.*
-

Card options

ISDN PRI NT Line Control defaults

Outgoing TE	Any.
Don't use as bus master	Disabled.
Don't use as reference clock	Disabled.

You do not normally need to change these parameters.

Changing ISDN PRI NT Line Control defaults

Outgoing TE	Select the required TE for all the module's lines. If you select Preferred or Exclusive , you must then select the TE from the list of available TEs.
Don't use as bus master	Enable this to prevent the IPMN-T1 being selected to provide the backplane bus clock.
Don't use as reference clock	Enable this if you do not want the IPMN-T1 to supply the reference clock.

DLI gain

DLI gain	Select Line Build Out (LBO) options for long haul or short haul operation. Long haul options include pulse shape and gain selection. Short haul options are selected on length of cable.
-----------------	--

Applications...

Connected to

Select the equipment that you are connecting to from the following available connections:

- VX IPMN
- Enbloc Receiving NT
- Overlap Receiving NT
- PBX TE or IPMT
- Custom (see below).

Custom settings

Suppress VX IE20	Select to strip from the message, any IE20s containing NETWORK IDs and PLANs that equal 0. Information Elements with these settings are specifically internal VX IEs. These are stripped before sending to the peer to avoid clashes of external IE20s (Network Specific Facilities) that may be generated by some PBXs and networks.
Suppress Display IE	Select to strip the DISPLAY IE before sending message to peer.
Convert Setup_Ack->Call Proc	Select to prevent SETUP ACKs messages being sent (this indicates that the connection is to an enbloc receiving peer).
ISDN National Number Mode	Select to set the Called Party Number 'Type of Number' field to 'National' for all calls sent to the peer.

Note *This parameter should not be selected if you have selected **Number Type Unknown** or **Number Type Subscriber** in the Options field. The three parameters are mutually exclusive.*

Convert User NSF (Network Specific Facilities)	When enabled, this parameter allows specialized handling of NSF IEs (IE20) generated by some PBXs and networks. The handling involves the conversion of IE20s before sending and their translation at the peer. Deselect to disable the specialised handling of peer NSF IEs.
Force EnBloc	
Force EnBloc Sending (ETSI only)	When enabled, the IPMT always sends the dialling digits to the network enbloc. The IPMN-T1 collects digits until the Timed Dialling value expires, and then sends all the digits. You must enable Select channel if Force EnBloc Sending is enabled.
Timed Dialling (ETSI only)	This sets the maximum interval between dialled digits (interdigit timeout period). When the set value expires, all of the collected digits are sent to the network. This is used in conjunction with Force EnBloc Sending , and must not be set to disabled , when Force EnBloc Sending is enabled.
# is Sending Complete (ETSI only)	If you have enabled Force EnBloc Sending , this option determines the function of the dialled hash (#). If enabled, the number collection stops when a hash is received, and the number is sent enbloc to the network. If this option is disabled, the hash is collected as another digit to be dialled to the network.
ESS Peer National->Unknown	Select to set the Called Party Number 'Type of Number' to 'Unknown' for all calls received from the peer.

Options

Idle fill Layer 2 (DMS-100)	Enable if you want to insert idle characters between messages instead of flags. This is necessary if you want the IPMN-T1 to provide a constant clock.
Channel Negotiation	Enable if you want channel negotiation on incoming calls. When this parameter is enabled , a 'preferred' channel is selected. The connected PABX may however want to use the designated channel to set up its own outgoing call and will require the incoming call to be set up on a different channel. In this case the PABX will attempt to negotiate the incoming call to another channel. This feature helps to prevent calls from being rejected should a call collision occur. When disabled , the call specifies an 'exclusive' channel, and if a collision occurs, the incoming call is rejected.
Extend T303 to 9.5s	Enable if you want to extend Timer T303 from 4 seconds to 9.5 seconds. Timer T303 supervises transmission of the Setup message. You should enable this if the equipment connected to the IPMN-T1 cannot respond within 4 seconds.
Add progress to alerting	Enable to add a progress indicator to the incoming alerting message.
Open B Channel early	Enable so that a B-channel path is opened immediately the module receives a response for an outgoing call.
Suppress CM/RM dial tone	Enable to prevent the CM/RM generating a dial tone (or anything on the speech path) as soon as INFORMATION messages are received for an incoming call.
Number type unknown	Select to set the Called Party Number 'Type of Number' to 'Unknown' for all calls received from the peer.
Number type subscriber	Select to set the Called Party Number 'Type of Number' to 'Subscriber' for all calls received from the peer. This is required by some networks.

Note *The **Number Type Unknown** and **Number Type Subscriber** parameters should not be selected if you have enabled **ISDN National Number Mode**. These three parameters are mutually exclusive. See page 4-10 for details of the **ISDN National Number Mode** parameter.*

Do not send progress for Cause 34	When enabled, disconnect messages do not have a progress indicator and the VX does not provide tone.
Support 4ESS	Selecting this enables support for 4ESS-specific Information Elements.
Enforce Billing Protection	When enabled, this parameter prevents outgoing calls being cleared within 2 seconds of receiving the answer. This is an FCC requirement.
Indial digits	
Limit DDI digits (UK and ROW only)	Enable if you want to limit the called party number to the number of digits set in 'DDI digits to send'. Only the last <i>n</i> called party number digits are sent to the IPMN-T1.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls.

Configure individual lines

The complete line configuration for each of the B-channels and D-channel is available using this dialog; however, the configuration is not normally used, as the IPMN ISDN messages are derived from those coming from the TE.



ISDN Primary Rate TE Module-T1 (IPMT-T1)

The ISDN Primary Rate TE Module-T1 (IPMT-T1) is a 1.544 Mbit/s Terminal End-point (TE) module. It allows access to 23 B-channels and one D-channel by an NT1 at the T interface point.

The IPMT allows the support of dual/multiple node access. For further information on this subject, refer to *Appendix H*.

Configuration

Base Number	<p>Enter the base number in local significant format (no area code).</p> <p>The base number is subtracted from an incoming called party number and the call is routed to the resulting number. It is added to the line number and the area code to produce the calling party number on outgoing calls.</p> <p>If a base number is not entered, individual line numbers should be the full local significant number.</p> <p>The base number allows lines to be accessed from different ISDN lines or multiple ISDN modules that have different number ranges.</p>
Group Number	<p>Enter the group number in local significant format (no area code). It is mandatory when a semipermanent connection is used.</p> <p>The group number is the main directory number within the number range allocated by the ISDN provider. It is shown on the label of the NT1 device to which the IPMT is connected.</p>
CPN Prefix	<p>Enter the prefix for the called party number. Call control prefixes the received called party number with the number set for the CPN prefix before performing any call routing.</p>
National Significant Area Code	<p>Enter the national significant area code. It is added to the calling line number for outgoing calls and is mandatory for semipermanent connections.</p> <p>The national significant area code is the area code defined by the ISDN provider, without the leading escape prefix.</p>
DLI gain	<p>Select Line Build Out (LBO) options for long haul or short haul operation. Long haul options include pulse shape and gain selection. Short haul options are selected on length of cable.</p>
B-channels available	<p>Accept the default, all 23 B-channels available. You do not need to change it.</p>
Disable T309	<p>T309 is a timer. If you enable it, and a prolonged failure occurs, calls are cleared after 15 seconds. This parameter is not mandatory.</p>
Don't use as reference clock	<p>Enable this if you do not want the IPMT-T1 to supply the reference clock.</p>
Don't use for D-channel calls	<p>Enable this to prevent the Resource Manager or Control Module from delivering outgoing D-channel calls through the IPMT-T1.</p>
Don't use as bus master	<p>Enable this to prevent the IPMT-T1 being selected to provide the backplane bus clock.</p>
Prefix called number	<p>Enable to change Group number to Called Party Number (CPN). See <i>Group number</i> and <i>CPN prefix</i> above.</p>

Add progress to alerting	Enable to add a progress indicator to the incoming alerting message.
DMS-100 (idle fill L2)	Enable this if you want to insert idle characters between messages instead of flags. This is required when interfacing with NorTel DMS-100.
Retain Layer 2	Enable this if you want to prevent the network from deactivating the link when it is idle. This is necessary if you want the IPMT-T1 to provide constant clock.
Unknown number type	Enable this if you want all calls leaving this port to be set to an unknown number, regardless of the type that was set when the call was generated.
Suppress CLI Octet 3a	Enable this if you want to prevent the sending of octet 3a of the Calling Party Number (CLI) information element. This octet contains information about the presentation and verification of the CLI. Some networks and PABXs will not recognize the calling party number without this option set.
Restrict CLI digits to	Enable this if you want to limit the number of digits sent in the CLI (Calling Party Number). In the UK, the number sent as the CLI must be the same length as the number received from the network in the Called Party Number. The number of digits set (<i>n</i>), is the number of digits from the end of the line number that are sent as the CLI.
Select channel	Enable to force selection of a channel for outgoing calls.
Exclusive/preferred TE use only	Enable so that this TE will only be selected for an outgoing call if the originating line in the device is set to 'Exclusive' or 'Preferred'.
Support 4ESS	Selecting this enables support for 4ESS-specific Information Elements.
Support Billing Protection	When enabled, this parameter prevents outgoing calls being cleared within 2 seconds of initiation. This is an FCC requirement.
Force EnBloc Sending (ETSI only)	When enabled, the IPMT always sends the dialling digits to the network enbloc. The IPMT-T1 collects digits until the Timed Dialling value expires, and then sends all the digits. You must enable Select channel if Force EnBloc Sending is enabled.
Timed Dialling (ETSI only)	This sets the maximum interval between dialled digits (interdigit timeout period). When the set value expires, all of the collected digits are sent to the network. This is used in conjunction with Force EnBloc Sending , and must not be set to disabled , when Force EnBloc Sending is enabled.
# is Sending Complete (ETSI only)	If you have enabled Force EnBloc Sending , this option determines the function of the dialled hash (#). If enabled, the number collection stops when a hash is received, and the number is sent enbloc to the network. If this option is disabled, the hash is collected as another digit to be dialled to the network.

ISDN Gateway Module (IPMN-GT)

The IPMN-GT is an E1 ISDN module with the ability to convert A-law to μ -law. Apart from this extra feature, the IPMN-GT is functionally identical to the IPMN-2.

The primary application for the module is to allow the connection of international E1/A-law links to devices operating in μ -law environments. The IPMN-GT runs the ETSI ISDN protocol.

The IPMN-GT is only required at one end of an international link. The module at the other end can be an IPMN-ETSI.

Configuration

Maximum	Enter the maximum number in the range for this module.
Minimum	Enter the minimum number in the range for this module. (It may be the same as the maximum number.)

Notes

- *These are complete numbers; the base number is not applied.*
 - *Configuring the maximum and minimum numbers creates a lineset within the device. It contains the 30 B-channels and the D-channel as members.*
-

ISDN PRI NT Line Control defaults

Outgoing TE	Any.
CRC Enable	Enabled.
Don't use as bus master	Disabled.
Don't use as reference clock	Disabled.

You do not normally need to change these parameters.

Changing ISDN PRI NT Line Control defaults

Outgoing TE	Select the required TE for all the module's lines. If you select Preferred or Exclusive , you must then select the TE from the list of available TEs.
CRC Enable	Enable this if your ISDN Terminal Equipment requires a Cyclic Redundancy Check (CRC) to monitor the line. This enables CRC-4 error checking at Layer 1. CRC-4 is always transmitted. If you enable this function, CRC-4 data becomes the Layer 1 line monitoring criteria. If it is disabled, framing errors are interpreted as Layer 1 line errors.
Don't use as bus master	Enable this to prevent the IPMN-GT being selected to provide the backplane bus clock.
Don't use as reference clock	Enable this to prevent the IPMN-GT being selected to provide the backplane reference clock.

Applications...

Connected to

Select the equipment that you are connecting to from the following available connections:

- VX IPMN
- Enbloc Receiving NT
- Overlap Receiving NT
- PBX TE or IPMT
- Custom (see below).

Custom settings

Suppress VX IE20	Select to strip from the message, any IE20s containing NETWORK IDs and PLANs that equal 0. Information Elements with these settings are specifically internal VXIEs. These are stripped before sending to the peer to avoid clashes of external IE20s (Network Specific Facilities) that may be generated by some PBXs and networks.
Suppress Display IE	Select to strip the DISPLAY IE before sending message to peer.
Convert Setup_Ack->Call Proc	Select to prevent SETUP ACKs messages being sent (this indicates that the connection is to an enbloc receiving peer).
Strip progress indicators	Select to strip progress indicators before sending the message to peer.
Convert User NSF (Network Specific Facilities)	Not applicable (T1 only).
Force Enbloc Sending	Not applicable (T1 only).
Options	
Idle fill Layer 2 (DMS-100)	Enable if you want to insert idle characters between messages instead of flags. This is necessary if you want the IPMN to provide a constant clock.
Channel Negotiation	<p>Enable if you want channel negotiation on incoming calls.</p> <p>When this parameter is enabled, a 'preferred' channel is selected. The connected PABX may however want to use the designated channel to set up its own outgoing call and will require the incoming call to be set up on a different channel. In this case the PABX will attempt to negotiate the incoming call to another channel. This feature helps to prevent calls from being rejected should a call collision occur.</p> <p>When disabled, the call from the device specifies an 'exclusive' channel, and if a collision occurs, the incoming call is rejected.</p>
Extend T303 to 9.5s	Enable if you want to extend Timer T303 from 4 seconds to 9.5 seconds. Timer T303 supervises transmission of the Setup message. You should enable this if the equipment connected to the IPMN cannot respond within 4 seconds.
Add progress to alerting	When enabled, a Progress Indicator No.8 is added to the Alerting message sent from the peer.
Do not send progress for Cause 34	When enabled the Progress Indicator is not sent to the peer. This allows the peer to do a resubmission.

- Convert progress to alerting** When enabled this causes alerting messages with a progress indicator sent from a peer to be converted to a progress message.
- Open B-Channel early** Select this to allow early connection of the B channel to the ISDN to prevent speech clipping.
- Suppress CM/RM dial tone** Select to prevent stray CM/RM dial tone being heard by the caller. Stray or delayed CM/RM dial tone occurs when the CM/RM is operating at high load.
- Number type unknown** This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Unknown'. This may be a requirement of some PBXs and exchanges.
- Number type subscriber** This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Subscriber'. This may be a requirement of some PBXs and exchanges.
- Line PCM coding** Select **A-law** to force line PCM coding to A-law.
Select **μ-law** to force line PCM coding to μ-law.
Select **None** to prevent conversion of PCM coding.
- Indial digits**
- Limit DDI digits (UK and ROW only)** Enable if you want to limit the called party number to the number of digits set in 'DDI digits to send'. Only the last *n* called party number digits are sent to the IPMN.

Protocol

Select the required protocol from the drop-down list. The list caters for minor differences in the ETSI protocol implementations of France and Germany.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls.

Configure individual lines

The complete line configuration for each of the B-channels and D-channel is available using this dialog; however, the configuration is not normally used, as the IPMN ISDN messages are derived from those coming from the TE.

Basic Rate TE Module (BRMT)

The ISDN Basic Rate TE Module (BRMT) is a Terminal End-point (TE) module that allows access to two B-channels and one D-channel provided by an NT1 at the S interface point.

The BRMT allows the support of Dual/Multiple node access. For further information on this subject, refer to *Appendix H*.

The parameters available for configuration are described below.

Configuration

Base Number Enter the base number in local significant format (no area code). This is optional, but if entered, it must conform with other numbers in the ISDN range.

The base number is subtracted from an incoming called party number and the call is routed to the resulting number. It is added to the line number and the area code to produce the calling party number on outgoing calls.

If the base number is not entered, individual line numbers must be the full local significant number.

The base number allows lines to be accessed from different ISDN lines or multiple ISDN modules that have different number ranges.

Multiple BRMTs may share the same number range where incoming calls are distributed according to the criteria employed by the network provider.

Group Number **Australia**

The group number is the main directory number within the number range allocated by your ISDN service provider. It is shown on the label of the NT1 device to which the BRMT is connected.

Enter the group number in local significant format.

UK

The significance and function of the group number depends upon the type of Basic Rate interface to which the BRMT is connected. The types available from British Telecom are:

- **Simple**
No called number is provided to the BRMT. All calls received through the BRMT are presented to the device with the Group Number as the called number. For example, if the group number is 123456, a call received will match a port with the line number 123456. In this case, the group number is mandatory. Enter the number that is used to address the Basic Rate Access.
- **Multiple Subscriber Number (MSN)**
Only the last digit dialled is provided to the BRMT, which appends it to the group number to make up the device number. Enter the first five digits of the ISDN Access number. For example, if the MSN interface number range is from 245670 to 245679, the group number is 24567.
- **DDI**
The network provides all six local digits to the BRMT. The group number is not used.

National Significant Area Code

Enter the National Significant Area Code.

The National Significant Area Code is the area code defined by the ISDN provider, without the leading zero.

Disable T309	T309 is a timer. If you enable it, and a failure occurs, calls are cleared after 2 seconds. The default is 'Disabled'. This parameter is not mandatory.
Do not use as reference clock	Enable this if you do not want the BRMT to supply the reference clock.
Do not use for D-channel calls	Enable this to prevent the Resource Manager or Control Module from delivering outgoing D-channel calls through the BRMT.
Do not use as bus master	Enable this to prevent the BRMT being selected to provide the backplane bus clock.
Retain Layer 2 establishment	Enable this if you want to prevent the network from deactivating the link when it is idle. This is necessary if you want the BRMT to provide constant clock.
Exclusive/preferred use only	Enabling this removes the port from the TE table. The TE can then only be used for outgoing calls if the NT making the call has Outgoing TE exclusive selected.
Add progress to alerting	Enable to add a progress indicator to the incoming alerting message.
Prefix called numbers (UK and ROW only)	Enable this to change the Group Number to the CPN (Called Party Number) prefix. Leave disabled if you want to use the Group Number. When enabled, the number entered in the CPN prefix is used to prefix the called party number received for incoming calls. Call control prefixes the received called party number before performing any call routing.
Point to point TE	Select this to allow the BRMT to operate in Point to Point mode. In this mode the BRMT accepts incoming calls delivered on a Point to Point TEI. Restart messages are also accepted in Point to Point mode. We recommend that a Fixed TEI of 0 be used when operating in Point to Point mode.
Restrict calling digits to (UK and ROW only)	Enable this if you want to limit the number of digits sent in the Calling Party Number. In the UK, the number sent as the Calling Party Number must be the same length as the number received from the network in the Called Party Number. The number of digits set (<i>n</i>), is the number of digits from the end of the line number that are sent as the Calling Party Number.
Send Calling Party Number (UK and ROW only)	Enable this if you want to send the Calling Party Number in the keypad information element rather than in the Called Party Number information element.
TEI assignment	The default is Automatic . Select Fixed and enter a number if you want to identify a specific connection endpoint within a service access point.
Protocol	The protocol defaults to the standard required for the selected area of operation. For example, TS13 in Australia, and ETSI in the UK.



Basic Rate NT Module (BRMN)

The ISDN Basic Rate Module NT (BRMN) is a network interface emulation module, providing two 64 kbit/s B-channels and one 16 kbit/s D-channel. The module has a single port which provides the physical characteristics of an NT1 at the S interface point.

Configuration

Maximum	Enter the maximum number in the range for this module.
Minimum	Enter the minimum number in the range for this module (it may be the same as the maximum number).

Notes

- *These are complete numbers; the base number is not applied.*
 - *Configuring the maximum and minimum numbers creates a lineset within the device. It contains the two B-channels and the D-channel as members.*
-

ISDN BRA NT Line Control defaults

Outgoing TE	Any.
TEI audit	Disabled.
Don't use as bus master	Disabled.
Limit DDI digits (UK and ROW only)	Disabled.
Point to point (UK and ROW only)	Disabled.
Protocol	The protocol defaults to the standard required for the selected area of operation. For example, TS13 in Australia, and ETSI in the UK.

Changing ISDN BRA NT Line Control defaults

Outgoing TE	Select the required TE. If you select Preferred or Exclusive , you must then select the TE from the displayed list. This setting affects all the module's lines.
TEI audit	Enable this if you want to check the TEIs on a regular basis. The time is in minutes.
Don't use as bus master	Enable this to prevent the BRMN being selected to provide the backplane bus clock.
Limit DDI digits (UK and ROW only)	Limit DDI restricts the number of digits that will be dialled into the PABX from the BRMN. If this is not enabled, the full number received by the BRMN will be forwarded. If the parameter is enabled, and a number 'n' entered, only the last 'n' digits of the number will be forwarded. This setting applies to the entire module, and is not line by line configurable.

Point to point Enable this if the BRMN is connected to a TE operating in a point-to-point mode.
(UK and ROW only)

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls.

Configure individual lines

The complete line configuration for each of the B-channels and D-channel is available using this dialog; however, the configuration is not normally used, as the BRMN ISDN messages are derived from those coming from the TE.



Dual Basic Rate Module (DBRM)

The Dual Basic Rate Modules (DBRM) provides two 'S' bus ports on a single card. Each interface can be configured by links on the board to provide the Network Termination (NT) function or the Terminal Equipment (TE) function. Any combination may be configured.

The ports provide the features of the BRMN and BRMT single port modules, with some enhancements. The operation of the multiple port BRM is dependent on the Resource Manager.

Configuration

- | | |
|-------------------------------------|--|
| Select individual port types | When you create a new configuration in an off-line session you must enter the port type for each of the two or four ports as appropriate – either Basic Rate TE or Basic Rate NT. This can then be downloaded at some time in the future. Port configuration of NTs and TEs must match the type of multiple port BRM that is to be used. |
| Configuring the ports | For further configuration select the port you require. Configuration details for each port type follow. |

Note *When a port is changed from TE mode to NT mode or vice-versa (by changing the hardware links) or when a module is replaced by one with a different port configuration, the original module must be deleted using the NMS prior to insertion.*

Configuring a TE port

- | | |
|---------------------|---|
| Base Number | <p>Enter the base number in local significant format (no area code). This is optional, but if entered, it must conform with other numbers in the ISDN range.</p> <p>The base number is subtracted from an incoming called party number and the call is routed to the resulting number. It is added to the line number and the area code to produce the calling party number on outgoing calls.</p> <p>If the base number is not entered, individual line numbers must be the full local significant number.</p> <p>The base number allows lines to be accessed from different ISDN lines or multiple ISDN modules that have different number ranges.</p> <p>Multiple TEs may share the same number range where incoming calls are distributed according to the criteria employed by the network provider.</p> |
| Group Number | <p>In Australia, the group number is the main directory number within the number range allocated by your ISDN service provider. It is shown on the label of the NT1 device to which the multiple port BRM is connected. Enter the group number in local significant format.</p> |

In the UK, the significance and function of the group number depends upon the type of Basic Rate interface to which the module is connected. The types available from British Telecom are:

- **Simple**
No called number is provided to the TE port. All calls received through the module are presented to the device with the Called Party Number prefix (CPN) as the called number. For example, if the CPN prefix is 123456, a call received will match a port with the line number 123456. In this case, the CPN prefix is mandatory. Enter the number that is used to address the Basic Rate Access.
- **Multiple Subscriber Number (MSN)**
Only the last digit dialled is provided to the TE port, which appends it to the CPN prefix to make up the device number. Enter the first five digits of the ISDN Access number. For example, if the MSN interface number range is from 245670 to 245679, the CPN prefix is 24567.
- **DDI**
The network provides all six local digits to the module. The CPN prefix is not used.

National Significant Area Code	Enter the National Significant Area Code. The National Significant Area Code is the area code defined by the ISDN provider, without the leading zero.
Drop first n digit in incoming call (Spectrum only)	Use this parameter to remove any unwanted numbers or area code from the incoming call. Typically a SPID is a 10 digit entity. If a called number is less than 10 numbers, padding must be added to make up the 10 digits.
Prefix number dialled with (Spectrum only)	This field signifies the digit that needs to be dialled to obtain an external line, prior to dialling the required number.
Local number size (Spectrum only)	This describes the size of a local number. The setting determines whether the local number requires additional information to dial out.
Indial number size (Spectrum only)	This describes the size of a number used to dial a local extension. The setting ensures that all indial numbers of this size do not require additional information before dialling.
Disable T309	T309 is a timer. If you enable it, and a failure occurs, calls are cleared after 2 seconds. The default is 'Disabled'. This parameter is not mandatory.
Do not use as reference clock	Enable this if you do not want the module to supply the reference clock.
Do not use as bus master	Enable this to prevent the module being selected to provide the backplane bus clock.
Exclusive/preferred use only	Enabling this removes the port from the TE table. The TE can then only be used for outgoing calls if the NT making the call has Outgoing TE exclusive selected.
Do not use for D-channel calls	Enable this to prevent the Resource Manager from delivering outgoing D-channel calls through the module.
Retain Layer 2 establishment	Enable this if you want to prevent the network from deactivating the link when it is idle. This is necessary if you want the module to provide constant clock.

Prefix called numbers (UK and ROW only)	Enable this to change the Group Number to the CPN (Called Party Number) prefix. Leave disabled if you want to use the Group Number. When enabled, the number entered in the CPN prefix is used to prefix the called party number received for incoming calls. Call control prefixes the received called party number before performing any call routing.
Restrict calling digits to (UK and ROW only)	Enable this if you want to limit the number of digits sent in the Calling Party Number. In the UK, the number sent as the Calling Party Number must be the same length as the number received from the network in the Called Party Number. The number of digits set (<i>n</i>), is the number of digits from the end of the line number that are sent as the Calling Party Number.
Unknown Number Type	This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Unknown'. This may be a requirement of some PBXs and exchanges.
Send Calling Party Number	Enable this if you want to send the Calling Party Number in the keypad information element rather than in the Called Party Number Information Element.
Add progress to alerting	Enable to add a progress indicator to the incoming alerting message.
Point to point TEI assignment	Enable if operating in Point-to-point mode.
Protocol	The default is Automatic . Select Fixed and enter a number if you want to identify a specific connection endpoint within a service access point.
	The protocol defaults to the standard required for the selected area of operation. For example, TS13 and Spectrum in Australia, and ETSI in the UK. Only the protocols supported by your module will be available.

Spectrum protocol (where available)

Spectrum is a public switched network service that uses network intelligence to provide PABX-like facilities to customers who do not purchase a PABX. Spectrum handles a limited multidrop S-bus. It can handle only two terminals on one S-bus unlike TS13 which can handle eight.

The only noted difference between TS13 and Spectrum is the need for the negotiation process of a Service Profile Identification (SPID). This process identifies terminal equipment with certain capabilities to the network. The network can then accept outgoing calls and route incoming calls. If the SPID is not negotiated, no network call routing occurs.

The SPID number is given by the service provider at the time the network connection is entered. A SPID is related to a B-channel and a Terminal Endpoint Identifier (TEI). The relationship with the TEI is physical (one-to-one), but it is only logical with the B-channel. A SPID has a unique TEI value within the S-bus and can use any one of the B-channels available in that S-bus.

An ISDN port requires a SPID and a Bearer Capability to be programmed for each TEI (or B-channel) to be used. The Bearer Capability is used to filter the types of calls being made from the line. This is to ensure that the call type matches the line's capabilities.

With Spectrum, ISDN Basic Rate ports are capable of making only the type of calls dictated by the SPIDs. A port can have a SPID set for data, and the other port set for voice, but then the ports will be capable of making only one call of each type.

For the ISDN ports to operate correctly, the Spectrum SPID must be of a type called Initialising SPID and **not** a Non-Initialising SPID. An Initialising SPID identifies itself to the network through negotiating a SPID on startup. The module's ISDN ports are of the Initialising type.

If two SPIDs are to be used then the S-bus must be configured to have two initialising SPIDs. A Spectrum S-bus can have either two Initialising SPIDs or a combination of Initialising and Non-Initialising. This can be configured only by the service provider. It is essential that the SPID is configured correctly or the Spectrum connection will not work.

SPID configuration

- Number of SPIDs** Select the number of SPIDs you wish to configure — one or two.
- SPID *n*** Enter the SPID number for each line as supplied by your ISDN provider. If required, select Bearer Layer Capabilities to change these properties. See *Bearer Capability* on page 2-6 of the *Parameters* section for more information.

National ISDN 1 protocol (USA only)

NI1 is a commonly used North American ISDN Basic Rate protocol. The DBRM currently supports TE mode over S/T physical interfaces. Standard NT1s can be used for 'U' interfaces.

A major feature of NI1 protocol is the use of Service Profile Identification (SPID). A SPID is a nine to twenty digit number and is used to uniquely identify a particular set of subscribed service capabilities. The NI1 network has a one-to-one relationship between SPID, Directory Number, TEI and Bearer Capability. The modules support a maximum of two different Initialising SPIDs per port.

Some NI1 services support only one Call Type (CT) bearer per SPID, where as other services can support multiple CT bearers per SPID. In the latter, linesets are used and calls are matched on bearer capability.

SPID and Directory Numbers require a one-to-one mapping. For this reason, user ports should have the **Exclusive/preferred TE Selection** set to the appropriate NI1 port with the SPID. See page 2-19 of the *Parameters* section for details.

Configuring the default SPID

- Default SPID** The SPID is used as a default under 'cold start-up' conditions, for example, an installed, but yet unconfigured device. The default SPID allows remote access even when the RM is not configured. The SPID value is stored in the module's non-volatile memory.

The SPID can also be programmed directly through the V.24 port on the module, without the need for an NMS connection. See the *Installation Manual* for details of configuring the default SPID.

Configuring an NT port

Maximum	Enter the maximum number in the range for this module.
Minimum	Enter the minimum number in the range for this module. (It may be the same as the maximum number.)

Notes

- *These are complete numbers; the base number is not applied.*
 - *Configuring the maximum and minimum numbers creates a lineset within the device. It contains the two B-channels and the D-channel as members.*
-

ISDN BRA NT Line Control defaults

Outgoing TE	Any.
TEI audit	Disabled.
Don't use as bus master	Disabled.
Limit DDI digits (UK and ROW only)	Disabled.
Point to point (UK and ROW only)	Disabled.

Changing ISDN BRA NT Line Control defaults

Outgoing TE	Select the required TE. If you select Preferred or Exclusive , you must then select the TE from the displayed list. This setting affects all the module's lines.
TEI audit	Enable this if you want to check the TEIs on a regular basis. The time is in minutes.
Don't use as bus master	Enable this to prevent the module being selected to provide the backplane bus clock.
Limit DDI digits (UK and ROW only)	Limit DDI restricts the number of digits that will be dialled into the PABX from the module. If this is not enabled, the full number received by the module will be forwarded. If the parameter is enabled, and a number 'n' entered, only the last 'n' digits of the number will be forwarded. This setting applies to the entire module, and is not line by line configurable.
Number type unknown	This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Unknown'. This may be a requirement of some PBXs and exchanges.
Point to point	Enable this if the NT port is connected to a TE operating in a point-to-point mode.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls.

Configure individual lines

The complete line configuration for each of the B-channels and D-channel is available using this dialog; however, the configuration is not normally used, as the module's ISDN messages are derived from those coming from the TE.



Quad Basic Rate Module (QBRM)

The Quad Basic Rate Module (QBRM) provides four 'S' bus ports on a single card. Each interface can be configured by links on the board to provide the Network Termination (NT) function or the Terminal Equipment (TE) function. Any combination may be configured.

The ports provide the features of the BRMN and BRMT single port modules, with some enhancements. The operation of the multiple port BRM is dependent on the Control Module or Resource Manager.

Configuration

- | | |
|-------------------------------------|--|
| Select individual port types | When you create a new configuration in an off-line session you must enter the port type for each of the two or four ports as appropriate – either Basic Rate TE or Basic Rate NT. This can then be downloaded at some time in the future. Port configuration of NTs and TEs must match the type of multiple port BRM that is to be used. |
| Configuring the ports | For further configuration select the port you require. Configuration details for each port type follow. |

Note *When a port is changed from TE mode to NT mode or vice-versa (by changing the hardware links) or when a module is replaced by one with a different port configuration, the original module must be deleted using the NMS prior to insertion.*

Configuring a TE port

- | | |
|---------------------|---|
| Base Number | <p>Enter the base number in local significant format (no area code). This is optional, but if entered, it must conform with other numbers in the ISDN range.</p> <p>The base number is subtracted from an incoming called party number and the call is routed to the resulting number. It is added to the line number and the area code to produce the calling party number on outgoing calls.</p> <p>If the base number is not entered, individual line numbers must be the full local significant number.</p> <p>The base number allows lines to be accessed from different ISDN lines or multiple ISDN modules that have different number ranges.</p> <p>Multiple TEs may share the same number range where incoming calls are distributed according to the criteria employed by the network provider.</p> |
| Group Number | <p>In Australia, the group number is the main directory number within the number range allocated by your ISDN service provider. It is shown on the label of the NT1 device to which the multiple port BRM is connected. Enter the group number in local significant format.</p> |

In the UK, the significance and function of the group number depends upon the type of Basic Rate interface to which the module is connected. The types available from British Telecom are:

- **Simple**
No called number is provided to the TE port. All calls received through the module are presented to the device with the Called Party Number prefix (CPN) as the called number. For example, if the CPN prefix is 123456, a call received will match a port with the line number 123456. In this case, the CPN prefix is mandatory. Enter the number that is used to address the Basic Rate Access.
- **Multiple Subscriber Number (MSN)**
Only the last digit dialled is provided to the TE port, which appends it to the CPN prefix to make up the device number. Enter the first five digits of the ISDN Access number. For example, if the MSN interface number range is from 245670 to 245679, the CPN prefix is 24567.
- **DDI**
The network provides all six local digits to the module. The CPN prefix is not used.

National Significant Area Code	Enter the National Significant Area Code. The National Significant Area Code is the area code defined by the ISDN provider, without the leading zero.
Drop first n digit in incoming call (Spectrum only)	Use this parameter to remove any unwanted numbers or area code from the incoming call. Typically a SPID is a 10 digit entity. If a called number is less than 10 numbers, padding must be added to make up the 10 digits.
Prefix number dialled with (Spectrum only)	This field signifies the digit that needs to be dialled to obtain an external line, prior to dialling the required number.
Local number size (Spectrum only)	This describes the size of a local number. The setting determines whether the local number requires additional information to dial out.
Indial number size (Spectrum only)	This describes the size of a number used to dial a local extension. The setting ensures that all indial numbers of this size do not require additional information before dialling.
Disable T309	T309 is a timer. If you enable it, and a failure occurs, calls are cleared after 2 seconds. The default is 'Disabled'. This parameter is not mandatory.
Do not use as reference clock	Enable this if you do not want the module to supply the reference clock.
Do not use as bus master	Enable this to prevent the module being selected to provide the backplane bus clock.
Do not use for D-channel calls	Enable this to prevent the Resource Manager or Control Module from delivering outgoing D-channel calls through the module.
Retain Layer 2 establishment	Enable this if you want to prevent the network from deactivating the link when it is idle. This is necessary if you want the module to provide constant clock.

Prefix called numbers (UK and ROW only)	Enable this to change the Group Number to the CPN (Called Party Number) prefix. Leave disabled if you want to use the Group Number. When enabled, the number entered in the CPN prefix is used to prefix the called party number received for incoming calls. Call control prefixes the received called party number before performing any call routing.
Restrict calling digits to (UK and ROW only)	Enable this if you want to limit the number of digits sent in the Calling Party Number. In the UK, the number sent as the Calling Party Number must be the same length as the number received from the network in the Called Party Number. The number of digits set (<i>n</i>), is the number of digits from the end of the line number that are sent as the Calling Party Number.
Unknown Number Type	This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Unknown'. This may be a requirement of some PBXs and exchanges.
Send Calling Party Number	Enable this if you want to send the Calling Party Number in the keypad information element rather than in the Called Party Number Information Element.
Exclusive/preferred use only	Enabling this removes the port from the TE table. The TE can then only be used for outgoing calls if the NT making the call has Outgoing TE exclusive selected.
Add progress to alerting	Enable to add a progress indicator to the incoming alerting message.
Point to point TE TEI assignment	Enable if operating in Point-to-point mode. The default is Automatic . Select Fixed and enter a number if you want to identify a specific connection endpoint within a service access point.
Protocol	The protocol defaults to the standard required for the selected area of operation. For example, TS13 and Spectrum in Australia, and ETSI in the UK. Only the protocols supported by your module will be available.

Spectrum protocol (where available)

Spectrum is a public switched network service that uses network intelligence to provide PABX-like facilities to customers who do not purchase a PABX. Spectrum handles a limited multidrop S-bus. It can handle only two terminals on one S-bus unlike TS13 which can handle eight.

The only noted difference between TS13 and Spectrum is the need for the negotiation process of a Service Profile Identification (SPID). This process identifies terminal equipment with certain capabilities to the network. The network can then accept outgoing calls and route incoming calls. If the SPID is not negotiated, no network call routing occurs.

The SPID number is given by the service provider at the time the network connection is entered. A SPID is related to a B-channel and a Terminal Endpoint Identifier (TEI). The relationship with the TEI is physical (one-to-one), but it is only logical with the B-channel. A SPID has a unique TEI value within the S-bus and can use any one of the B-channels available in that S-bus.

An ISDN port requires a SPID and a Bearer Capability to be programmed for each TEI (or B-channel) to be used. The Bearer Capability is used to filter the types of calls being made from the line. This is to ensure that the call type matches the line's capabilities.

With Spectrum, ISDN Basic Rate ports are capable of making only the type of calls dictated by the SPIDs. A port can have a SPID set for data, and the other port set for voice, but then the ports will be capable of making only one call of each type.

For the ISDN ports to operate correctly, the Spectrum SPID must be of a type called Initialising SPID and **not** a Non-Initialising SPID. An Initialising SPID identifies itself to the network through negotiating a SPID on startup. The module's ISDN ports are of the Initialising type.

If two SPIDs are to be used then the S-bus must be configured to have two initialising SPIDs. A Spectrum S-bus can have either two Initialising SPIDs or a combination of Initialising and Non-Initialising. This can be configured only by the service provider. It is essential that the SPID is configured correctly or the Spectrum connection will not work.

SPID configuration

- Number of SPIDs** Select the number of SPIDs you wish to configure — one or two.
- SPID *n*** Enter the SPID number for each line as supplied by your ISDN provider. If required, select Bearer Layer Capabilities to change these properties. See *Bearer Capability* on page 2-6 of the *Parameters* section for more information.

National ISDN 1 protocol (USA only)

NI1 is a commonly used North American ISDN Basic Rate protocol. The QBRM currently supports TE mode over S/T physical interfaces. Standard NT1s can be used for 'U' interfaces.

A major feature of NI1 protocol is the use of Service Profile Identification (SPID). A SPID is a nine to twenty digit number and is used to uniquely identify a particular set of subscribed service capabilities. The NI1 network has a one-to-one relationship between SPID, Directory Number, TEI and Bearer Capability. The modules support a maximum of two different Initialising SPIDs per port.

Some NI1 services support only one Call Type (CT) bearer per SPID, where as other services can support multiple CT bearers per SPID. In the latter, linesets are used and calls are matched on bearer capability.

SPID and Directory Numbers require a one-to-one mapping. For this reason, user ports should have the exclusive/preferred TE selection set to the appropriate NI1 port with the SPID. See page 2-19 of the *Parameters* section for details.

Configuring the default SPID

- Default SPID** The SPID is used as a default under 'cold start-up' conditions, for example, an installed, but yet unconfigured device. The default SPID allows remote access even when the RM or CM is not configured. The SPID value is stored in the module's non-volatile memory.

The SPID can also be programmed directly through the V.24 port on the module, without the need for an NMS connection. See the *Installation Manual* for details of configuring the default SPID.

Configuring an NT port

Maximum	Enter the maximum number in the range for this module.
Minimum	Enter the minimum number in the range for this module (it may be the same as the maximum number).

Notes

- *These are complete numbers; the base number is not applied.*
 - *Configuring the maximum and minimum numbers creates a lineset within the device. It contains the two B-channels and the D-channel as members.*
-

ISDN BRA NT Line Control defaults

Outgoing TE	Any.
TEI audit	Disabled.
Don't use as bus master	Disabled.
Limit DDI digits (UK and ROW only)	Disabled.
Point to point (UK and ROW only)	Disabled.

Changing ISDN BRA NT Line Control defaults

Outgoing TE	Select the required TE. If you select Preferred or Exclusive , you must then select the TE from the displayed list. This setting affects all the module's lines.
TEI audit	Enable this if you want to check the TEIs on a regular basis. The time is in minutes.
Don't use as bus master	Enable this to prevent the module being selected to provide the backplane bus clock.
Limit DDI digits (UK and ROW only)	Limit DDI restricts the number of digits that will be dialled into the PABX from the module. If this is not enabled, the full number received by the module will be forwarded. If the parameter is enabled, and a number 'n' entered, only the last 'n' digits of the number will be forwarded. This setting applies to the entire module, and is not line by line configurable.
Number type unknown	This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Unknown'. This may be a requirement of some PBXs and exchanges.
Point to point (UK and ROW only)	Enable this if the NT port is connected to a TE operating in a point-to-point mode.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls.

Configure individual lines

The complete line configuration for each of the B-channels and D-channel is available using this dialog; however, the configuration is not normally used, as the module's ISDN messages are derived from those coming from the TE.

Dual and Quad Basic Rate Module-U Interface (DBRM-U/QBRM-U)

The Dual and Quad Basic Rate Module-U Interface (DBRM-U and QBRM-U) provide multiple U-interface ports on a single card. Each interface can be configured by links on the board to provide the Network Termination (NT) function or the Terminal Equipment (TE) function. Any combination may be configured.

Note *The NT and TE functions referred to in this section are consistent with other NT and TE functions described in this manual. The functions do **not** define the physical NT and LT interfaces.*

The operation of the DBRM-U and QBRM-U is dependent on the Resource Manager.

Configuration

Select individual port types

When you create a new configuration in an off-line session you must enter the port type for each of the ports – either Basic Rate TE or Basic Rate NT. This can then be downloaded at some time in the future. Port configuration of NTs and TEs must match the module that is to be used.

Configuring the ports

For further configuration select the port you require. Configuration details for each port type follow.

Note *When a port is changed from TE mode to NT mode or vice-versa (by changing the hardware links) or when a module is replaced by one with a different port configuration, the original module must be deleted from the NMS database prior to insertion.*

Configuring a TE port

Numbering

Base Number

Enter the base number in local significant format (no area code). This is optional, but if entered, it must conform with other numbers in the ISDN range.

The base number is subtracted from an incoming called party number and the call is routed to the resulting number. It is added to the line number and the area code to produce the calling party number on outgoing calls.

If the base number is not entered, individual line numbers must be the full local significant number.

The base number allows lines to be accessed from different ISDN lines or multiple ISDN modules that have different number ranges.

Multiple TEs may share the same number range where incoming calls are distributed according to the criteria employed by the network provider.

Group Number

For Australia, the group number is the main directory number within the number range allocated by your ISDN service provider. It is shown on the label of the NT1 device to which the U-interface module is connected. Enter the group number in local significant format.

For ETSI, the significance and function of the group number depends upon the called party number presentation options available at the Basic Rate interface to which the U-interface module is connected. The types available from British Telecom are:

- **Simple**
No called number is provided to the TE port. All calls received through the module are presented with the Called Party Number prefix (CPN) as the called number. For example, if the CPN prefix is 123456, a call received will match a port with the line number 123456. In this case, the CPN prefix is mandatory. Enter the number that is used to address the Basic Rate Access.
- **Multiple Subscriber Number (MSN)**
Only the last digit dialled is provided to the TE port, which appends it to the CPN prefix to make up the Manager number. Enter the first five digits of the ISDN Access number. For example, if the MSN interface number range is from 245670 to 245679, the CPN prefix is 24567.
- **DDI**
The network provides a number of digits to the module. The number of digits is a subscription option. The CPN prefix is not used.

National Significant Area Code	Enter the National Significant Area Code. The National Significant Area Code is the area code defined by the ISDN provider, without the leading escape prefix.
Drop first n digit in incoming call	Use this parameter to remove any unwanted numbers or area code from the incoming call. Typically a SPID is a 10 digit entity. If a called number is less than 10 numbers, padding must be added to make up the 10 digits.
Options	
Disable T309	T309 is a timer. If you enable it, and a failure occurs, calls are cleared after 2 seconds. The default is 'Enabled'. This parameter is not mandatory.
Do not use as reference clock	Enable this if you do not want the module to supply the reference clock.
Do not use as bus master	Enable this to prevent the module being selected to provide the backplane bus clock.
Do not use for D-channel calls	Enable this to prevent the Resource Manager from delivering outgoing D-channel calls through the U-interface module.
Retain Layer 2 establishment	Enable this if you want to prevent the network from deactivating the link when it is idle. This is necessary if you want the module to provide constant clock. The default is 'enabled'.
Exclusive/preferred use only	Enabling this removes the module from the TE table. The TE can then only be used for outgoing calls if the NT making the call has Outgoing TE exclusive selected.
Add progress to alerting	Enable to add a progress indicator to the incoming alerting message.

Prefix called numbers	Enable this to change the Group Number to the CPN (Called Party Number) prefix. Leave disabled if you want to use the Group Number. When enabled, the number entered in the CPN prefix is used to prefix the called party number received for incoming calls. Call control prefixes the received called party number before performing any call routing.
Point to point TE	Enable if operating in Point-to-point mode and Restart procedures are not supported.
Send Calling Party Number	Enable this if you want to send the Calling Party Number in the Called Party Number Information Element. This is on by default
Restrict calling digits to	Enable this if you want to limit the number of digits sent in the Calling Party Number.
Unknown Number Type	This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Unknown'. This may be a requirement of some PBXs and exchanges.

TEI assignment

The default is **Automatic**. Select **Fixed** and enter a number if you want to identify a specific connection endpoint within a service access point.

Protocol

The protocol defaults to the standard required for the selected area of operation. You should select only the protocols supported by your service.

Configuring an NT port

Maximum	Enter the maximum number in the range for this module.
Minimum	Enter the minimum number in the range for this module. (It may be the same as the maximum number.)

Notes

- *These are complete numbers; the base number is not applied.*
 - *Configuring the maximum and minimum numbers creates a lineset. It contains the two B-channels and the D-channel as members.*
-

ISDN BRA NT Line Control defaults

Card options

Outgoing TE	Any.
TEI audit	Disabled.
Limit DDI digits	Disabled.
Don't use as bus master	Disabled.
Point to point	Disabled.
Protocol	The protocol defaults to the standard required for the selected area of operation.

Changing ISDN BRA NT Line Control defaults

Card options

Outgoing TE	Select the required TE. If you select Preferred or Exclusive , you must then select the TE from the displayed list. This setting affects all the module's lines.
TEI audit	Enable this if you want to check the TEIs on a regular basis. The time is in minutes.
Limit DDI digits	Limit DDI restricts the number of digits that will be dialled into the PABX from the U-interface module. If this is not enabled, the full number received by the module will be forwarded. If the parameter is enabled, and a number 'n' entered, only the last 'n' digits of the number will be forwarded. This setting applies to the entire module, and is not line by line configurable.
Don't use as bus master	Enable this to prevent the module being selected to provide the backplane bus clock.
Number type unknown	This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Unknown'. This may be a requirement of some PBXs and exchanges.
Point to point	Enable this if the NT port is connected to a TE operating in a point-to-point mode, or if the TE is a member of this series of equipment that is using the Callback facility.
Protocol	The protocol defaults to the standard required for the selected area of operation. You should select only the protocols supported by your service.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls.

Configure individual lines

The complete line configuration for each of the B-channels and D-channel is available using this dialog; however, the configuration is not normally used, as the ISDN messages are derived from those coming from the TE.



E1 Digital Module (E1M)

The E1 Digital Module (E1M) operates as an Network Termination (NT) module and provides one 2.048 Mbit/s NT port for connecting 2.048 Mbit/s terminal equipment, such as PABXs, video conferencing equipment and Local Area Network (LAN) bridges.

The E1M supports both fixed (leased line) and switched calls. It can be configured in two mutually exclusive modes:

- Channel transparent which enables timeslots 1 to 31 to be transferred over ISDN B-channels
- Channel Associated Signalling (CAS) which provides 30 data channels with channel 16 used for signalling between a PABX and the E1M.

In CAS mode, the E1M interprets the call control signalling from the terminal equipment and translates it for use by ISDN modules.

The E1M derives its reference clock from the chassis internal Bus clock. This provides synchronisation to connected terminal equipment.

Configuration

CAS Enable Enable this for PABX switched calls and tie lines that use CAS. CAS provides 30 data channels with channel 16 used for signalling between the terminal equipment and the E1M.

OPS Mode Enable this if the E1M is connected to an AT&T PABX running OPS (Off Premises Station) mode on the connected E1 trunk.
With an ALEM at the other end of the link, OPS signalling information can be transmitted between the two, allowing internal and external ring signals to be passed on to the customer handsets.

CRC Enabled Enable this if your ISDN terminal equipment requires Cyclic Redundancy Check (CRC) to monitor lines. You would not normally need to change the default (disabled).
This enables CRC-4 error checking at Layer 1. It is always transmitted. Enabling this function causes CRC-4 data to be the Layer 1 line monitoring criteria. If it is disabled, framing errors are interpreted as Layer 1 line errors.

The settings for the above parameters affect the operation of the entire module. Having configured the module, you can select individual lines and configure them as required.

Do not use as reference clock Select this option if you do not require the E1M to supply the clock reference to synchronise the device to the line.

Do not use as bus master Enable this to prevent the E1M being selected to provide the backplane bus clock.

Force idle output Enable this to provide a call to the trunk when a SEIZE is detected from the PABX.

The call clears when the PABX signal changes from SEIZE to any other value (normally IDLE). The output signal remains IDLE to the PABX regardless of the call state.

This is intended for PABXs (NEC) that use other methods for signalling (for example V.24) but still provide SEIZE to the trunk channels when they are used. The provision of IDLE as a back signal ensures the PABX accesses the channel. Other signals indicate a fault status and prevent the PABX from selecting the channel. If the line is faulty, a FAULT signal is applied. CAS signalling needs to be enabled for this option.

TL7 OPS Selecting this implements the ERICSSON TL7 version of the Off Premises Station protocol, as an alternative to the AT&T (ANSI) version.

The OPS feature is designed to transfer the ring cadence from an E1M-connected PABX to a remote ALEM-connected phone. It then transfers the hook status from the phone to the PABX, thus providing all the PABX services to a remote phone station.

Protocol

Select to display the following protocols:

- R2D/T6 – the default
- R2D/T6 In Only
- R2D/T6 Out Only
- ARD – used for Channel Bank applications (see page 5-104)
- OPS (Station Side)
- OPS (PBX Side)
- OPS TL7/P7 (Station Side)
- OPS TL7/P7 (PBX Side)
- MRD – used for Channel Bank applications (see page 5-104).

Line Configuration

Each channel (1–31 if CAS not enabled, 1–15 and 17–31 if CAS enabled) can have a live configuration.

Line Number Enter the line number if required.

For outgoing calls, or if the line is a member of a lineset, this parameter can be left blank.

Subaddress Enter the subaddress if required. It is not mandatory to enter anything in this parameter.

Port ID Enter the Port ID if required. It is not mandatory to enter anything in this parameter.

Port Name Enter the port name if required. It is not mandatory to enter anything in this parameter.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. The following table shows the default settings.

Bearer Capability defaults

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital

This parameter is important when configuring the E1M as it determines whether analog or digital line capability is provided by the network.

Note *Please refer to your PABX user documentation before changing to analog line capability.*

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults are enabled for incoming calls and disabled for outgoing calls. For further information about the default settings, refer to page 2-7 of the *Parameters* section of this manual.

ISDN Interface — High Layer Compatibility

These parameters are not normally used. Refer to page 2-10 of the *Parameters* section of this manual for further information.

ISDN Interface — Call Establishment Modes defaults

The following table shows the Call Establishment Modes defaults.

	Parameter	Default
Semipermanent (where available)	Enable	Cleared
	Label	—
Hotline	Enable	Cleared
	Number	—
	XSPC	Cleared
	Subaddress	—
DDI (Indial)	Enable	Selected (E1MN) Cleared (E1MT)
	Ignore first	4 digits (E1MN) 0 digits (E1MT)
If Channel is Subrate, It's	Voice (use compander)	Selected
	Data (use Subrate switch)	Cleared
Call Control	Auto Answer	Cleared
	Connect after indialling digits	Cleared
Signalling	National Significant Remote Number	—
	End-to-End	Cleared
	E2E answer xfer	Cleared
	TS0 E2E xfer	Cleared

Changing Call Establishment Modes defaults

The settings for these parameters control the way in which calls are made. You can change the following settings to meet your requirements.

Semipermanent (where available) Enter the semipermanent number in the Label parameter and select **Enable**. Valid for CAS and non-CAS.

Hotline Enter the Hotline number and the subaddress number, if required. Select **Enable**. Valid for CAS and non-CAS.

You can also select **XSPC** to enable a Switched Semipermanent Connection. This is only available if Hotline is enabled.

DDI (Indial) Select **Enable** and enter the number of digits to be ignored. For example, if the call is for extension 23 of the telephone number 8096923, the number of digits to ignore is five.

This is only available in CAS mode.

If Channel is Subrate, It's Select data if the channel is being used for framed data rather than A-law voice.

- Auto Answer** Select this to connect the incoming call to the line as soon as it is received, irrespective of the signalling state of the connected equipment.
- When a tie line has end-to-end enabled, select **Auto Answer** to connect the end-to-end tie line call request rather than returning the connected equipment's signalling indications.
- This is only available in CAS mode.
- Connect after indialling digits** Select to connect the incoming call after the digits have been indialled to the connected equipment, rather than providing a *ring* indication to the ISDN. This is used in conjunction with DDI (Indial) and is only available in CAS mode.
- Signalling** Enter the national significant remote number in the number parameter for end-to-end signalling applications, if required.
- This is only available in CAS mode.
- End-to-End** Enable this where the line is to be used in equipment tie line applications, and CAS mode is required.
- E2E Answer Transfer** Select to provide an end-to-end answering mechanism to support some inter-PABX operations. It provides for the answering of an ISDN switched call followed by the end-to-end transfer of an Answer signal from the called terminal equipment back to the calling line. This is only available in CAS mode.
- TS0 E2E xfer** Timeslot 0 End-to-End Transfer is only available if you select End-to-End first. Select this to enable the end-to-end transfer of timeslot 0 information in any one of the lines (1-31) which has a call to the destination E1M established.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E164
	Hotline	E164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

- AOC (where available)** Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

- Calling Line ID** Do not select **CLIP** (CLI Presentation).
If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.
- CLIVE List Assignment** Select this option to apply a CLI/Number checking list to this line. See page 3-49 of the *System Configuration* section for more information.
- Numbering format** Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

- Teleservice (where available)** None.

ISDN Internal Control defaults

- Restriction** None.
Outgoing TE Any.

Changing ISDN Internal Control defaults

- Restriction** Select **Disable** to put the line into a fault state so that it will not be used by the PABX. If CAS is enabled and the line is configured as a tie line with end-to-end signalling enabled, **Disable** will send a blocking signal to both ends of the connection.
- Outgoing TE** Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.



T1 Digital Module (T1M)

The T1M provides a single T1 port for voice and data communications. The T1 port provides 24 channels, each having 64 or 56kbit/s bandwidth, depending on its configuration.

The T1M has the following Layer 1 features:

- The port can be configured for ESF or SF/D4 operation
- The zero code suppression can be configured for B8ZS or ZCS, or it can be disabled
- The transceiver gain can be configured to suit the user's line properties.

The T1M supports the creation of Trunk Groups made possible by independent channel configuration. Each channel can be configured for transparent 64 kbit/s, which would suit leased-line data applications such as video-conferencing. Alternatively, the channels can be set up for Robbed Bit Signalling (RBS) which is embedded in each channel for switched call control. The following protocols that utilise RBS are supported:

- E&M Wink Start
- E&M Immediate Start
- FXO Ground Start
- FXS Ground Start
- FXO Loop Start
- FXS Loop Start
- PLAR.

The T1M supports switched and end-to-end call signalling. Either can be used with any of the above protocols. End-to-end signalling can be used in conjunction with a switched call (XSPC) or a semipermanent call (SPC).

T1M configuration

Card options

- Don't use as reference clock** Select this option to prevent the T1M supplying the reference clock to the device.
- Don't use as bus master** Select this option to prevent the T1M supplying the bus clock to the device.
- Release on Progress** Enabling this option releases a call even if a Progress Indicator is sent. In this instance, any clearing tones will not be heard.
This option is useful when using message recording equipment, as clearing tones are not recorded at the end of a message.

Framing

- SF (D4)** Enables Superframe Framing. The superframe consists of 12 TDM frames. It supports A, B robbed-bit signalling. It has no Facility Data Link (FDL).
- ESF** Enables Extended Superframe Framing. The extended superframe consists of 24 TDM frames. It supports A, B, C and D robbed-bit signalling as well as a 4 kbit/s Facility Data Link (FDL).

Zero suppression

- None** When selected, 1s (ones) are represented on the Digital Line Interface (DLI) by alternating positive and negative pulses. Zeros are represented by zero signal levels. This is known as Alternate Mark Inversion (AMI) coding.
- ZCS** This enables Zero Code Suppression. This is similar to AMI but channels containing all zeros have bit seven forced high prior to transmission. This ensures that sufficient pulse density is on the line to maintain clock synchronisation. This results in data corruption as there is no mechanism to remove the forced bit.
- B8ZS** This enables Bipolar Eight Zero Substitution. This is similar to AMI, but if eight zeros occur in a row, bipolar violations are inserted to ensure that sufficient pulse density is on the line to maintain clock synchronisation. The T1 Framer recognises these violations and removes them, thus no data corruption occurs.
- DLI gain** Select Line Build Out (LBO) options for long haul or short haul operation. Long haul options include pulse shape and gain selection. Short haul options are selected on length of cable.

Line Configuration for T1M

Each channel can have a live configuration.

- Line Number** Enter the line number if required.
For outgoing calls, or if the line is a member of a lineset, this parameter can be left blank.
- Subaddress** Enter the subaddress if required. It is not mandatory to enter anything in this parameter.
- Port ID** Enter the Port ID if required. It is not mandatory to enter anything in this parameter.
- Port Name** Enter the port name if required. It is not mandatory to enter anything in this parameter.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. The following table shows the default settings.

Bearer Capability defaults

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital

This parameter is important when configuring the T1M as it determines whether analog or digital line capability is provided by the network.

Note *Please refer to your PABX user documentation before changing to analog line capability.*

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults are enabled for incoming calls and disabled for outgoing calls. For further information about the default settings, refer to page 2-7 of the *Parameters* section of this manual.

ISDN Interface — High Layer Compatibility

The following parameters are the commonly used ones for the T1M. Refer to page 2-10 of the *Parameters* section of this manual for further information.

ISDN Interface — Call Establishment Modes defaults

The following table shows the Call Establishment Modes defaults.

	Parameter	Default
Subrate Virtual Line member or manually connected	Enable	Cleared
	Label	—
Hotline	Enable	Cleared
	Number	—
	XSPC	Cleared
	Subaddress	—
DDI (Indial)	Enable	Selected
	Ignore first	0 digits
If Channel is Subrate, It's	Voice (use compander)	Selected
	Data (use Subrate switch)	Cleared
Call Control	Auto Answer	Cleared
Signalling	National Significant Remote Number	—
	End-to-End	Cleared
	Interpreted	Selected

Changing Call Establishment Modes defaults

The settings for these parameters control the way in which calls are made. You can change the following settings to meet your requirements.

Subrate Virtual Line member...

Enter the required number in the Label parameter and select **Yes**.

Hotline

Enter the Hotline number and the subaddress number, if required. Select **Enable**.

You can also select **XSPC** to enable a Switched Semi Permanent Connection. This is only available if Hotline is enabled.

- DDI (Indial)** Select **Enable** and enter the number of digits to be ignored. For example, if the call is for extension 23 of the telephone number 8096923, the number of digits to ignore is five.
- If Channel is Subrate, It's** Select data if the channel is being used for framed data rather than voice.
- Auto Answer** Select **Auto Answer** if incoming calls from the backplane are to be automatically answered.
- Signalling** Select **End-to-end** to enable an out-of-band signalling path to be provided with calls on this line. The out-of-band signalling path carries call information to, and from, the far end of the call.
- Select **Interpreted** to enable interpreted end-to-end signalling (IDAS). Clear to enable uninterpreted end-to-end signalling (UDAS).
- Enter the **Group Number** of the ISDN link connected to the remote device. This number is not required when the end-to-end connection is an switched call.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (Where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E164
	Hotline	E164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

- AOC (Where available)** Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via Call Log within OmniVision. For further details, refer to the *OmniVision User Manual*.
- This is only available if you subscribe to an AOC service.
- Calling Line ID** Do not select **CLIP** (CLI Presentation).
- If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.
- CLIVE List Assignment** Select this option to apply a CLI/Number checking list to this line. See page 3-49 of the *System Configuration* section for more information.
- Numbering format** Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

Teleservice None.

ISDN Internal Control defaults

Restriction None.

Dial Type DTMF.

PCM Data Normal

Protocol Protocol - None.

Glare resolution - not supported in this release.

Outgoing TE Any.

Changing ISDN Internal Control defaults

Restriction Select **Disable** to put the line into a fault state so that it will not be used by the PABX. If RBS is enabled and the line is configured as a tie line with end-to-end signalling enabled, **Disable** will send a blocking signal to both ends of the connection.

Dial Type Select either **DTMF** or **Pulse**.

PCM Data Select **Invert** to logically invert the data on this line as it is passed to the T1 trunk, and as it enters the T1M. Select **Normal** to leave the data polarity unaltered.

Protocol Available protocols are:

- **None** - RBS is disabled. Connections manually established.
- **E&M Wink Start**
- **FXS Ground Start**
- **FXO Ground Start**
- **FXS Loop Start**
- **FXO Loop Start**
- **E&M Immediate Start**
- **PLAR**.

Glare Resolution is not supported in this release.

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.



E1M-DPNSS

The E1M-DPNSS Module enables PABXs and other equipment that support DPNSS (Digital Private Network Signalling Scheme) to connect to other networks.

The E1M-DPNSS can use the full range of proprietary networking features including SPCs, MicropleX switching, and private and public networks to carry DPNSS traffic. There is an additional feature specifically for the E1M-DPNSS. This allows private to public number mapping and substitution/replacement. The signalling for the feature transparency can be passed over the D-channel, or over the B-channel using the Inband Signalling feature.

The E1M-DPNSS supports two modes of operation:

- Transparent — where two or more DPNSS PABXs communicate using DPNSS
- Gateway — where a DPNSS PABX communicates with other non-DPNSS equipment such as a telephone.

See page 5-55 in the *Applications* section for information on a typical DPNSS application.

Operation

The way in which DPNSS operates varies with the application scenario. In DPNSS to DPNSS connections, for instance, the module provides a fully featured DPNSS end-to-end connection. Where the connection involves non-DPNSS devices, the module provides mapping of a subset of DPNSS call facilities onto the non-DPNSS device signalling.

The DPNSS also has to cater for the different methods used by networks for signalling and transporting supplementary services. The module supports DPNSS in the same way as the other end-to-end signalling schemes and the rules for D-channel or B-channel end-to-end signalling apply.

The E1M-DPNSS collects dialled digits from the DPNSS equipment. Using this information, it decides how to route the call and determines whether the destination is DPNSS or not. If it is DPNSS, the call is Transparent, if not, it is a Gateway call.

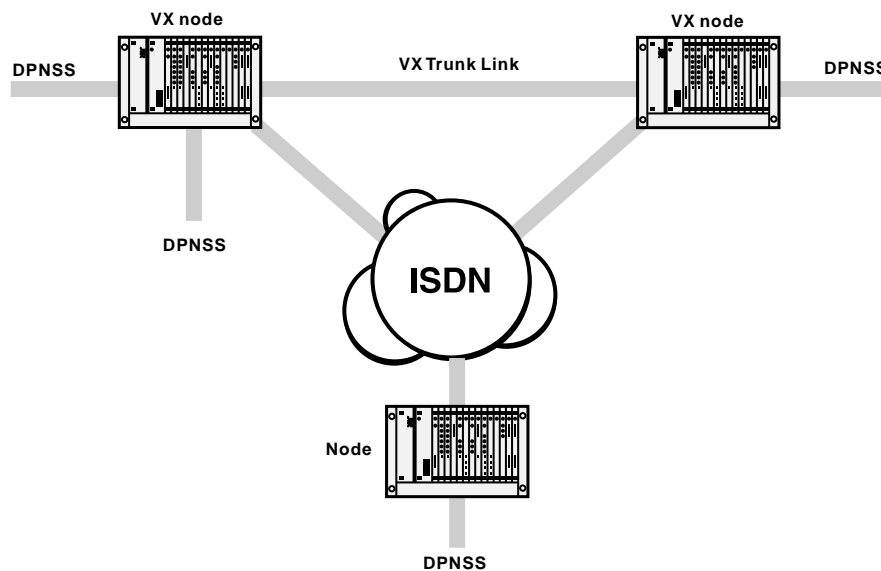
The module can distinguish between route-optimised calls and normal calls, and will automatically connect the bearer of route-optimised calls. This limits extra call charging to only route-optimised calls.

The indication of the type of call is sent back to the DPNSS equipment by the E1M-DPNSS using the Called Line Category supplementary service string. CLC-ORDinary indicates DPNSS and CLC-DASS2 indicates non-DPNSS.

DPNSS supports both Real Calls and Virtual Calls. Real Calls can be Transparent or Gateway, Virtual Calls must be Transparent. Attempts to make Virtual Calls to non-DPNSS destinations are rejected.

Transparent operation

If two E1M-DPNSS modules are inter-operating across a private network or an ISDN, they are said to be operating Transparently. This would be the case when the two modules are in the same device or in two separate devices, connected across a network. Transparent operation is possible only with VX systems. The figure below shows a typical application using the transparent mode of operation.



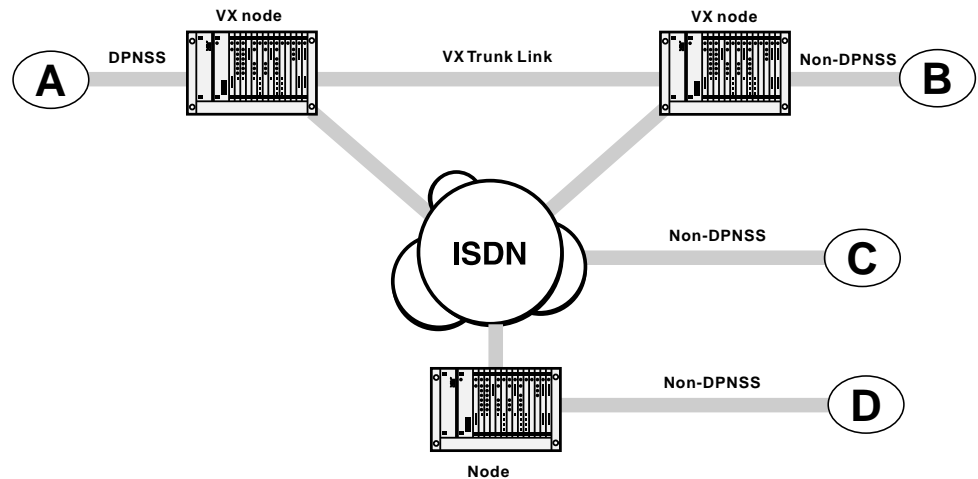
The passing of DPNSS Supplementary Services is achieved using end-to-end signalling over either the VX Trunk Link or the public ISDN connections. When the ISDN does not support User-To-User messaging, the In-Band Signalling feature is used to pass the end-to-end signalling within a B-channel. Connections may transit both the VX Trunk Link and the public ISDN. This may occur in the above example where one of the ISDN connections was down.

A call is deemed to be Transparent (DPNSS end-to-end) if the source is DPNSS and the (translated) destination number is in the Router Table. Once the decision is made that a call is Transparent DPNSS, the device will analyse the route to the destination to decide how to carry the call. If there is a signalling path associated with the call, that is, it is over a VX IPMN-IPMN link or a MicropleX switched link, then this signalling path will be used to carry the DPNSS supplementary services. If there is no signalling path associated with the call, the device will make a B-channel call to carry the actual call traffic. It will use the information in the Router Table to set up a separate signalling channel to carry the DPNSS supplementary services. The B-channel call and the signalling call are uniquely tagged to allow the destination to reconstruct the original DPNSS call, complete with supplementary services.

Virtual Calls use the Router Table signalling path only.

Gateway operation

If E1M-DPNSS modules are inter-operating with non-DPNSS devices they are said to be operating in a Gateway mode. This mode of operation is supported when the connection is across private network links or public switched networks. The figure below shows a typical example of the Gateway mode of operation.



In this example, the equipment supporting [A] is operating in Gateway mode. The non-DPNSS devices can be connected to any appropriate interface [B] or [D] or other access mechanism [C].

The level of functionality in this mode depends on the commonality between DPNSS and the call signalling protocol of the non-DPNSS devices. An additional requirement is that the DPNSS device has a transmission format that is common to the non-DPNSS device (voice, data and so on).

Interfaces that support this mode of operation include:

- ISDN Basic Rate and Primary Rate interfaces;
- G.703/704 CAS R2D;
- 4 Wire E & M SSDC5;
- 2 Wire Ring-In/Loop-out and Ring-out/Loop-in;
- Data interfaces (V.24, V.35 and X.21);
- DTM.

When the destination is non-DPNSS, a Gateway call is made. This translates the DPNSS call signalling into ISDN call signalling and the basic supplementary services into ISDN ones. Supplementary services mapped are shown in the following table:

DPNSS	ISDN
OLI - Originating Line Identity	CLI - Calling Line Identity
CH-UU Charge Reporting - Units Used	CCI - Call Charge Information (DASS2) AOC - Advice of Charge (Q.931)
SIC - Service Identifier Code	SIC - Service Identifier Code (DASS2) Bearer/LLC (Q.931)
NPR-A - Number Presentation Restriction - "A" Party (Calling Party).	CLIR - Calling Line Identification Restricted (Q.931)

The destination interface translates the ISDN signalling into its native signalling and forwards any supported supplementary services.

Gateway Call Diversion

RM

This provides support for interworking between DPNSS and other signalling systems for the Call Diversion Supplementary service. This feature allows DPNSS Gateway calls that originate from a non-DPNSS user to participate in Call Diversion.

Note *BTNR 189 defines the required interworking for a Gateway PBX, and Section 11 defines the interworking support for Call Diversion. Our implementation of Call Diversion conforms to this section of the specification.*

Diverted calls originating from PBX extensions in a DPNSS network are routed to the diverted number. However, calls originating from a non-DPNSS user have to rely on the J5000, in its role as 'gateway PBX', to perform the call diversion to either another PBX extension or to a non-DPNSS user.

It is also possible to chain diversions together. To avoid unlimited chaining, a maximum length of three diversions is supported.

Gateway Call Diversion is implemented on a call by call basis and can be configured in conjunction with Feature Transparency. Any call destined for the configured E1M-DPNSS has the ability to participate in Call Diversion.

For more information on the parameters associated with this feature, see *Gateway Call Diversion* on page 4-57.

Under certain circumstances, number manipulation may be required. Normally, the number presented by the DPNSS network in the divert string is complete and requires no modification. If a modification is required, you can use *CLI/Number Checking* (see page 3-49) and *ISDN Interworking* (see page 3-26) to manipulate the number. A diverted call is treated as any other call for routing purposes.

If number manipulations become complex, you can also trunk prefix diverted calls. This option follows all the same rules as trunk prefixing for normal calls. For information on the parameters associated with *Trunk Prefixing* see page 4-58.

Intermediate Nodes

If a node receives a DPNSS call that is already tagged as a Transparent Call, the node checks to see whether the destination is in its router table. If the device has the destination in its router table, it knows that it is just an intermediate node for the call and will pass the call on unchanged. If the destination is **not** in the router table, the device assumes that the call terminates in this node and it will attempt to reunite the signalling and B-channel calls.

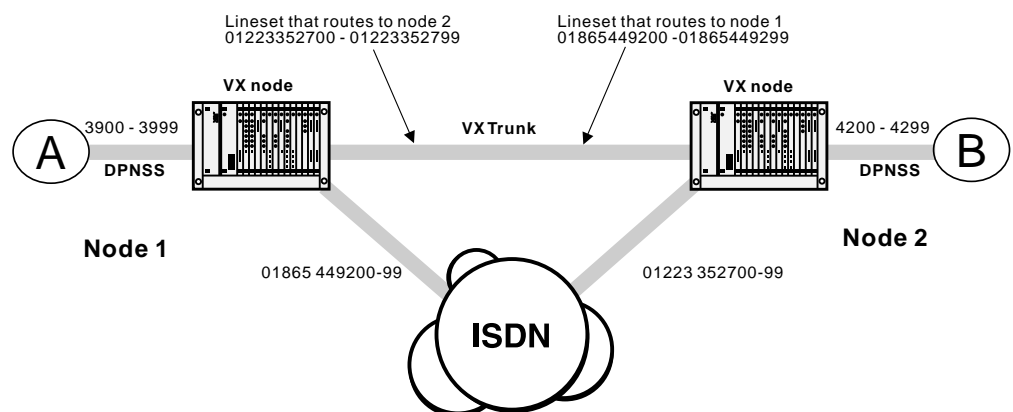
Numbering

Typically a DPNSS link between two PABXs will use the PABX's private numbering scheme. This will usually be 3 to 6 digit numbers that bear no relevance to any public network numbering.

A typical network will have a mix of public and private links, which must carry the PABX private numbering transparently.

The rule when configuring networks with DPNSS is to configure the entire network using public numbering, and use the number translation tables to translate the PABX private numbering into a public number that can be used to reach the destination device.

The following diagram shows this:



PABX A in Node 1 dials 42xx which is an extension on PABX B in Node 2. 42xx matches a number table in the Node 1 VX and is replaced by 01223352700. All routing decisions are then based on this public number.

First preference is a lineset in Node 1 VX that routes the call over the VX trunk. If the VX trunk is full or broken, the call will route over the public ISDN. This gives full backup and overflow functionality for the VX trunks without any further number manipulation.

When the call arrives at the Node 2 VX, it is recognised as a DPNSS call that has undergone number replacement, and the original 42xx is replaced before final routing to PABX B.

Additional digits dialled after the 42xx will be passed across to node 2 over the signalling channel and sent on to PABX B.

Note *This is significantly different to previous E1M-DPNSS configurations which used additional interworks, linesets and trunk prefixing to achieve a similar end result.*

These previous methods are now obsolete and no longer supported. Configurations using these previous methods may still work, but we recommend that wherever possible, configuration are updated to use the number tables for private/public substitutions and public number routing.

DPNSS Configuration

Configuring a system that uses the E1M-DPNSS is a four stage process.

- Stage 1 — Configure the E1M-DPNSS
- Stage 2 — Configure the number translation
- Stage 3 — Configure the signalling routing tables
- Stage 4 — Configure the bandwidth for the calls.

E1M-DPNSS

This section describes how to configure the E1M-DPNSS.

Number range The number range selects which range of numbers received by the device will be delivered to the E1M-DPNSS.

Where number translation is used, the **Private Number Range** of the PABX will go to the DPNSS module, and **not** the range of public numbers that are delivered to the device.

Enter the number range. The number range need not be the full length of the dialled number. For example, 31-39 will match any number beginning with 31 through to 39. This creates a lineset which includes all channels of the E1M-DPNSS. Frequently however, different number ranges will be applied to different channels. In these cases use the **Lineset** configuration in OmniVision to configure them.

Private/Public Number Clashes

You should normally configure Linesets with the **Local** attribute set. The exception to this is when there is a conflict between the private number range and public numbers that are required to be routed within the device. For example, the PABX number range may be 42xx, but the device is required to route 420-0100 to a different interface. In this case, you should set the DPNSS lineset to 4200-4299 and enable the **Exact Number Length** attribute. This will prevent 4200100 matching the set which it would otherwise do.

Channel Selection Order

Entering the card number range creates a 'Local' lineset with all 30 DPNSS channels, in ascending order. If the PABX normally selects channels for outgoing calls in ascending order, the E1M-DPNSS lineset should list the DPNSS channels in descending order. This will minimise channel collisions between incoming and outgoing calls.

See *Linesets* on page 3-17 for information on reversing the order of the E1M-DPNSS lineset.

Configure other parameters

Enable/Edit line

The default configuration of the E1M-DPNSS is for all 60 Channels (30 real call channels and 30 virtual call channels) enabled. Individual channels can be enabled or disabled as required.

Select the channels that are to be enabled.

Note Channels are always paired, that is, one real and one virtual. Real channels are numbered 1-31 and virtual channels 33-63. Enabling channel 1 enables real channel 1 and virtual channel 33. Disabling channel 2 disables real channel 2 and virtual channel 34, and so on.

Card options

Outgoing TE

Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

CRC enable

Enable this if your ISDN terminal equipment requires Cyclic Redundancy Check (CRC) to monitor lines. You would not normally need to change the default (disabled). This enables CRC-4 error checking at Layer 1. It is always transmitted. Enabling this function causes CRC-4 data to be the Layer 1 line monitoring criteria. If it is disabled, framing errors are interpreted as Layer 1 line errors.

Do not use as reference clock

Select this option if you do not require the E1M-DPNSS to supply the clock reference to synchronise the device to the line.

The clock reference to synchronise the device is supplied by an ISDN TE module. If no TE module is present in the chassis and there is no other source of a clock reference, you can configure the E1M-DPNSS to supply the device with a reference clock obtained from the terminal equipment to which it is connected.

The settings for the above parameters affect the operation of the entire module. Having configured the module, you can select individual lines and configure them as required.

Do not use as bus master

Enable this to prevent the E1M-DPNSS being selected to provide the backplane bus clock.

Applications...

On Startup

The E1M-DPNSS has three options on startup:

DPNSS-CPE Establishes

This is the default. This allows the PABX to establish the LAPs and clear any calls that might be up.

Establish LAPs	The E1M-DPNSS will attempt to establish data links for all enabled channels, but not reset any calls.
Establish LAPs and send CRMs	<p>The E1M-DPNSS will establish data links and then send a CRM (Clear Request Message) on each channel which will clear any stray calls.</p> <p>We recommend that this setting is left at 'DPNSS-CPE Establishes' unless the DPNSS PABX does not establish data links. If the Link LED on the front of the E1M-DPNSS does not come on once the DPNSS PABX is connected and enabled, change to 'Establish LAPs'. If the Link LED comes on, but the Call LED stays active, change it to 'Establish LAPs and send CRMs'.</p>
	Indial digits
Limit DDI	Limit DDI restricts the number of digits that will be dialled into the DPNSS PABX from the E1M-DPNSS. If this is not enabled, the full number received by the E1M-DPNSS will be forwarded. If the parameter is enabled, and a number 'n' entered, only the last 'n' digits of the number will be forwarded. This setting applies to the entire module, and is not line by line configurable. This is only likely to be used if number translation is not used.
	Dialled digits
Minimum number length	<p>When enabled, this determines how many digits the E1M-DPNSS collects before forwarding the call to the device for routing. It is important to set this to the same as the length of the lineset range configured for the destination E1M-DPNSS. This should also be the same length as the numbers entered into the Number Translation lists, or the router table if Number Translation is not used.</p> <p>For example, if the PABX is dialling numbers in the range 4200 to 4299, then the Number Translation list would be configured to 4200-4299, and the Minimum number of dialled digits should be set to 4.</p> <p>This setting is only useful where a PABX sends individual digits as they are dialled. In a lot of cases, the PABX will collect digits itself and not forward the call to the DPNSS link until it has collected 4 digits. If this is the case, then this setting will be ignored.</p>
	Options
Idle fill Layer 2	In normal operation, DPNSS Layer 2 has flag characters between Layer 2 frames. This is known to cause problems with some equipment which can indicate Layer 2 CRC errors when flag fill is used. If an abnormal number of Layer 2 CRC errors are reported, enable this parameter. Equipment known to exhibit this characteristic includes the NorTel DMS-100 (FeatureNet in UK).
Ignore NSI strings	Select this to disable NSI (Non Specified Information) strings. A particular PABX may send NSI strings as proprietary information to another PABX of the same type. These strings may not be required for normal operation.
Add progress to alerting	Enable to add a progress indicator to the incoming alerting message.
Convert progress to NAM (Number Acknowledgement Message)	Select this to enable a more robust operation of calls from a DPNSS PABX to an EMM that is using IDAS.

RM**Gateway Call Diversion**

When enabled, the module implements DPNSS interworking of the Supplementary Service Call Diversion (BTNR 189 Section 11). This allows DPNSS Gateway calls that originate from a non-DPNSS user to participate in Call Diversion.

Note *The E1M-DPNSS module, when operating as a Gateway PBX, only provides those features required to implement the supplementary service Call Diversion*

Multiple NIMs

When enabled (the default), all NIMs (Network Indication Messages) from the peer are passed End-to-end. This parameter can be enabled for all DPNSS applications. NIMs are used to notify the originating PBX of events that take place during the routing phase of the call, for example, 'Alternative route encountered'.

Do not send progress for cause 34

When enabled the Progress Indicator is not sent to the peer. This allows the peer to do a resubmission.

Delay on Route Optim Calls

By default, a delay of 3 seconds is used for a Route-optimised call using a Number Acknowledgement Message (NAM). This allows backward compatibility. Disabling this option removes the 3 second delay.

Feature transparency

When enabled, all supplementary services are passed End-to-end. When disabled, supplementary services are not passed and only gateway mode is supported.

DPNSS disconnect to NAM mode

When enabled, a Number Acknowledgement Message (NAM) is sent to the peer when the DPNSS module receives a disconnect message from Call Control. The NAM opens the voice path so that the disconnect tone generated by the VX can be detected by the PABX.

Convert NIM (*51*4#) to Alerting with PI

When enabled, a Network Indication Message (NIM *51*4#) sent from the peer is converted to 'Alerting with Progress Indicator'. PABXs usually generate this type of NIM when sending a call to the PSTN or ISDN. The conversion opens the voice path so that tones and Recorded Voice Announcements can be heard.

DPNSS binary encoding

When enabled, the DPNSS module supports binary encoding of the DPNSS supplementary service Network Address Extension-Destination Complete (NAE-DC). This may be needed to allow a called party sub address to be passed through the DPNSS network if PABXs in the network are set for strict syntax checking.

DPNSS configuration**A-B / X-Y**

When two pieces of equipment are connected using DPNSS, one piece of equipment is designated A and the other B. Similarly they can be designated X and Y respectively. Normally, A and X are configured together, and B and Y together. The E1M-DPNSS can be configured in any combination of A-B-X-Y but should normally be configured to be the opposite to the equipment to which it is connected.

Satellite / Terrestrial

This should be set to 'Satellite' only if the link between the E1M-DPNSS and the DPNSS equipment uses a satellite link. This setting increases timeouts to allow for the additional delay caused by the satellite link. Normally this should be left at the default 'Terrestrial' setting.

Call establishment modes

No VX Tones Normally the Virtual eXchange provides service tones for calls that switch via the VX. In some circumstances however it may be desirable to disable these tones and allow the connected PABXs to supply tones. For example, if all calls on a particular channel will definitely go to a specific PABX destination and that PABX will provide the service tones.

VX Tones can be disabled for calls originating from DPNSS channels by enabling the 'No VX Tones' parameter.

Add Trunk Prefix Gateway calls that originate from a non-DPNSS user are prefixed when this option is enabled and a Prefix is entered. This operates for normal calls. To add a Trunk Prefix to calls that have undergone Call Diversion, enable **Diverted**. You can enable both normal and diverted, none or one of the options. When using both normal and diverted, the prefix must be the same for both.

Delete Trunk Prefix Enabling this will delete the configured number of digits from both normal and diverted calls.

Configuring number translation

This section details the configuration of the private to public number translation used for the E1M-DPNSS.

To configure the number translation, you need to know the private number range that will be dialled out of the PABX, and a public number of the destination device that has the destination PABX connected. That number is by convention, the lowest number or base number of the ISDN link at the destination.

This section uses the network shown on page 4-53 as an example of setting up Number Translation — PABX in Node 1, dialling extensions 4200 to 4299 on a PABX in Node 2, which is on an ISDN service (01223) 352700.

This method uses the Number Checking facility. For further information see *CLI/Number checking* on page 3-49.

Note *In a 'DPNSS Transparency over QSIG' network, sites that require **Feature Transparency** without the use of Router Tables and SAM channels cannot use **CLI/Number Checking** in the E1M-DPNSS configuration.*

Number list name Enter a name for the Number Translation list and select **New List**.

Which Party To Apply To Select **Called Party Number** to set up the number list for called party translation rather than CLI/Number checking.

Min/Only - Max Enter the minimum and maximum numbers (4200 and 4299 in this example) and select **Add Range**.

What To Do With Accepted Calls

Select **Modify Called Number** and **Select Replace With**, and enter the full number of the destination including area code. (01223352700 in this example). Choose **OK**.

This sets up a number translation from 4200-4299 to 01223352700 which now needs to be applied to a particular DPNSS module.

Select the number list from the main Number Checking dialog, and choose **Apply...**

Note *The destination number must be in a lineset or VX range of the destination device. If not, the call is rejected with a cause 'Unassigned Number'.*

Possible modules/lines/sets

Select the DPNSS module, and choose **Add**. This will add all the lines from the DPNSS module (that is, a call from any LAP) to the list of lines affected by this Number Translation.

You do not have to have all LAPs entered. If only certain LAPs are used to reach certain destinations, then these can be added or deleted on a line by line basis. Similarly you can add more than one DPNSS module.

Configuring Signalling paths

The signalling router tables and associated signalling channels have to be configured to tell the device how to get DPNSS signalling from one node to another. This is used for supplementary services on Real DPNSS calls and carries the entire call information for DPNSS Virtual calls.

Signalling can be carried on an ISDN D-Channel call, or inband, using a SAM B-channel over an ISDN B-Channel call. The choice of these depends on the call routing that is to be used.

- The D-channel would normally be selected where calls are carried over a private link such as a 2Mbit/s leased line with IPMN(VX) at either end, or an n x 64K digital link with a DTM(VX) at either end.
- The D-channel should be selected where calls are carried over the Australian public network which supports D channel calls.
- A SAM B-channel should be selected where calls are carried over UK or other public network that does not support D-channel calls.

The Router tables allow for two routes to be configured, a primary route and a secondary route. If backup of the primary route is required, a secondary route must be configured even if it specifies the same SAM channel as the primary route. For example, if the first choice route is to use a SAM B-channel across a private link and the second choice is to use a SAM B-channel over the public network, the primary and secondary route would both be configured to use the same SAM B-channel.

A typical network has a leased line with overflow onto the public network. In this case, where the public network does not support D-channel calls, the primary and secondary route feature is used, with the primary route being a D-channel, and the secondary route being a B-channel.

In this instance the extra B-channel for signalling is only used when the leased line fails. In the normal overflow situation the signalling uses the D-channel of the leased line and only the traffic channel uses the public network.

Configuring the numbers

The numbers that are configured into the signalling routing tables are the full public numbers, including the area code, of the destination. At the destination, the absence of the number in the routing table indicates that the call is to terminate there.

Numbers can be configured as a single number, or a range of numbers into the SAM or CM/RM-SAM line control. Refer to *Inband signalling facility* on page 3-43 and *SAM configuration* on page 4-223 for more information.

Min/Only Primary Routes/ Secondary Routes

Enter the number in Min/Only or number range.

Select the primary and secondary routes from the lists, and choose **Add** or **Add Range**.

For example, for calls between Node 1 and Node 2 nodes set as follows:

Node 1	Node2
Router table entry 01223352700	Router table entry 01865449200

Only the Resource Manager or Control Module Router Table is used when determining DPNSS transparency, however if signalling routes use SAM channels from a separate SAM card, then the SAM Router table should also be configured to include the DPNSS destinations that use these SAM paths.

Note *Router Table entries are matched in a similar way to Linesets and the Number Translation Tables. A number matches even if it is longer than the Router Table entry as long as the digits / characters match the length of the Router Table entry.*

Configuring the signalling channels

Configuring the signalling channels involves choosing a pair of either B- or D-channels, one at each end of the link, and setting matching and opposite line number and Hotline numbers at each end.

The channels should be in Hotline mode and auto-answer (default). For SAM B-channels, enable the **Do Not Use for Incoming Manager calls** parameter in Call Establishment Modes.

Note *When using an RM we recommend that you use channels 1 to 16 to configure **Feature Transparency**. This leaves the SAM Inband Channel free to support Remote Manager Access and allows you to easily terminate incoming Manager calls.*

For example, to configure primary D-Channel and secondary B-channels between Node 1 and Node 2 enter the following:

Node 1			Node 2		
D Ch 1	Line number	449295	D Ch 1	Line number	352795
	Hotline number	01223352795		Hotline number	01865449295
B Ch 1	Line number	449296	B Ch 1	Line number	352796
	Hotline number	01223352796		Hotline number	01865449296

D Channel Acknowledgement

When using a SAM D-channel for DPNSS signalling there is an option to use either **Acknowledged** or **UnAcknowledged** signalling. This Acknowledgement is only required where signalling is carried across a public network that supports D-channel calls, such as that in Australia. If this signalling is across a private link, then Acknowledgement is unnecessary and can be turned off. This will result in a better call signalling performance.

For SAM D-channel line configuration, disable the **Send Acknowledge Message**. You should configure both ends of a link in the same way.

Intermediate nodes

If a DPNSS transparent call has to pass through a VX node without terminating there, you should also add a router table entry to that node. The entry should be the public number because when a DPNSS call is received, the device uses the absence of a router table entry for a particular number to indicate that the call will terminate on this node.

This router table entry can point to a SAM channel if calls also originate from this node and a signalling path is required. If this is not the case, select the trunk card that will be carrying the call.

For example, if a call is arriving at an IPMN in slot 3, and departing from an IPMN in slot 4, add the public number of the DPNSS call to the router table in Resource Manager or Control Module and select 4,0,0 as the primary route.

SAM Channel hold time

This allows programming of the time that a SAM channel will hold a call up for after the last user of that channel has cleared the call. This is used to prevent two successive uses of a channel from incurring two flag fall call charges. The default for this is 30 seconds, but it can be changed in multiples of 10 seconds between 10 and 2550 seconds. It should be set to a time equivalent in call cost to a flag fall.

For example, if a call costs 20p flagfall + 10p per minute you would set the holding time to two minutes.

Or if a call costs 25c flagfall + 5c per minute you would set the holding time to five minutes.

See *Inband signalling facility* on page 3-43 for further details.

Configuring the bandwidth

The final item to configure is the bandwidth that will carry the call. This configuration depends on whether 64K uncompressed, or compressed bandwidth is used. 64K uncompressed has a faster call setup time and higher load capacity, but uses more bandwidth.

Uncompressed

If the traffic is to be carried over the public network via an IPMT, no further configuration is required.

If the traffic is to be carried over a private network link, with or without public network overflow and/or backup, create a lineset with the full public number range of the destination device. Add the required lines of the IPMN or DTM and set the lineset to **Non-local** attributes.

Compressed

There are two options when configuring compressed Virtual Lines for use with DPNSS.

- The use of on-demand establishment
- Manually connect the Virtual Lines to the IPMN B-channels.

With on-demand, the backup and overflow will also be compressed, however there is a slight penalty in terms of call setup time.

Manual connections remove those particular B-channels from the pool of channels available for other non-DPNSS traffic that may be sharing the link. Manual connections also give a slightly improved call setup time, but are more complex to configure.

Refer to *Interworking* on page 3-26 for further information.

To configure compressed bandwidth:

- 1 Add sufficient interwork lines to have 1 pair per compressed traffic channel, that is one interwork unit, 30 lines per DPNSS module.
- 2 Create pairs of Subrate Virtual Lines with right hand side interwork lines as members. Choose the appropriate compression algorithm.
- 3 We recommend that you assign some Virtual Lines for traffic in each direction rather than using them bidirectionally to avoid call collisions.
- 4 The left hand sides of the interworks are then placed into a lineset. The number range of the lineset should be the translated public number. The lineset attributes should be **Non-Local**.

In the example configuration, the lineset in the Node 1 would be 01223352700-01223352700, members 0,0,1 + 0,0,2 + 0,0,3 + 0,0,4 and so on.

On-demand

The interwork lines do not require any special configuration. Set the Virtual Lines into end-to-end signalling using CM-SAM and auto-answer. The Virtual Lines should then be assigned pairs of line numbers / Hotline numbers with their peers in the other node.

If the traffic is to be carried over the public network via an IPMT, it requires no further configuration.

If the traffic is to be carried over a private network link, with or without public network overflow and/or backup, create a lineset with the full public number range of the destination device except for the translated public number. Add the required lines of the IPMN or DTM and set the lineset to **Non-local** attributes.

In the example configuration, this lineset in the Node 1 node would be 01223352701-01223352799, with IPMN channels as members.

Manually Connected

The interwork lines that are members of the Virtual Lines (RHS by convention) should be set to **SPC mode (AUS) / Member of Subrate Virtual Line or manually connected (UKS/ROW) End-to-end mode** and **Transparent mode** on.

Each one needs a unique label to identify it for signalling. This should match its peer in the other node.

Each one requires an end to end signalling number, which is the national significant public base number of the destination for signalling that is, an area code but no leading zero. In the example configuration, the remote signalling number for Node 1 is 1223352700.

The Virtual Lines are also set to **SPC mode (AUS) / Member of Subrate Virtual Line or manually connected (UKS/ROW)**.

The Virtual Lines are manually connected (bi-directional, external, non-volatile) to a B-channel on the IPMN that is across the private link, and the corresponding one connected to the same channel at the destination.

The B-channels used for this should be excluded from the linesets that are routing other traffic across the private link.

E1M-QSIG

The E1M-QSIG Module enables Private Integrated Network Exchanges (PINXs) and other equipment that support QSIG to connect to other networks.

The E1M-QSIG can use the full range of proprietary networking features including SPCs, MicropleX switching, and private and public networks to carry QSIG traffic. There is an additional number translation feature specifically for the E1M-QSIG. This allows private to public number mapping and substitution/replacement. The signalling for the feature transparency can be passed over the D-channel, or over the B-channel using the SAM feature.

The module supports Layer 3 Segmentation and Reassembly (SAR), and is compliant with ETS 300 172.

SAR is a function used when the QSIG Layer 3 messages are too large to fit into a single Layer 2 INFO frame, that is, the Layer 3 message is more than 260 bytes. The large Layer 3 message is 'Segmented' into a number of smaller (SEGMENT) messages. These are then sent in sequence and 'Reassembled' into the original message at the destination.

The E1M-QSIG supports two modes of operation:

- Transparent — where two or more QSIG PINXs communicate using QSIG
- Gateway — where a QSIG PINX communicates with other non-QSIG equipment such as a telephone.

See page 5-66 in the *Applications* section for information on a typical E1M-QSIG application.

Operation

The way in which QSIG operates varies with the application scenario. In QSIG to QSIG connections, for instance, the device provides a fully featured QSIG end-to-end connection. Where the connection involves non-QSIG devices, the module provides mapping of a subset of QSIG call facilities onto the non-QSIG device signalling.

The QSIG also has to cater for the different methods used by networks for signalling and transporting supplementary services. The module supports QSIG in the same way as the other end-to-end signalling schemes and the rules for D-channel or B-channel end-to-end signalling apply.

The E1M-QSIG collects dialled digits from the QSIG equipment. Using this information, it decides how to route the call and determines whether the destination is QSIG or not. If it is QSIG, the call is Transparent, if not, it is a Gateway call.

QSIG, like DPNSS, supports both Real Calls and Virtual Calls. QSIG, however, supports two types of Virtual call and one type of Real call.

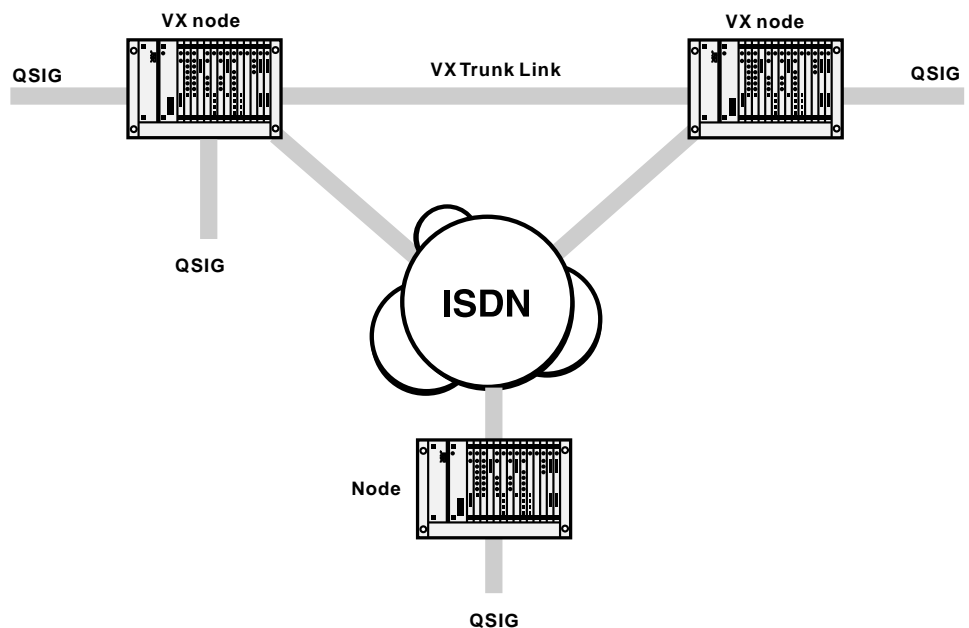
- Real Call Connection orientated call association
- Virtual Call Connection orientated non call associated
- Virtual Call Connectionless call.

Real Calls can be Transparent or Gateway, Virtual Calls must be Transparent.

Note *Local switching between two E1M-QSIG modules is not supported in this release.*

Transparent operation

If two E1M-QSIG modules are inter-operating across a private network or a public network, they are said to be operating 'Transparently'. This would be the case when the two modules are in the same device or in separate systems connected across a network. The figure below shows a typical application using the transparent mode of operation.



Note *In Transparent mode, all QSIG supplementary services work as expected.*

Local connection

The passing of QSIG Supplementary Services is achieved using a simple call switched by NeST to the destination E1M-QSIG. Supplementary service information is passed as 'User-to-User associated with the call'. There is no need for a separate signalling channel.

Network connection

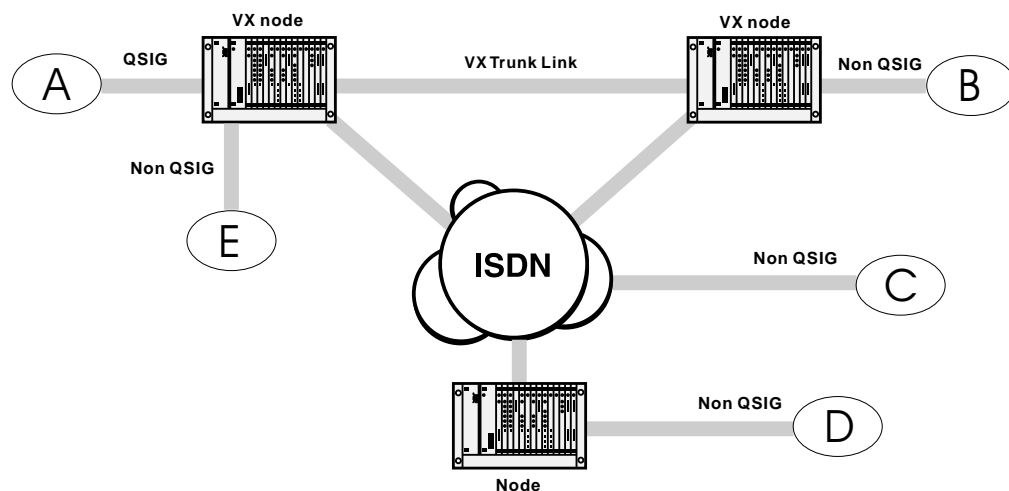
The passing of QSIG Supplementary Services is achieved using end-to-end signalling over either the VX Trunk Link or the public ISDN connections. When the ISDN does not support User-To-User messaging, the In-Band Signalling feature is used to pass the end-to-end signalling within a B-channel. Connections may transit both the VX Trunk Link and the public ISDN. This could occur in the above example when one of the ISDN connections is down.

A call is deemed to be Transparent (QSIG end-to-end) if the source is QSIG and the (translated) destination number is in the device's Router Table. Once the decision is made that a call is Transparent QSIG, the device will analyse the route to the destination to decide how to carry the call. If there is a signalling path associated with the call, that is, it is over a VX IPMN-IPMN link or a Microplex switched link, then this signalling path is used to carry the QSIG supplementary services. If there is no signalling path associated with the call, the device makes a B-channel call to carry the actual call traffic. It will use the information in the Router Table to set up a separate signalling channel to carry the QSIG supplementary services. The B-channel call and the signalling call are uniquely tagged to allow the destination to reconstruct the original QSIG call, complete with supplementary services.

Virtual Calls use the Router Table signalling path only.

Gateway operation

If E1M-QSIG modules are inter-operating with non-QSIG devices they are said to be operating in a Gateway mode. This mode of operation is supported when the connection is across private network links, public switched networks or switched locally. The figure below shows a typical example of the Gateway mode of operation.



In this example, the VX supporting [A] is operating in Gateway mode. The non-QSIG devices can be connected to any appropriate interface [B] [D] [E] or other access mechanism [C].

The level of functionality in this mode depends on the commonality between QSIG and the call signalling protocol of the non-QSIG devices. An additional requirement is that the QSIG device has a transmission format that is common to the non-QSIG device (voice, data and so on).

Interfaces that support this mode of operation include:

- ISDN Basic Rate and Primary Rate interfaces;
- G.703/704 CAS R2D;
- 4 Wire E & M SSDC5;
- 2 Wire Ring-In/Loop-out and Ring-out/Loop-in;
- Data interfaces (V.24, V.35 and X.21);
- DPNSS.

Protocol Interworking

When the destination is non-QSIG, a Gateway call is made. This translates the QSIG call signalling into ISDN call signalling and maps the basic supplementary services into ISDN ones. Supplementary services can then be mapped to other non-compatible protocols.

The mappings are shown in the following table:

QSIG	Module/Q.931	DASS2, DPNSS, ETS-102, ETS-403
CLI - Calling Line Identity	Called Party Number in SETUP	OLI - Originating Line Identity (DASS2) CLI - Calling Line Identity (ETS-102) OLI - Originating Line Identity (DPNSS)
Called Party Subaddress	Called Party Subaddress in SETUP	NAE - Network Address Extension (DASS2) NAE-DC - Network Address Extension -Destination Complete (DPNSS) Called Party Subaddress (ETS-102)
Bearer/LLC/HLC	Bearer/LLC/HLC	SIC - Service Identifier Code (DASS2/DPNSS) Bearer/LLC/HLC (ETS-102)

The QSIG-ETSI supplementary services are covered in more detail in *Appendix N*

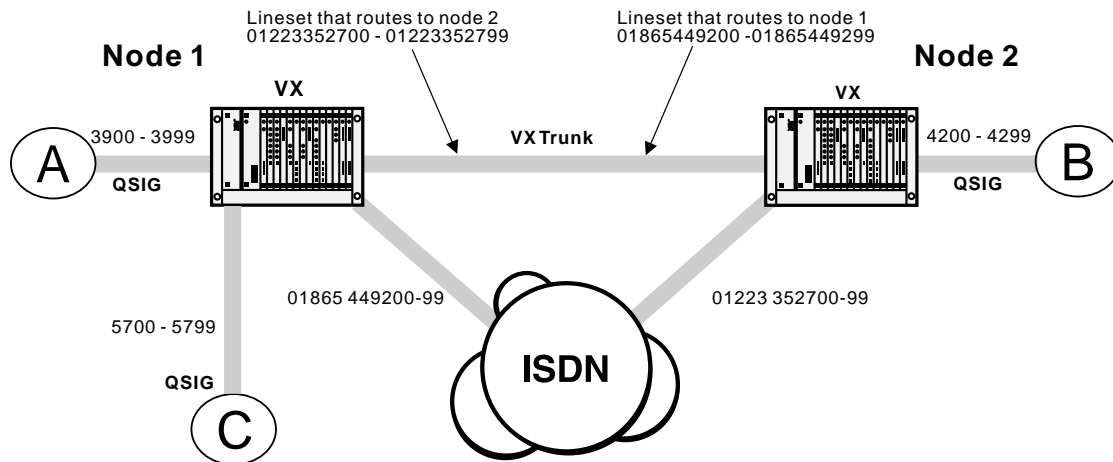
Intermediate Nodes

If a node receives a QSIG call that is already tagged as a Transparent Call, the node checks to see whether the destination is in its Router Table. If the device has the destination in its router table, it knows that it is just an intermediate node for the call and will pass the call on unchanged. If the destination is **not** in the Router Table, the device assumes that the call terminates in this node and it will attempt to reunite the signalling and B-channel calls.

Numbering

Typically a QSIG link between two PINXs will use the PINX's private numbering scheme. This will usually be 3 to 6 digit numbers that have no relevance to public network numbering.

A typical network will have a mix of public and private links, which must carry the PINX private numbering transparently. The following diagram shows this:



Local connections

Calls that are to be switched locally require no signalling or traffic channel to another device via a private link or public network. Thus, Router Table entries and number manipulation are not required. Calls are instead switched by the VX Network Simulation Task (NeST) on Lineset match.

In the above example, PINX A in Node 1 dials 57xx which is an extension in PINX C in Node 1. The dialled number 57xx matches a lineset in Node 1 VX and the call is switched locally. This is the case for Real and Virtual calls.

Additional digits dialled after the 57xx are passed in INFORMATION messages as part of a normal call setup to PINX C.

Connections across networks

The rule when configuring networks with QSIG is to configure the entire network using public numbering, and use the number translation tables to translate the PINX private numbering into a public number that can be used to reach the destination device.

In the above example, PINX A in Node 1 dials 42xx which is an extension on PINX B in Node 2. 42xx matches a number table in the Node 1 VX and is replaced by 01223352700. All routing decisions are then based on this public number.

First preference is a lineset in device Node 1 which routes the call over the VX trunk. If the VX trunk is full or broken, the call will route over the public ISDN. This gives full backup and overflow functionality for the VX trunks without any further number manipulation.

When the call arrives at the Node 2 VX, it is recognised as a QSIG call that has undergone number replacement, and the original 42xx is replaced before final routing to PINX B.

Additional digits dialled after the 42xx will be passed across to node 2 over the signalling channel and sent on to PINX B.

QSIG Configuration

Configuring a system that uses the E1M-QSIG is a four stage process.

- Stage 1 — Configure the E1M-QSIG
- Stage 2 — Configure the number translation
- Stage 3 — Configure the signalling routing tables
- Stage 4 — Configure the bandwidth for the calls.

Note *For calls across a network you need to carry out all four steps. For locally switched calls only the first step is required.*

Configuring the E1M-QSIG

This section describes how to configure the E1M-QSIG.

Number range The number range selects which range of numbers received by the device will be delivered to the E1M-QSIG.

Where number translation is used, the **Private Number Range** of the PINX will go to the QSIG module, and **not** the range of public numbers that are delivered to the device.

Enter the number range. The number range need not be the full length of the dialled number. For example, 31-39 will match any number beginning with 31 through to 39. This creates a lineset which includes all channels of the E1M-QSIG. Frequently however, different number ranges will be applied to different channels. In these cases use the NMS Lineset editing facility to configure them.

Private/Public Number Clashes

You should normally configure Linesets with the **Local** attribute set. The exception to this is when there is a conflict between the private number range and public numbers that are required to be routed within the device. For example, the PINX number range may be 42xx, but the device is required to route 420-0100 to a different interface. In this case, you should set the QSIG lineset to 4200-4299 and enable the **Exact Number Length** attribute. This will prevent 4200100 matching the set which it would otherwise do.

Channel Selection Order

If you configure the PINX as 'A', it selects channels for outgoing calls in ascending order. When configured as 'B', channels are selected in descending order. This helps to minimise channel collisions between incoming and outgoing calls.

If the PINX connected to an E1M-QSIG is configured as 'B', the E1M-QSIG should be configured as 'A' and the Lineset should list the QSIG channels in ascending order. If they are configured in the opposite direction, the Lineset should list the channels in descending order.

Entering the card number range creates a 'Local' lineset with all 30 QSIG channels, in ascending order. This is the default setting and corresponds to the default 'A' setting of the E1M-QSIG.

See *Linesets* on page 3-17 for information on reversing the order of the E1M-QSIG lineset.

Configure other parameters

Edit line The default configuration of the E1M-QSIG is for all 30 Channels enabled. You can not modify this number.

Card options

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

CRC enable Enable this if your ISDN terminal equipment requires Cyclic Redundancy Check (CRC) to monitor lines. You would not normally need to change the default (disabled).

This enables CRC-4 error checking at Layer 1. It is always transmitted. Enabling this function causes CRC-4 data to be the Layer 1 line monitoring criteria. If it is disabled, framing errors are interpreted as Layer 1 line errors.

Do not use as reference clock Select this option if you do not require the E1M-QSIG to supply the clock reference to synchronise the device to the line.

The clock reference to synchronise the device is supplied by an ISDN TE module. If no TE module is present in the device and there is no other source of a clock reference, you can configure the E1M-QSIG to supply it with a reference clock obtained from the terminal equipment to which it is connected.

The settings for the above parameters affect the operation of the entire module. Having configured the module, you can select individual lines and configure them as required.

Do not use as bus master Enable this to prevent the E1M-QSIG being selected to provide the backplane bus clock.

Applications...

Connected to...

This option is used to specify how the E1M-QSIG is being used.

PBX Select PBX if the E1M-QSIG is connected to a PINX or a QSIG network. PBX must be selected for Feature Transparency to work.

Network supporting UUS Select this if you connect to a QSIG network with non-PINX equipment, and the network supports and uses User-to-user signalling.

Network Not supporting UUS Select this if you connect to a QSIG network with non-PINX equipment, and the network does **not** support User-to-user signalling.

QSIG configuration

A/B When two pieces of equipment are connected using QSIG, one side is designated 'A' and the other side 'B'. This is used to determine the actions of each end of the inter PINX link under call collision conditions.

If the PINX is configured as 'A' the E1M-QSIG should be configured as 'B' and vice versa.

Note *If this parameter is changed, the Channel selection order should also be changed. See Channel selection order on page 4-69 for details.*

Set Channel Exclusive/Preferred These parameters determine the value of the Channel Selection bit sent in the Channel Identification IE. The IE is sent by the E1M-QSIG to the connected PINX. If you choose 'Exclusive', and a channel collision occurs, the call is either successful or is cleared. If you choose 'Preferred', channel negotiation is attempted. The actions performed to resolve the collision are dependent on whether the E1M-QSIG is configured as 'A' or 'B'.

Network options

These options are only available if you have selected a network option in the **Connected to...** section.

Local Switching Select this option if the application requires QSIG Transparency between E1M-QSIG modules in the same device.

Note *Local switching between two E1M-QSIG modules is not supported in this release.*

Encapsulate DPNSS Select this option if the application is required to provide DPNSS Transparency over the QSIG network without the need for a SAM channel. Sites that require this feature cannot use Number Translation.

Note *If this option is selected then SAM router table entry configuration should be omitted from the DPNSS configuration (see DPNSS configuration on page 4-54 for more details).*

Indial digits

Limit DDI Limit DDI restricts the number of digits that will be dialled into the QSIG PINX from the E1M-QSIG. If this is **not** enabled, the full number received by the E1M-QSIG is forwarded. If the parameter is enabled, and a number 'n' entered, only the **last** 'n' digits of the number are forwarded. This setting applies to the entire module, and is not line by line configurable. This is only likely to be used if number translation is not used.

Dialled digits

Minimum number length When enabled, this determines how many digits the E1M-QSIG collects before forwarding the call to the device for routing. It is important to set this to the same as the length of the lineset range configured for the destination E1M-QSIG. This should also be the same length as the numbers entered into the **Number Translation lists**, or the router table if **Number Translation** is not used.

For example, if the PINX is dialling numbers in the range 4200 to 4299, then the **Number Translation list** would be configured to 4200-4299, and the **Minimum Number Length** of dialled digits should be set to 4.

This setting is only useful where a PINX sends individual digits as they are dialled. In a lot of cases, the PINX collects digits itself and does not forward the call to the QSIG link until it collects 4 digits. If this is the case, then this setting is ignored.

Options

Idle fill Layer 2 In normal operation, QSIG Layer 2 has flag characters between Layer 2 frames. This is known to cause problems with some equipment which can indicate Layer 2 CRC errors when flag fill is used. If an abnormal number of Layer 2 CRC errors are reported, enable this parameter.

Extend T303 to 9.5s	Enable this to extend the timer to 9.5 secs. This is useful when the E1M-QSIG is connected to equipment or a network that takes longer than normal to respond to call setup procedures. This avoids unwanted timeouts and premature clearing of calls.
Add progress to alerting	When making calls to certain Private Integrated Service Network (PISN) equipment, the tone supplied by the VX may not be required. Select this option to enable the ring tone and RVAs provided by the connected PINX.
Number type unknown	This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Unknown'. This may be a requirement of some PISN equipment.
Number type subscriber	This allows the Type Of Number field in the Called Party Number Information Element to be forced to 'Subscriber'. This may be a requirement of some PISN equipment.
PBX options	
	This option is only available when you have selected PBX in the Connected to... section.
Feature transparency	You should enable this option when the purpose of your application is to provide QSIG Transparency over an intermediate network between PINXs.

Note *For uncompressed QSIG calls supporting **Feature transparency** over ISDN or Leased Lines, both parts of the split call (i.e. the B-channel bearer and the SAM signalling channel) must follow the same path. This also implies that the SAM B-channel should be used to carry the signalling rather than the SAM D-channel.*

*If the same path is not used, only the **Cause Information Element (IE)**, and not the **Facility IE** or other IEs, is returned in the clearing message received by the caller. This means Supplementary Services such as 'Divert on Busy' that require other information in the clearing message, will not function properly.*

Call Establishment Modes

Add Trunk Prefix	Gateway calls that originate from a non-QSIG user are prefixed when this option is enabled and a Prefix is entered. This operates for normal calls. To add a Trunk Prefix to calls that have undergone Call Diversion, enable Diverted . You can enable both normal and diverted, none or one of the options. When using both normal and diverted, the prefix must be the same for both.
Delete Trunk Prefix	Enabling this will delete the configured number of digits from both normal and diverted calls.

Note *The Trunk Prefix options are not supported for SAMD-channel calls. SAMB-channel calls are supported.*

Configuring number translation

This section details the configuration of the private to public number translation used for the QSIG Transparent calls that are connected across a network. If you are operating over a local link, you do not need to perform Number Translation.

To configure the number translation, you need to know the private number range that will be dialled out of the PINX, and a public number of the destination device that has the destination PINX connected. That number, by convention, is the lowest number or base number of the ISDN link at the destination.

This section uses the network shown on page 4-67 as an example of setting up Number Translation — PINX in Node 1, dialling extensions 4200 to 4299 on a PINX in Node 2, which is on an ISDN service (01223) 352700.

Note *In a 'DPNSS Transparency over QSIG' network, sites that require **Feature Transparency** without the use of Router Tables and SAM channels cannot use **CLI/Number Checking** from the DPNSS configuration.*

This method uses the Number Checking facility. For further information see *CLI/Number checking* on page 3-49.

Number list name	Enter a name for the Number Translation list and select New List .
Apply To	Select QSIG Called Party Number to set up the number list for called party translation rather than CLI/Number checking.
Min/Only - Max	Enter the minimum and maximum numbers (4200 and 4299 in this example) and select Add Range .
What To Do With Accepted Calls	Select Modify Called Number and choose Replace With , and enter the full number of the destination including area code. (01223352700 in this example). Choose OK . This sets up a number translation from 4200-4299 to 01223352700 which now needs to be applied to a particular QSIG module.

Note *When using number translation the 'Replaced With' number has to be in a lineset or VX range of the destination device. If it is not, the call is rejected with the cause 'Unassigned Number'.*

Select the number list from the main Number Checking dialog, and choose **Apply...**

Possible modules/lines/sets	Select the QSIG module, and choose Add . This will add all the lines from the QSIG module to the list of lines affected by this Number Translation. You do not need to have all channels entered. If only certain channels are used to reach certain destinations, then these can be added or deleted on a line by line basis. Similarly you can add more than one QSIG module.
------------------------------------	---

Configuring Signalling paths

The signalling router tables and associated signalling channels have to be configured to tell the device how to get QSIG signalling from one node to another. This is used for supplementary services on Real QSIG calls and carries the entire call information for QSIG Virtual calls when connected across a network. For locally switched calls, the router is not required, and therefore no signalling path needs to be configured.

Signalling can be carried on an ISDN D-Channel call, or inband, using a SAM B-channel over an ISDN B-Channel call. The choice of these depends on the call routing that is to be used.

- The D-channel would normally be selected where calls are carried over a private link such as a 2Mbit/s leased line with IPMN(VX) at either end, or a n*64k digital link with a DTM(VX) at either end.
- The D-channel should be selected where calls are carried over the Australian public network which supports D-channel calls.
- A SAM B-channel should be selected where calls are carried over public networks that do not support D-channel calls.

The Router tables allow for two routes to be configured, a **primary route** and a **secondary route**. If backup of the **primary route** is required, a **secondary route** must be configured even if it specifies the same SAM channel as the primary route. For example, if the first choice route is to use a SAM B-channel across a private link and the second choice is to use a SAM B-channel over the public network, the primary and secondary route would both be configured to use the same SAM B-channel.

A typical network has a leased line with overflow onto the public network. In this case, where the public network does not support D-channel calls, the primary and secondary route feature is used, with the primary route being a D-channel, and the secondary route being a B-channel.

In this instance the extra B-channel for signalling is only used when the leased line fails. In the normal overflow situation the signalling uses the D-channel of the leased line and only the traffic channel uses the public network.

Configuring the numbers

The numbers that are configured into the signalling routing tables are the full public numbers, including the area code, of the destination. At the destination, the absence of the number in the routing table indicates that the call is to terminate there.

Numbers can be configured as a single number, or a range of numbers into the SAM or RM/CM-SAM line control. Refer to *Inband signalling facility* on page 3-43 and *SAM configuration* on page 4-223 for more information.

Min/Only Enter the number in Min/Only or number range.

**Primary Routes/
Secondary Routes**

Select the primary and secondary routes from the lists, and choose **Add** or **Add Range**.

For example, for calls between Node 1 and Node 2 nodes set as follows:

Node 1	Node2
Router table entry 01223352700	Router table entry 01865449200

Only the Resource Manager or Control Module Router Table is used when determining QSIG transparency, however if signalling routes use SAM channels from a separate SAM card, then the SAM Router table should also be configured to include the QSIG destinations that use these SAM paths.

Note Router Table entries are matched in a similar way to Linesets and the Number Translation Tables. A number matches even if it is longer than the Router Table entry as long as the digits / characters match the length of the Router Table entry.

Configuring the signalling channels

Configuring the signalling channels involves choosing a pair of either B- or D-channels, one at each end of the link, and setting matching and opposite line number and Hotline numbers at each end.

The channels should be in Hotline mode and auto-answer (default). For SAM B-channels, enable the **Do Not Use for Incoming Manager calls** parameter in Call Establishment Modes.

Note We recommend that you use channels 1 to 16 to configure **Feature Transparency**. This leaves the SAM Inband Channel free to support Remote Manager Access and allows you to easily terminate incoming Manager calls.

For example, to configure primary D-Channel and secondary B-channels between Node 1 and Node 2 enter the following:

Node 1			Node 2		
D Ch 1	Line number	449295	D Ch 1	Line number	352795
	Hotline number	01223352795		Hotline number	01865449295
B Ch 1	Line number	449296	B Ch 1	Line number	352796
	Hotline number	01223352796		Hotline number	01865449296

D Channel Acknowledgement

When using a SAM D-channel for QSIG signalling there is an option to use either **Acknowledged** or **UnAcknowledged** signalling. This Acknowledgement is only required where signalling is carried across a public network that supports D-channel calls, such as that in Australia. If this signalling is across a private link, then Acknowledgement is unnecessary and can be turned off. This will result in a better call signalling performance.

For SAM D-channel line configuration, disable the **Send Acknowledge Message**. You should configure both ends of a link in the same way.

Intermediate nodes

If a QSIG transparent call has to pass through a VX node without terminating there, you should also add a router table entry to that node. The entry should be the public number because when a QSIG call is received, the device uses the absence of a router table entry for a particular number to indicate that the call will terminate on this node.

This router table entry can point to a SAM channel if calls also originate from this node and a signalling path is required. If this is not the case, select the trunk card that will be carrying the call.

For example, if a call is arriving at an IPMN in slot 3, and departing from an IPMN in slot 4, add the public number of the QSIG call to the router table in the Resource Manager or Control Module and select 4,0,0 as the primary route.

SAM Channel hold time

This allows programming of the time for which a SAM channel will hold up a call after the last user of that channel has cleared the call. This is used to prevent two successive uses of a channel from incurring two flag fall call charges. The default for this is 30 seconds, but it can be changed in multiples of 10 seconds between 10 and 2550 seconds. It should be set to a time equivalent in call cost to a flag fall.

For example, if a call costs 20p flagfall + 10p per minute you would set the holding time to two minutes.

Or, if a call costs 25c flagfall + 5c per minute you would set the holding time to five minutes.

See *Inband signalling facility* on page 3-43 for further details.

Configuring the bandwidth

The final item to configure is the bandwidth that will carry the call. This configuration depends on whether 64K uncompressed, or compressed bandwidth is used. 64k uncompressed has a faster call setup time and higher load capacity, but uses more bandwidth.

Uncompressed

For locally switched calls, no further configuration is necessary. You should however ensure that the Lineset for the destination E1M-QSIG is properly configured.

If the traffic is to be carried over the public network via an IPMT, no further configuration is required.

If the traffic is to be carried over a private network link, with or without public network overflow and/or backup, create a lineset with the full public number range of the destination device. Add the required lines of the IPMN or DTM and set the lineset to **Non-local** attributes.

Compressed

This is only used when connecting across a network.

There are two options when configuring compressed Virtual Lines for use with QSIG.

- The use of on-demand establishment
- Manually connect the Virtual Lines to the IPMN B-channels.

With on-demand, the backup and overflow will also be compressed, however there is a slight penalty in terms of call setup time.

Manual connections remove those particular B-channels from the pool of channels available for other non-QSIG traffic that may be sharing the link. Manual connections also give a slightly improved call setup time, but are more complex to configure.

Refer to *Interworking* on page 3-26 for further information.

To configure compressed bandwidth:

- 1 Add sufficient interwork lines to have 1 pair per compressed traffic channel, that is one interwork unit, 30 lines per QSIG module.
- 2 Create pairs of Subrate Virtual Lines with right hand side interwork lines as members. Choose the appropriate compression algorithm.

We recommend that you assign some Virtual Lines for traffic in each direction rather than using them bidirectionally to avoid call collisions.

- 3** Place the left hand sides of the interworks into a lineset. The number range of the lineset should be the translated public number. The lineset attributes should be **Non-Local**.

In the example configuration, the lineset in the Node 1 would be 01223352700-01223352700, members 0,0,1 + 0,0,2 + 0,0,3 + 0,0,4 and so on.

On-demand

The interwork lines do not require any special configuration. Set the Virtual Lines into end-to-end signalling using RM/CM-SAM and auto-answer. The Virtual Lines should then be assigned pairs of line numbers / Hotline numbers with their peers in the other node.

If the traffic is to be carried over the public network via an IPMT, it requires no further configuration.

If the traffic is to be carried over a private network link, with or without public network overflow and/or backup, create a lineset with the full public number range of the destination device except for the translated public number. Add the required lines of the IPMN or DTM and set the lineset to **Non-local** attributes.

In the example configuration, this lineset in the Node 1 node would be 01223352701-01223352799, with IPMN channels as members.

Manually Connected

The interwork lines that are members of the Virtual Lines (RHS by convention) should be set to **SPC mode (AUS) / Member of Subrate Virtual Line or manually connected (UKS/ROW) End-to-end mode and Transparent mode on**.

Each one needs a unique label to identify it for signalling. This should match its peer in the other node.

Each one requires an end to end signalling number, which is the national significant public base number of the destination for signalling that is, an area code but no leading zero. In the example configuration, the remote signalling number for Node 1 is 1223352700.

The Virtual Lines are also set to **SPC mode or Member of Subrate Virtual Line or manually connected** as applicable.

The Virtual Lines are manually connected (bi-directional, external, non-volatile) to a B-channel on the IPMN that is across the private link, and the corresponding one connected to the same channel at the destination.

The B-channels used for this should be excluded from the linesets that are routing other traffic across the private link.

Digital Trunk Module (DTM)

The Digital Trunk Module (DTM) is a trunk module capable of interfacing with Digital Data Network trunks. The type of interfaces supported are X.21 and V.35 for data rates of 48 and 56 kbit/s and Nx64 kbit/s up to 2.048 Mbit/s respectively.

This module provides the Virtual eXchange with a trunking gateway to private networks or private microwave. ISDN signalling protocols are adopted to provide call routing. This feature enables the device to be a hybrid networking multiplexer, incorporating private networks with ISDN.

Operation

The DTM supports up to four independent trunk ports. Submodules allow each port to be either X.21 or V.35. Each port operates in either DTE or DCE mode with various clock source configurations.

The trunk bandwidth is partitioned to provide nominal proprietary framing, a D-channel and a number of B-channels. The nominal framing bandwidth is fixed at 4 kbit/s. The D-channel bandwidth is configurable from 4 to 60 kbit/s in 8 kbit/s steps. The rest of the bandwidth is filled with as many B-channels as possible. Any remaining bandwidth is treated as a subB-channel that can be used for Subrate user information that only partially uses a full B-channel.

The DTM is a private network interface that provides ISDN-like services over non-ISDN links. The services provided resemble those of the IPM over private networks. It can be used to interconnect devices and their various line interfaces with the same call capabilities as the IPM. The subB-channel is provided to fully utilise trunk bandwidth and has a limited usage to that of normal B-channels.

OmniVision management calls cannot be made over a subB-channel without a Subrate Multiplexing or Voice Compression Module because an OmniVision call requires the full 64 kbit/s bandwidth of a B-channel (although 8 kbit/s of user data is actually passed). The subB-channel can be used to pass any of the following user information:

- compressed voice (using a Voice Compression Module)
- Subrate multiplexed low speed data (using a Subrate Multiplexing Module)
- a combination of compressed voice and low speed data (using VCM/SRMM) over a TimeFrame Virtual Line
- low speed data with bearer intermediate rate (or non-CCITT (ITU-T) user rate) less than or equal to the subB-channel speed.

Each port of the DTM is hardware link configured to operate in either NT or TE (call signalling protocol) mode. NT mode is required in typical Virtual eXchange applications where linesets or line numbers are associated with the trunk for call routing purposes. TE mode is required in applications where the DTM port is connected to a DTM port of another device operating in NT mode. This means DTM NT to NT and NT to TE connections are supported but TE to TE connections are not possible. Refer to *Virtual eXchange* on page 3-14 for more information.

The DTM can be a reference clock master, using an incoming trunk clock (from any trunk port) as a reference clock for the device backplane. The DTM can also be a bus clock master for backplane 4 Mbit/s, 2 Mbit/s and 8 kbit/s clocks locked to the backplane reference clock. Data clocked in and out of a DTM trunk is slip buffered to allow for differences in ISDN and private network clocks.

Trunk port interface options

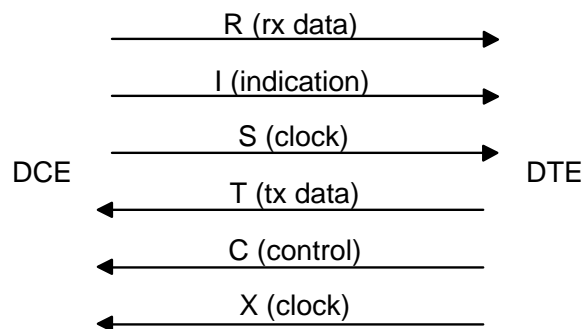
Plug-in submodules are used to provide the X.21 and V.35 interfaces. These are the same as those used by the QDLM (Quad Digital Line Module). Up to four submodules can be plugged into each DTM, supporting any combination of independent trunk ports.

Both interface types are configurable for either DTE or DCE mode and support a variety of clock source configurations. Each trunk port has independent selectable trunk speeds. The trunk speeds supported are 48, 56 and $N \times 64$ kbit/s for $N=1$ to 32.

In either DTE or DCE mode, the DTM will compare the selected trunk speed with the incoming clock speed of the trunk interface and report any discrepancies.

X.21 interface

The X.21 interface provides the R, I, T, C, S, X and G signals. The X signal replaces the B signal (byte timing). The X signal provides a clock signal that is co-directional with the T data. The X.21 interface leads are shown below:



Each port has a separate 15-pin female D-type connector according to ISO 4903.

V.35 interface

The V.35 interface provides TxD, RxD, RTS, DTR, CTS, DSR, CD, TxC, RxC, ETxC, LLB, RLB, TI and signal ground. It supports both DCE and DTE sourced transmit clocks. Each port has a separate 26-pin female high density D-type connector. Adapter cables are required for conversion to standard 34-pin rectangular connectors.

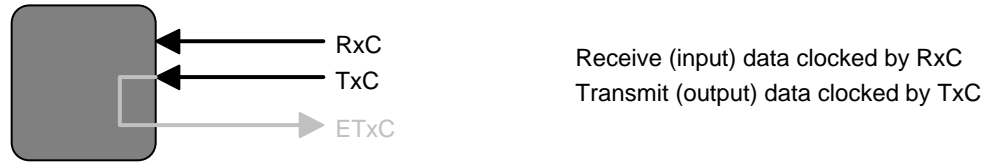
Clock source configuration

The different possible ways of configuring the clock source for transmit and receive data on the interface leads are shown in the following diagrams. These options allow for a wide range of application scenarios:

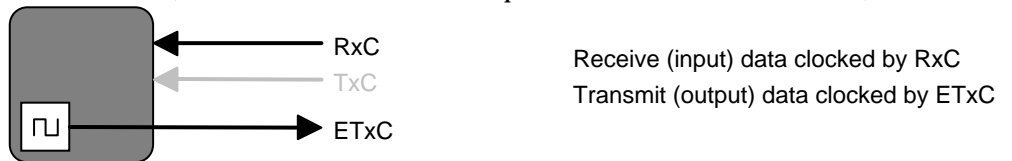
V.35 DTE port clocking options

The V.35 DTE port always uses RxC to clock RxD. The clock for TxD can be any one of the three options shown below:

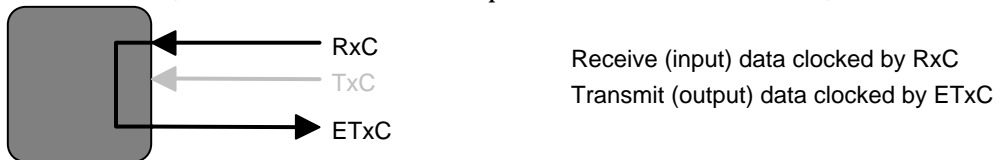
RxC+TxC (DCE sourced Rx and Tx clocks)



RxC+ETxC (DCE sourced Rx clock, independent DTE sourced Tx clock)



ETxC=RxC (DCE sourced Rx clock, looped to DTE sourced Tx clock)



V.35 DCE port clocking options

There are three different options for a V.35 DCE port:

RxC+TxC (DCE sourced Rx and Tx clocks)



RxC+ETxC (DCE sourced Rx clock, DTE sourced Tx clock)



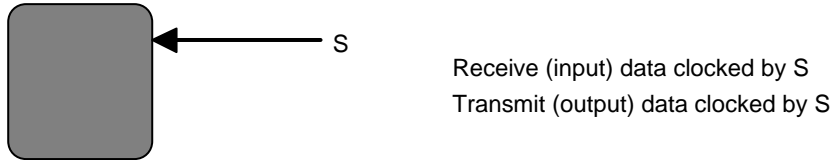
RxC=ETxC (DTE supplied Tx clock, looped to Rx clock)



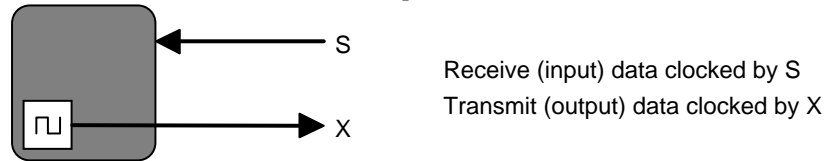
X.21 DTE port clocking options

There are three different X.21 DTE clocking options as shown below:

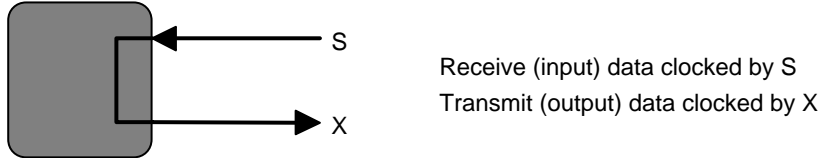
S only (DCE sourced clock for Rx and Tx)



S+X (DCE sourced Rx clock, independent DTE sourced Tx clock)



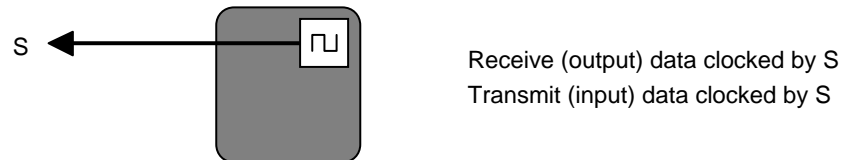
X=S (DCE sourced Rx clock, looped to DTE sourced Tx clock)



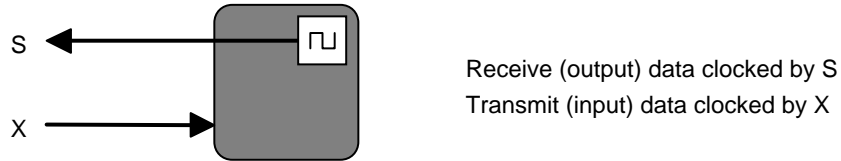
X.21 DCE port clocking options

There are three different X.21 DCE port clocking options as shown below:

S Only (DCE sourced clock for Rx and Tx)



S+X (DCE sourced Rx clock, DTE sourced Tx clock)



S=X (DTE sourced Tx clock, looped to Rx clock)



Trunk quality reporting

Private carriers claim performance statistics for their private trunk links. The DTM performs running trunk quality measurements on each trunk and provides two types of reports:

Trunk Quality Log

This logs the following trunk quality statistics:

- Errored Seconds
- approximate BER (based on CRC errors)
- Number of Synchronisation Losses
- Number of Received Data Slips
- Number of Transmitted Data Slips.

Trunk Quality Alarm

This logs alarms when trunk quality statistics reach or exceed specified thresholds for the following parameters:

- Errored Seconds
- CRC Errors (400 CRC checks/ second)
- Number Synchronisation Losses
- Number of Received Data Slips
- Number of Transmitted Data Slips.

Both report logging types are performed over specified intervals while selected. Normal alarms are also generated when Trunk Quality Alarm is selected and a transition occurs for any of the above specified parameters (see below).

Alarms

The following alarm messages are generated by the DTM and will appear in the Alarm Log when the corresponding events occur:

- Card OK
- Card Failure
- Layer 2 OK
- Layer 2 Failure
- Network Clocks Gone
- Network Clocks Present
- Bus Clocks Gone
- Bus Clocks Present
- Trunk Clocks Incorrect
- Errored Second Threshold Reached
- Errored Second Threshold Not Reached
- CRC Error Threshold Reached
- CRC Error Threshold Not Reached
- Synchronisation Threshold Reached
- Synchronisation Threshold Not Reached
- Receive Data Slips Threshold Reached
- Receive Data Slips Threshold Not Reached
- Transmit Data Slips Threshold Reached
- Transmit Data Slips Threshold Not Reached

See the *OmniVision User Manual* for further information.

subB-channel usage

When the subB-channel is used in a TimeFrame Virtual Line, a voice compression module (in a COMBO, MFCM or LDCM) or a Subrate Multiplexing Module (SRMM) is required. See the relevant module section for information on the modules. The bits, as displayed by the Network Management System, that will be available for use are:

subB	From Bit	To Bit
56 kbit/s	7	1
48 kbit/s	7	2
40 kbit/s	7	3
32 kbit/s	7	4
24 kbit/s	7	5
16 kbit/s	7	6
8 kbit/s	7	7

The subB-channel can be also used on-demand with DLM data bandwidth less than or equal to the configured subB-channel speed. The DLM used should have the Bearer Capability selected to reflect the DLM speed (refer to any DLM configuration section in this manual). The DLM (if able) can use the following CCITT (ITU-T) speeds: 600, 1.2 k, 2.4 k, 4.8 k, 9.6 k, 19.2 k (bit/s) and all non-CCITT (ITU-T) speeds less than or equal to 56 kbit/s.

Configuration

Overview

One to four plug-in submodules can be plugged in each DTM, supporting any combination of independent trunk ports. You can configure the DTM using OmniVision. The exception is NT/TE selection, which is made by hardware links on a per port basis.

A DTM trunk port can automatically synchronise to a remote port if the remote port has already been correctly configured.

For further information on the configuration procedure, refer to *Virtual eXchange* on page 3-14. The perspective used here is that line interface cards (for example, QDLM, ALEM and so on) are viewed as NTs. Calls routed between line interface cards and a DTM NT port can only be achieved through the VX.

Trunk numbering

NT trunk numbering

Port number range The NT's **Minimum Number** and **Maximum Number** define the valid number range for this NT trunk. Any calls that arrive at this device (either by TE or another NT when running the VX) that match this number range will be routed to this trunk.

The Minimum Number and Maximum Number automatically create a lineset that has all configured B-, subB- and D-channels as its members. This lineset can be viewed through Edit Lineset.

More number ranges can be attributed to each trunk using the Edit Lineset command. If Edit Lineset is used and the subB-channel is to be used in an on-demand call situation, the Attributes for Destination Type must be set to Linear Hunting (custom selection) and the Member Lines must start with the subB-channel. This ensures that calls are size checked for subB-channel connection first before whole B-channels. To reduce call collisions in an NT-NT connection, it is recommended to make one NT reverse the Member Lines ordering.

TE trunk numbering

Numbering TE line numbering is made up of a Base Number, Group Number and area code.

The Base Number has two implications:

- Incoming Called Party Number - Base Number = line interface called
- Base Number + line interface number of caller = Calling Party Number.

The Base Number is useful when there is more than one network card access because it allows each line to have an alias for each network access.

The Group Number is used to form the Calling Party Number for outgoing calls when the line making the call cannot supply a Calling Party Number.

For Australia, the 'area code' (without any subscriber trunk dialling code digits) refers to the area code without the initial zero. This area code is used in conjunction with a line number when forming the calling party number for outgoing calls.

Trunk Port Setup

There are four configurable parameters: Interface Mode, Clock Sources, Trunk Speed and D/subB Speed.

Interface Mode Interface Mode selects the trunk port to be either a DTE (default) or DCE.

Notes

- *The V.35 requires different 26-pin female high density D-type to standard 34-pin rectangular connector adaptor cables for DTE and DCE mode (see **Installation Manual**).*
 - *This selection is made independent of NT/TE trunk type.*
-

Clock Sources The Clock Sources selections are a function of Interface Mode and interface type, that is, X.21 or V.35. The relationship is shown as follows:

- **DCE X.21** - S only, S+X or S=X
- **DTE X.21** - S only (default), S+X or X=S
- **DCE V.35** - RxC+TxC, RxC+ETxC or RxC=ETxC
- **DTE V.35** - RxC+TxC (default), RxC+ETxC or ETxC=RxC.

To maximise cable length between DTE and DCE, make a Clock Sources selection that allows clocks to run in the same direction as the data for each direction. For example, in the case of an X.21 DTE port the selection of S+X or X=S would enable clocks to run in the same direction as the data for both R and T. The difference is that S+X uses its own clock source (synchronised to its own reference clock) and X=S loops the externally received S clock back (preserving the frequency) as the X clock.

Trunk Speed The available Trunk Speeds are given in two forms: bits/s and B-channels. The selectable speeds are: 48k, 56k, and 64k up to 2.048Mbit/s in 64kbit/s steps. No full B-channels are available for a Trunk Speed less than 128kbit/s. For a Trunk Speed of 128 kbit/s one B-channels is available, with one B-channel being added for every additional 64 kbit/s. Note that it is also possible to turn DTM trunk ports off.

The default startup condition of unconfigured DTM ports, is automatic speed detection. In this mode the trunk speed is sourced from the interface clock, providing there is an idle configured DTM port at the trunk's remote end. Once the speed is determined, the port is automatically configured to the detected speed. This allows remote devices, connected through the DTM, a remote OmniVision session for further configuration.

Channel speeds There is a trade-off between D-channel and subB-channel speeds. The available D/subB Speed is therefore given in two forms: subB bit/s and D bit/s. The available selections of speed are also a function of the Trunk Speed. D/subB speeds are in steps of 8 kbit/s, with the available speeds as follows:

48 kbit/s Trunk		56 kbit/s Trunk		≥ 64 kbit/s Trunk	
D-Channel	subB-Channel	D-Channel	subB-Channel	D-Channel	subB-Channel
4	40	4	48	4	56
12	32	12	40	12	48
20	24	20	32	20	40
28	16	28	24	28	32
36	8	36	16	36	24
44	0	44	8	44	16
		52	0	52	8
				60	0

The faster the subB-channel, the slower the D-channel signalling will be. This will become more evident as the amount of simultaneous calls increases or during remote D-channel management sessions, where a lot of data is passed across the D-channel.

Options

NT Options

The NT-specific options consist of three parameters: Outgoing TE, Edit lines and Channel Negotiation.

Outgoing TE You can select the TE to be used for calls going out from this NT port. The outgoing TE can be: Any (default), Preferred or Exclusive.

When **Preferred** is selected, calls will be routed from this DTM port to the nominated TE if possible, otherwise another TE will be used. When **Exclusive** is selected, calls will be routed from this DTM port to the nominated TE only. If the call cannot be routed through the nominated TE then the call will be cleared.

Note *The terms outgoing and incoming are with respect to a TE port. Thus the term outgoing applies to a call that originates from a device that is connected to an NT port.*

Edit lines Edit Lines allows access to the Line Configuration dialog for each data (B/subB) channel. This is normally not used because the NT port usually receives call parameters from its attached TE. Refer to *Line configuration* on page 2-5 for further information.

Applications Select this to connect to either a VX DTM NT port (default) or a DTM TE port.

Channel Negotiation Enable if you want channel negotiation on incoming calls.

When this parameter is **enabled**, a 'preferred' channel is selected. The network may however want to use the designated channel to set up its own outgoing call and will require the incoming call to be set up on a different channel. In this case the network will attempt to negotiate the incoming call to another channel. This feature helps to prevent calls from being rejected should a call collision occur.

When **disabled**, the call from the device specifies an 'exclusive' channel, and if a collision occurs, the incoming call is rejected.

TE Options

The TE-specific options consist of two check boxes: T309 Disable and Retain layer 2 establishment.

T309 Disable When enabled (default), the T309 is a timer that causes calls to clear 15 seconds after a D-channel link failure is detected. If it is disabled, calls clear as soon as D-channel link failure is detected.

Retain layer 2 establishment When enabled (default) this option tells the DTM port to maintain the D-channel LAPD data link in the established state at all times. When disabled the data link is maintained only while it is required for call signalling. The setting of this option affects call processing as follows:

- When Retain layer 2 establishment is enabled: If the D-channel LAPD data link fails, the DTM will report a Layer 2 failure alarm. The CM or RM will not submit outgoing calls to the port until the alarm is cleared by Layer 2 being re-established.
- When Retain layer 2 establishment is disabled: The CM or RM can submit calls to the port regardless of the status of Layer 2. If the D-channel LAPD data link cannot re-establish, the CM or RM will resubmit the call to another network port. The process of detecting failure to re-establish the data link takes four seconds if T309 is disabled or 15 seconds if T309 is enabled.

Notes

- *Retain layer 2 establishment should be enabled for unused DTM ports. This ensures that a layer 2 alarm condition exists so that the CM or RM will not route outgoing calls to the port.*
 - *When Retain layer 2 establishment is disabled, T309 should also be disabled so as to avoid long delays in call resubmission.*
-

Exclusive/Preferred use only Enabling this removes the module from the TE table. The TE can then only be used for outgoing calls if the NT making the call has **Outgoing TE exclusive** selected.

NT/TE common options

There are four options common to both NT and TE. These options are: 'Don't use as clock reference', 'Trunk Quality Reporting', 'Don't use as bus master' and 'Don't use for D-channel calls'.

Don't use as clock reference This option prevents a DTM port from being selected as the backplane reference clock provider. When a port is chosen to be the reference clock provider, the reference clock is phase locked to the trunk port's incoming clock, according to the table below.

Interface Type	Incoming Clock
DCE X.21	X
DTE X.21	S
DCE V.35	ETxC
DTE V.35	RxC

If the incoming trunk clock becomes unstable or non-existent while the DTM port is providing the reference clock to the backplane, the DTM free runs.

Note *Care should be taken when both ends of a DTM trunk are providing reference clock to the backplanes. This is because both ports will be deriving their clocks from each other, forming a feedback loop. This situation can be avoided by allowing only one end of a trunk link the ability to provide reference clocks to the backplane.*

Don't use for D-channel calls Enable this to prevent the Resource Manager or Control Module from delivering outgoing D-channel calls through the DTM.

Don't use as bus master Enable this to prevent the DTM being selected to provide the backplane bus clock.

Trunk quality reporting

Trunk quality reporting has three main parameters:

- Report Type: Trunk Quality Log and Trunk Quality Alarm
- Reporting Period
- Alarm Thresholds.

Report Type

There are two options: Trunk Quality Log and Trunk Quality Alarm. The Reporting Period and Alarm Thresholds areas are unavailable when both Report Types are unselected (default).

Trunk Quality Log When Trunk Quality Log is selected, the DTM continuously reports Trunk Quality Data over time periods as specified by the Reporting Period.

Trunk Quality Alarm When Trunk Quality Alarm is selected, the DTM continuously reports Trunk Quality Alarms over time periods as specified by the Reporting Period. The contents of the logged alarms reflect whether each specified Alarm Threshold has or has not been reached. Normal alarms are also generated when Trunk Quality Alarm is selected and a transition occurs for any of the Alarm Threshold selections (see *Alarms* on page 4-84).

Reporting period When either or both Report Types are selected, the Reporting Period is activated. The selections for the Reporting Period are: 1 minute, 15 minutes, 1 hour and 12 hours.

Alarm thresholds When Trunk Quality Alarm is selected, the Alarm Thresholds section is also activated. The data field selections for the Alarm Thresholds are: Errored Seconds, CRC Errors, Sync Losses, Receive Slips and Transmit Slips.

V.24 Asynchronous Digital Line Module (ADLM V.24)

The V.24 Asynchronous Digital Line Module (ADLM V.24) is a V.24 DCE as defined in the ITU-T (CCITT) recommendations. It can connect two asynchronous V.24 DTEs to the ISDN, or DCEs if null modem cables are used. These DTEs can be computers, modems, or other devices which communicate using asynchronous V.24 (or RS-232) data formats.

The ADLM V.24 simulates the operation of an AT compatible modem by providing a subset of the AT command set which is translated to the ISDN D-channel protocol. See *Appendix C — AT Commands* for more details.

The ADLM V.24 provides data rate adaption of user data to the 64 kbit/s ISDN B-channel, or nx8 kbit/s sub-channels, using I.463 (V.110) and ECMA 102 compatible data rate adaption.

In Australia, the data word size can be 8-bit, 7-bit with parity or 7-bit without parity. Parity is only allowed for 7-bit data (in accordance with ECMA 102 and TPH 1856 specifications). The parity bit can be odd, even, forced to 0, or forced to 1. The number of stop bits can be either 1 or 2.

In the UK, parity is not used.

The ADLM V.24 allows you to select normal or Hotline dialling, or semipermanent operation for each V.24 port. You can also choose to establish and clear calls using either the AT command set, the DTR signal alone or OmniVision.

Note Refer to *Appendix C* for a list of the available AT commands.

Configuration

Line Number	Enter the line number for this line. See <i>Parameters</i> , page 2-5 for information about the Line Number.
Subaddress	Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.
Port ID	Enter the Port ID if required. The default setting for this parameter is 0. See <i>Parameters</i> , page 2-5 for more information. It is not mandatory to enter anything in this parameter.
Port Name	Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. The following table shows the default settings. Accept the defaults for all the parameters.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults for both incoming calls and outgoing calls are 'disabled'. For further information about the default settings, refer to *Parameters*, page 2-7.

ISDN Interface — High Layer Compatibility defaults

	Parameter	Default
Call Direction	Incoming	None
	Outgoing	None

ISDN Interface — Call Establishment Modes defaults

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Backup	Disabled
	Label	—
Member of Subrate Virtual Line or manually connected (UK only)	Yes	Disabled
Hotline	Enable	—
	Number	—
	XSPC	Disabled
	Subaddress	—
Active Call Facilities	Don't release call if V110 frame loss detected	Disabled
Call Control	DTR blocking	Disabled
	Auto Answer	Disabled
	Release on progress	Enabled
	AT	Disabled
	DTR call	Enabled
GSM	Pass CLI in	Disabled
	Pass CLI out	Disabled
	Pass B number in	Disabled
	Pass Causes	Disabled
Control Leads	Local Only	Enabled
	Transferred Inband	Disabled
	GSM mode	Disabled
DTR Alarm Timeout	—	5 secs

Changing Call Establishment Modes defaults

Semipermanent (where available)

Select **Enable** and enter the semipermanent label to set up a semipermanent call from the V.24 line.

Select **Backup** to replace the SPC with a switched call if the V.110 three second disconnection rule is broken. When the Backup call is initiated, an alarm is invoked (if enabled).

To set up the semipermanent backup operation

- 1 Enter an SPC label at both ends of the link and select **Enable**.
- 2 Select **Backup** at both ends of the link.
- 3 Set up a Hotline call to the line number at the other end of the link.
- 4 Select **Auto Answer** at the other end of the link.

Member of Subrate Virtual Line or manually connected (UK only)

Select **Yes** if the line is a part of a Subrate Virtual Line or manually connected. It enables data to be transferred directly onto the B-channel without the call being set up. If you do not select this, the port does not transmit or receive data until a call is received or initiated by the port.

Hotline

To set up a Hotline call, enter the called party number and select **Enable**. Enter the subaddress, if required.

Select **XSPC** to enable a Switched semipermanent connection. This is only available with a Hotline call.

Active Call Facilities

Select **Don't release call if V110 frame loss detected** to ignore the V.110 three second disconnection rule.

Call Control

If you clear all of the Call Control checkboxes, the ADLM only answers calls to this line when the DTR signal is on.

Select **AT** to use the AT protocol for call control between the DTE and the ADLM.

Select **DTR Call** to initiate and clear calls with the DTR lead.

Select **DTR Blocking** to accept incoming calls when DTR lead is high and to fail calls when DTR lead is low.

Set **Release on progress** to release a call even if a Progress Indicator is sent.

Note *You cannot select both AT and DTR Call at the same time.*

Control Leads

Select **Local Only** to turn all leads on after calls are started.

Select **Transferred Inband** to transmit the state of the control leads to the other end.

If you have not selected this, an **ON** state will be transmitted for each lead irrespective of the actual states presented to the port.

GSM mode

Select if you want end-to-end control mapping for GSM applications.

GSM

Set **Pass CLI In** to allow the CLI (Calling Line Identifier/ Calling Party Number) from the ADLM port to pass to the user TE to which it is physically connected. If disabled, the CLI numbers will not be passed. This setting only affects incoming ISDN calls to the ADLM port and not the outgoing ones.

Set **Pass CLI Out** to allow for the CLI numbers from the user TE originating the call to pass to the ADLM port to which it is physically connected. If disabled, an error indication is returned by the ADLM port to the user TE to which it is physically connected, and the initiated call by the user TE is rejected. This setting only applies to outgoing ISDN calls from the ADLM and not the incoming ones.

Note *When set, **Pass CLI Out** causes the port to be seized as soon as the 'ATD' command is received. This reduces the chance of call collisions*

Set **Pass B number in** to allow the Called Party Number from the ADLM port to be passed to the user TE. If disabled, B numbers are not passed. This setting only affects incoming ISDN calls to the ADLM.

Set **Pass causes** to allow the CAUSES of the termination of ISDN calls from the ADLM port to pass to the USER TE. If disabled, causes are not passed. This setting is effective only if ADLM port is not the initiator of the termination of the ISDN call.

DTR alarm timeout This parameter is effective only when DTR blocking mode is enabled. Four settings are available: 0, 5, 10 and 30 seconds. If DTR goes to a low state for more than the time specified, an alarm is generated by the RM or CM.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC (where available) Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via Call Log within the NMS. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID Do not select **CLIP** (CLI Presentation).
If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

CLIVE List Assignment Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

Teleservice (where available) None.

Changing ISDN Interface defaults

Teleservice (where available) Select **Timelink** if you have subscribed to this service.

ISDN Internal Control defaults

Restriction None.
Outgoing TE Any.

Changing ISDN Internal Control defaults

Restriction Select **Disable** from the list to prevent calls being made to or from this line.
 Select **In Only** or **Out Only** to allow calls in the selected direction only.

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

V.24 Synchronous Digital Line Module (SDLM V.24)

The V.24 Synchronous Digital Line Module (SDLM V.24) is a V.24 DCE as defined in the ITU-T (CCITT) recommendations. It provides interfaces for two synchronous V.24 DTEs to the ISDN. The DTEs can be computers, terminals, or other devices which communicate using synchronous V.24 (or RS-232) data formats.

The SDLM V.24 provides data rate adaption of user data to the 64 kbit/s ISDN B-channel, or nx8 kbit/s subchannel, using ITU-T (CCITT) X.30 and I.463 (V.110) and ECMA 102 compatible data rate adaption.

Different signalling modes can be configured for an SDLM V.24 interface, and each interface on a module can be configured differently. In each case, translation to the appropriate ISDN signalling is performed by the device.

Hotline dialling

When Hotline dialling is used, the SDLM V.24 is perceived by the DTE as a leased line modem. The number to be called by a particular SDLM V.24 interface is configured before any calls are made. The DTE can make a call by raising the DTR signal. The device sends a call setup message on the ISDN link to the exchange with the complete called party number. The call is connected to the remote party and when end-to-end communication is established the DCD signal is raised by the SDLM. You can also initiate and clear calls through OmniVision.

When DTR Call is enabled as the call control method the DTE can disconnect a Hotline call by dropping the DTR signal.

Semipermanent operation (where available)

Semipermanent operation is fully supported in Australia. In semipermanent operation, the SDLM is perceived by the DTE as a leased line modem.

Configuration

Line Number	Enter the line number for this line. See <i>Parameters</i> , page 2-5 for further information.
Subaddress	Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.
Port ID	Enter the Port ID if required. The default setting for this parameter is 0. See <i>Parameters</i> , page 2-5 for more information. It is not mandatory to enter anything in this parameter.
Port Name	Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.

ISDN Interface — Bearer Capability defaults

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. The following table shows the default settings. Accept the defaults for all the parameters.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults for both incoming calls and outgoing calls are 'disabled'. For further information about the default settings, refer to *Parameters*, page 2-7.

ISDN Interface — High Layer Compatibility defaults

	Parameter	Default
Call Direction	Incoming	None
	Outgoing	None

ISDN Interface — Call Establishment Modes defaults

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Backup	Disabled
	Label	—
Member of Subrate Virtual Line or manually connected (UK)	Yes	Disabled
Hotline	Enable	—
	Number	—
	XSPC	Disabled
	Subaddress	—
Active Call Facilities	Don't release call if V110 frame loss detected	Disabled
Call Control	Auto Answer	Disabled
	DTR Call	Enabled
Control Leads	Local Only	Enabled
	Transferred Inband	Disabled

Changing Call Establishment Modes defaults

Semipermanent (where available)	<p>Select Enable and enter the semipermanent label at both ends to set up a semipermanent call from the V.24 line.</p> <p>You can select Backup to replace the semipermanent connection with a switched call if the V.110 three second disconnection rule is broken. When the Backup call is initiated, an alarm is invoked.</p> <p>To set up the semipermanent backup operation</p> <ol style="list-style-type: none"> 1 Enter an SPC label at both ends of the link and select Enable. 2 Select Backup at both ends of the link. 3 Set up a Hotline call to the line number at the other end of the link. 4 Select Auto Answer at the other end of the link.
Member of Subrate Virtual Line or manually connected	<p>Select Yes if the line is a part of a Subrate Virtual Line or manually connected. It enables data to be transferred directly onto the B-channel without the call being set up.</p> <p>If you do not select this, the port does not transmit or receive data until a call is received or initiated by the port.</p>
Hotline	<p>To set up a Hotline call, enter the called party number and select Enable. Enter the subaddress, if required.</p> <p>Select XSPC to enable a Switched Semi-Permanent Connection.</p>
Active Call Facilities	<p>Select Don't release call if V110 frame loss detected if you want to ignore the V.110 three second disconnection rule.</p>
Call Control	<p>Select DTR Call to use the DTR signal to initiate and clear calls on this line.</p> <p>Select Auto Answer if you want incoming calls to be connected automatically.</p> <p>If you clear all of the Call Control checkboxes, the SDLM only answers calls to this line when the DTR signal is on.</p>
Control Leads	<p>Select Local Only to turn all leads on after calls are started.</p> <p>Select Transferred Inband to transmit the state of the control leads to the other end. This only occurs if a V.110 rate (LLC) is selected. If this is not selected, an ON state will be transmitted for each lead irrespective of the actual states presented to the port.</p>

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC (where available)

Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*. This is only available if you subscribe to an AOC service.

Calling Line ID

Do not select **CLIP** (CLI Presentation).

If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

CLIVE List Assignment

Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format

Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

Teleservice
(where available) None.

Changing ISDN Interface defaults

Teleservice
(where available) Select **Timelink** if you subscribe to this service.

ISDN Internal Control defaults

Restriction None.
Outgoing TE Any.

Changing ISDN Internal Control defaults

- Restriction** Select **Disable** from the list to prevent calls being made to or from this line.
Select **In Only** or **Out Only** to allow calls in the selected direction only.
- Outgoing TE** Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

X.21 Synchronous Digital Line Module (SDLM X.21)

The X.21 Synchronous Digital Line Module (SDLM X.21) is an X.21 DCE as defined in the recommendations of ITU-T (CCITT). It can connect two X.21 DTEs to the ISDN, providing:

- conversion of the X.21 protocol to the ISDN protocol
- bit rate adaption from the user rate to the B-channel rate at 64 kbit/s or nx8 kbit/s.

The SDLM X.21 allows you to select normal or Hotline dialling, or semipermanent operation for each X.21 port. You can also choose to establish and clear calls using either full X.21 protocols, the C signal alone or OmniVision.

For further discussion on X.21 interfaces, refer to *Appendix G*.

Configuration

Line Number	Enter the line number for this line. See <i>Parameters</i> , page 2-5 for further information.
Subaddress	Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.
Port ID	Enter the Port ID if required. The default setting for this parameter is 0. See <i>Parameters</i> , page 2-5 for more information. It is not mandatory to enter anything in this parameter.
Port Name	Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. The following table shows the default settings. Accept the defaults for all the parameters.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults for both incoming calls and outgoing calls are 'disabled'. For further information about the default settings, refer to *Parameters*, page 2-7.

ISDN Interface — High Layer Compatibility defaults

	Parameter	Default
Call Direction	Incoming	None
	Outgoing	None

ISDN Interface — Call Establishment Modes defaults

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Backup	Disabled
	Label	—
Member of Subrate Virtual Line or manually connected (UK only)	Yes	Disabled
Hotline	Enable	—
	Number	—
	XSPC	Disabled
	Subaddress	—
Active Call Facilities	Don't release call if V.110 frame loss detected	Disabled
Call Control	Auto Answer	Disabled
	X.21 Call	Disabled
	C Signal	Enabled
Control Leads	Local Only	Enabled
	Transferred Inband	Disabled

Changing Call Establishment Modes defaults

Semipermanent Select **Enable** and enter the semipermanent label to set up a semipermanent call from the X.21 line.

Select **Backup** to replace the SPC with a switched call if the V.110 three second disconnection rule is broken. When the Backup call is initiated, an alarm is invoked (if enabled).

To set up the semipermanent backup operation

- 1 Enter an SPC label at both ends of the link and select **Enable**.
- 2 Select **Backup** at both ends of the link.
- 3 Set up a Hotline call to the line number at the other end of the link.
- 4 Select **Auto Answer** at the other end of the link.

- Member of Subrate Virtual Line or manually connect (UK only)** Select **Yes** if the line is a part of a Subrate Virtual Line or manually connected. It enables data to be transferred directly onto the B-channel without the call being set up.
If you do not select this, the port does not transmit or receive data until a call is received or initiated by the port.
- Hotline** To set up a Hotline call, enter the called party number and select **Enable**. Enter the subaddress, if required.
Select **XSPC** to enable a switched semipermanent connection.
- Active Call Facilities** Select **Don't release call if V.110 frame loss detected** if you want to ignore the V.110 three second disconnection rule.
- Call Control** Select **X.21** to use the X.21 protocol for call control between the DTE and the SDLM.
Select **C Signal** to use the C Signal line to initiate and clear calls on this line.
Select **Auto Answer** to connect incoming calls automatically.
If you clear all of the Call Control checkboxes, the SDLM only answers calls to this line when the C signal is on.

Note *You cannot select X.21 and either C Signal or Auto Answer at the same time.*

- Control Leads** Select **Local Only** to turn all leads on after calls are started.
Select **Transferred Inband** to transmit the state of the control leads to the other end. If this is not selected, an **ON** state will be transmitted for each lead irrespective of the actual states presented to the port. Control lead states can only be transferred at data rates less than or equal to 19.2 kbit/s.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Control defaults

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

- AOC (where available)** Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.
- This is only available if you subscribe to an AOC service.
- Calling Line ID** Do not select **CLIP** (CLI Presentation).
- If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.
- CLIVE List Assignment** Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.
- Numbering format** Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

- Teleservice (where available)** None.

Changing ISDN Interface defaults

- Teleservice (where available)** Select **Timelink** if you have subscribed to this service.

ISDN Internal Control defaults

- Restriction** None.
- Outgoing TE** Any.

Changing ISDN Internal Control defaults

- Restriction** Select **Disable** from the list to prevent calls being made to or from this line. Select **In Only** or **Out Only** to allow calls in the selected direction only.
- Outgoing TE** Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

V.35 Synchronous Digital Line Module (SDLM V.35)

The V.35 Synchronous Digital Line Module (SDLM V.35) provides two DCE ports for connecting DTE to the ISDN. Equipment connecting to these ports must comply with ITU-T (CCITT) V.35 recommendations and be configured to synchronise with the clocking signal supplied by the module.

Each port allows a DTE to communicate at data rates from 600 bit/s to 64 kbit/s for connection onto a digital network. The SDLM V.35 provides data rate adaption of user data to the 64 kbit/s ISDN B-channel, or nx8 kbit/s sub-channels, using I.463 (V.110) and ECMA 102 compatible data rate adaption.

Configuration

- Line Number** Enter the line number for this line. See *Parameters*, page 2-5 for more information.
- Subaddress** Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See *Parameters*, page 2-5 for more information.
- Port ID** Enter the Port ID if required. The default setting for this parameter is 0. See *Parameters*, page 2-5 for more information. It is not mandatory to enter anything in this parameter.
- Port Name** Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See *Parameters*, page 2-5 for more information.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. The following table shows the default settings. Accept the defaults for all the parameters.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults for both incoming calls and outgoing calls are 'disabled'. For further information about the default settings, refer to *Parameters*, page 2-7.

ISDN Interface — High Layer Compatibility defaults

	Parameter	Default
Call Direction	Incoming	None
	Outgoing	None

ISDN Interface — Call Establishment Modes defaults

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Backup	Disabled
	Label	—
Member of Subrate Virtual Line or manually connected (UK only)	Yes	Disabled
Hotline	Enable	—
	Number	—
	XSPC	Disabled
	Subaddress	—
Active Call Facilities	Don't release call if V.110 frame loss detected	Disabled
Call Control	Auto Answer	Disabled
	RS Call	Enabled
Control Leads	Local Only	Enabled
	Transferred Inband	Disabled

Changing Call Establishment Modes defaults

Semipermanent (where available)

Select **Enable** and enter the semipermanent label to set up a semipermanent call from the V.35 line.

Select **Backup** to replace the semipermanent connection with a switched call if the V.110 three second disconnection rule is broken. When the Backup call is initiated, an alarm is invoked.

To set up the semipermanent backup operation

- 1 Enter an SPC label at both ends of the link and select **Enable**.
- 2 Select **Backup** at both ends of the link.
- 3 Set up a Hotline call to the line number at the other end of the link.
- 4 Select **Auto Answer** at the other end of the link.

Member of Subrate Virtual Line or manually connect

Select **Yes** if the line is a part of a Subrate Virtual Line or manually connected. It enables data to be transferred directly onto the B-channel without the call being set up.

If you do not select this, the port does not transmit or receive data until a call is received or initiated by the port.

Hotline

To set up a Hotline call, enter the called party number and select **Enable**. Enter the subaddress, if required.

Select **XSPC** to enable a switched semipermanent connection.

Active Call Facilities

Select **Don't release call if V110 frame loss detected** if you want to ignore the V.110 three second disconnection rule.

Call Control Clear the **RS Call** checkbox if you do not want to use the RS signal to make and clear calls on this line.

Select **Auto Answer** to connect incoming calls automatically.

If you clear all of the Call Control checkboxes, the SDLM only answers calls to this line when the RS signal is on.

Control Leads Clear **Local Only** if you do not want to turn all leads on after calls are started.

Select **Transferred Inband** to transmit the state of the control leads to the other end. This only occurs if a V.110 rate (LLC) is selected. If this is not selected, an **ON** state will be transmitted for each lead irrespective of the actual states presented to the port.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID Do not select **CLIP** (CLI Presentation).

If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

CLIVE List Assignment Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

**Teleservice
(where available)** None.

Changing ISDN Interface defaults

Teleservice (where available) Select **Timelink** if you have subscribed to this service.

ISDN Internal Control defaults

Restriction None.

Outgoing TE Any.

Changing ISDN Internal Control defaults

Restriction Select **Disable** to prevent calls being made to or from this line.
Select **In Only** or **Out Only** to allow calls in the selected direction only.

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

Quad Digital Line Module (QDLM)

The Quad Digital Line Module (QDLM) combines most of the functions of the ADLM and SDLMs on the same module. It has four ports each of which provides either an X.21, V.24 or V.35 interface. They can all operate as synchronous DCE ports at data rates from 600 bit/s to 64 kbit/s. The V.24 interface can operate asynchronously from 300 bit/s to 38.4 kbit/s with programmable data format.

The QDLM adapts the user data to a 64 kbit/s B-channel in accordance with ITU-T (CCITT) recommendations V.100 and X.30. This allows the user data to be carried on digital leased lines or switched digital connections.

The QDLM adds features for data devices such as:

- DTE operating mode; that is, 'tail circuit' modems and NTUs can be connected
- Automatic back-up of Leased Lines
- Transfer of interface control leads at speeds up to 56 kbits/s
- Bit Error Rate Testing (BERT).

Configuration

Line Number	Enter the line number for this line. See <i>Parameters</i> , page 2-5 for more information.
Subaddress	Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.
Port ID	Enter the Port ID if required. The default setting for this parameter is 0. See <i>Parameters</i> , page 2-5 for more information. It is not mandatory to enter anything in this parameter.
Port Name	Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. This information element is used by the ISDN to determine whether or not interworking equipment is necessary. It does not affect the characteristics of the QDLM interfaces. Usually, only octets 3 and 4 are required. The following table shows the default settings. If the other octets are required they should be set to the same values as the Low Layer Compatibility information element. The default values are suitable for almost every application.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital
Octet 4	Mode	Circuit
	Rate	64 kbit/s

ISDN Interface — Low Layer Compatibility

The Low Layer Compatibility information element is transmitted by the device when setting up an outgoing call, and optionally is used to check incoming calls. It is used to configure the data rate and asynchronous data format. The default values are shown below for Low Layer Compatibility at speeds less than 64 kbit/s where V.110/X.30 rate adaption is required.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest. Digital
Octet 4	Mode	Circuit
	Outgoing	None
Octet 5	Layer 1	CCITT (ITU-T) rate adaption

The settings for these parameters are determined by the capability of the equipment used. The defaults for both incoming calls and outgoing calls are 'disabled'. For further information about the default settings, refer to *Parameters* page 2-7.

User Rate Select the required rate. The rates available are:

- 300 (asynchronous only)
- 600, 1200, 2400, 4800, 9600, 19200, 38400 (synchronous and asynchronous)
- 48000, 56000, 64000 (synchronous only).

ISDN Interface — High Layer Compatibility defaults

	Parameter	Default
Call Direction	Incoming	None
	Outgoing	None

ISDN Interface — Call Establishment Modes

This information is used to control how the QDLM establishes outgoing calls and how it accepts incoming calls. See the previous sections on the *ADLM V.24*, *SDLM V.24*, *SDLM X.21* and *SDLM V.35* for detailed information on Call Establishment Modes for each type of DTE.

Before an outgoing call can be made using Hotline mode, the Hotline number and the Hotline subaddress must be entered if required. Hotline mode must then be enabled.

For lead call control, the DTR Call option must be enabled for V.24 and V.35 interfaces. The C Call option must be enabled for X.21 interfaces. All other options must be disabled with the exception of Auto Answer which may be either enabled or disabled.

A DTE with synchronous X.21 interface can set up, answer or clear a switched call by exchanging signals with the QDLM port. When using this X.21 call control, the X.21 option must be enabled and all other Call Control options must be disabled. This allows the attached DTE to communicate with the QDLM according to the X.21 call establishment protocol recommended by the ITU-T (CCITT).

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Backup	Disabled
	Label	—
Member of Subrate Virtual Line or manually connected	Yes	Disabled
Hotline	Enable	Disabled
	Number	—
	XSPC	Disabled
	Subaddress	—
Active Call Facilities	Don't release call if V.110 frame loss detected	Disabled
Call Control	C Blocking	Disabled
	Auto Answer	Disabled
	DTR or C Call	Enabled
	V.25 bis	Disabled
	X.21	Disabled
	AT	Disabled
Control Leads	Local Only	Enabled
	Transferred Inband	Disabled
	GSM Mode	Disabled

Changing Call Establishment Modes defaults

See the previous sections on the *ADLM V.24*, *SDLM V.24*, *SDLM X.21* and *SDLM V.35* for detailed information on changing call establishment modes for each type of interface.

Semipermanent (where available)

Enter the semipermanent number in the **Label** parameter and select **Enable**.

Hotline

Enter the Hotline number and the subaddress number, if required, and select **Enable**.
Select **XSPC** to enable a Switched Semipermanent Connection.

- Call Control** Select **V.25 bis** to use Cisco V.25 bis (software release version 10) protocol for call control between the DTE and the QDLM. All other Call Control options must be disabled. This is available over X.21, V.35 and V.24 synchronous line interfaces. Selecting V.25 bis allows DTEs to make outgoing calls through the QDLM.
- Select **C Blocking** to accept incoming calls when C lead is high and to fail calls when C lead is low.
- GSM Mode** Enable GSM mode to provide mapping of end-to-end control leads to interwork with a GSM data service.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

- AOC (where available)** Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.
- This is only available if you subscribe to an AOC service.
- Calling Line ID** Do not select **CLIP** (CLI Presentation).
- If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.
- CLIVE List Assignment** Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.
- Numbering format** Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

- Teleservice (where available)** None.

Changing ISDN Interface defaults

Teleservice (where available) Select **Timelink** if you have subscribed to this service.

ISDN Internal Control defaults

	V.24	V.35	X.21
Restriction	None	None	None
Operation	DCE	DCE	DCE
Synch mode	Synch	—	—
Tx Clock	DCE	DCE	—
Outgoing TE	Any	Any	Any

Changing ISDN Internal Control defaults

- Restriction** Select **Disable** to prevent calls being made to or from this line. Select **In Only** or **Out Only** to allow calls in the selected direction only.
- Operation** Select DCE or DTE. Separate adaptor cables are available for DCE and DTE operation for V.24 and V.35 interfaces.
- Synch Mode** This mode is only available on QDLM with V.24 interface. It allows you to set the interface to operate in either synchronous or asynchronous mode. Note that to select AT Call Control, Synch Mode must be selected as **Asynch**.
- Tx Clock** Select either **DCE Tx Clock** or **DTE Tx Clock** for the V.24 and V.35 interfaces. The receive clock is always sourced by the DCE. These clock source rules are true when the interface is configured for either DTE or DCE operation.
- Outgoing TE** Select the required TE. Any of the IPMT or BRMT modules in the chassis may be selected.

QDLM Bit Error Rate Testing (BERT)

Bit Error Rate Testing (BERT) is a QDLM diagnostic facility which enables you to check the integrity of a data circuit. It uses the 511 bit pattern specified in ITU-T (CCITT) O.153 to test either a switched or permanent ISDN B-channel connection for errors. (The 511 bit pattern is used by most stand-alone Bit Error Rate Testers, such as the HP4952.)

The QDLM BERT facility can be used between two QDLMs of any interface type and between a QDLM and an SDLM (relevant to some testing applications).

It assumes that:

- the appropriate loopbacks (internal and external to a DLM) are configured
- any required external equipment is installed and
- a call is in place between the QDLM, using BERT and another DLM.

Before running a Bit Error Rate Test, you must establish a connection across the network. You can do this either:

- manually, from OmniVision, before entering Diagnostic Mode or
- automatically, by selecting **Start BERT**. This is only possible if the QDLM is in Hotline mode and a call is not in progress.

To clear the connection, select **Stop BERT**.

Configuration

There are no specific BERT parameters to configure.

Warning

BERT can only operate from one port at a time.
The maximum data rate that QDLM BERT supports is 4800 bit/s.

Refer to the *Installation Manual* for an explanation of how BERT works.

High Speed Data Module (HSDM)

The HSDM Module provides a means of connecting to high speed data networks. The data interfaces on the HSDM typically provide high data rate links to routers and Frame Relay networks.

The HSDM can be used across Subrate and TimeFrame Virtual Lines between nodes in a network.

Configuration

Configuring an HSDM port applies the defaults or chosen settings to all physical and Virtual Lines associated with the port. Individual lines within a port can be configured separately.

Port Number	Enter the port number for this port. See <i>Parameters</i> , page 2-5 for more information.
Subaddress	Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.
Port ID	Enter the Port ID if required. The default setting for this parameter is 0. See <i>Parameters</i> , page 2-5 for more information. It is not mandatory to enter anything in this parameter.
Port Name	Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 and <i>TimeFrame</i> on page 3-56 for more information.

Port Setup

Interface Mode Configures the HSDM port to act as DTE or DCE. The default is DCE.

Note *In DTE mode, the HSDM can provide the Reference Master Clock, which is controlled by the **Don't use a Clock Reference** checkbox.*

Rate Adaption Allows the Rate Adaption method to be selected. See page 2-8 for further information. The type of rate adaption selected will affect the data rates that are available on the port.

Max Port Speed This allows the selection of the following speeds:

- 600, 1200, 2400, 4800, 9600, 19200, 38400 48000, 56000, 64000 (synchronous only)
- n x 8 kbit/s up to 256 kbit/s
- n x 64 kbit/s from 256 kbit/s up to 2048 kbit/s.

Note *The combined port speed of the two data ports cannot exceed 2048 kbit/s.*

Call Establishment Modes

Click on this to open the Call Establishment Modes dialog.

ISDN Interface — Call Establishment Modes

This information is used to control how the HSDM establishes outgoing calls and how it accepts incoming calls. See the sections on the *SDLM V.24*, *SDLM X.21* and *SDLM V.35* for detailed information on Call Establishment Modes for each type of DTE.

Before an outgoing call can be made using Hotline mode, the Hotline number and the Hotline subaddress must be entered if required. Hotline mode must then be enabled.

For lead call control, the DTR Call option must be enabled for V.24 and V.35 interfaces. The C Call option must be enabled for X.21 interfaces. All other options must be disabled with the exception of Auto Answer which may be either enabled or disabled.

A DTE with synchronous X.21 interface can set up, answer or clear a switched call by exchanging signals with the HSDM port. When using this X.21 call control, the X.21 option and CCITT mode must be enabled and all other Call Control options must be disabled. This allows the attached DTE to communicate with the HSDM according to the X.21 call establishment protocol recommended by the ITU-T (CCITT).

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Backup	Disabled
	Label	—
Member of SRVL or manually connected (where available)	Yes	Disabled
Hotline	Enable	Disabled
	Don't increment Hotline No.	Disabled
	XSPC	Disabled
	Number	—
	Subaddress	—
Active Call Facilities	Don't release call if V.110 frame loss detected	Disabled
Call Options	Auto Answer	Disabled
	C Signal	Enabled
	V.25 bis	Disabled
	X.21	Disabled
	C Blocking	Disabled
Control Leads	Local Only	Enabled
	Transferred Inband	Disabled

Changing Call Establishment Modes defaults

See the sections on the *SDLM V.24*, *SDLM X.21* and *SDLM V.35* for detailed information on changing call establishment modes for each type of interface.

Hotline Enter the Hotline number and the subaddress number, if required, and select **Enable**. If a subaddress is not used, the Hotline number for subsequent (64kbit/s) lines is automatically incremented, based on the Hotline number configured. The automatic Hotline number increment can be disabled by checking the **Don't increment Hotline No.** checkbox.

Select **XSPC** to enable a Switched Semipermanent Connection.

Call options Select **V.25 bis** to use Cisco V.25 bis (software release version 10) protocol for call control between the DTE and the HSDM. This is available over X.21, V.35 and V.24 synchronous line interfaces. Selecting V.25 bis allows DTEs to make outgoing calls and accept incoming calls through the HSDM.

Note *CCITT must be enabled before V.25bis can be enabled.*

Advanced

Click on the Advanced button to open a Line Configuration dialog. All configuration data entered in this dialog is applied individually to all lines of the port.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. This information element is used by the ISDN to determine whether or not interworking equipment is necessary. It does not affect the characteristics of the digital interfaces. Usually, only octets 3 and 4 are required. The following table shows the default settings.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital
Octet 4	Mode	Circuit
	Rate	64 kbit/s

ISDN Interface — Low Layer Compatibility

The Low Layer Compatibility information element is transmitted when setting up an outgoing call, and optionally is used to check incoming calls. It is used to configure the data rate and asynchronous data format. The default values are shown below for Low Layer Compatibility at speeds less than 64 kbit/s where V.110/X.30 rate adaption is required.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest. Digital
Octet 4	Mode	Circuit
	Rate	64 kbit/s
Octet 5	Layer 1	CCITT (ITU-T) rate adaption

The settings for these parameters are determined by the capability of the equipment used. The settings are applied to incoming calls only. For further information about the default settings, refer to *Parameters* page 2-7.

ISDN Interface — High Layer Compatibility defaults

	Parameter	Default
Call Direction	Incoming	None
	Outgoing	None

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

- AOC (where available)** Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.
- This is only available if you subscribe to an AOC service.
- Calling Line ID** Do not select **CLIP** (CLI Presentation).
- If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.
- CLIVE List Assignment** Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.
- Numbering format** Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

- Teleservice (where available)** None.

Changing ISDN Interface defaults

- Teleservice (where available)** Select **Timelink** if you have subscribed to this service.

ISDN Internal Control defaults

	V.24	V.35	X.21
Restriction	None	None	None
Operation	DCE	DCE	DCE
Synch mode	Synch	—	—
Tx Clock	DCE	DCE	—
Outgoing TE	Any	Any	Any

Changing ISDN Internal Control defaults

- Restriction** Select **Disable** to prevent calls being made to or from this line. Select **In Only** or **Out Only** to allow calls in the selected direction only.
- Tx Clock** Select either **DCE Tx Clock** or **DTE Tx Clock** for the V.24 and V.35 interfaces. The receive clock is always sourced by the DCE. These clock source rules are true when the interface is configured for either DTE or DCE operation.
- Outgoing TE** Select the required TE. Any of the IPMT, BRMT or ISO-BCAM modules may be selected.

Subrate switching

As with the SRMM, MFCM and LDCM, the Subrate multiplexing function of the submodule can be used to transit-switch Data Leased Lines and Voice Tie Lines. In this case, you may need to change some of the Call Establishment Modes settings to accommodate the type of connection and signalling used, and set up an internal connection within the device.

Leased Line Backup

The Leased Line Backup (LLB) facility is provided by a factory-fitted dual port submodule on the QDLM. LLB allows a switched ISDN B-channel to be used as a backup link for a leased data link. The status of various signals at the DCE/DTE interface are monitored to detect failure of the leased data link, issue appropriate alarms to the Network Management System and automatically set up a backup link. The backup link is disconnected when the leased link recovers.

The term 'Leased Line' is a general term that is used to describe various leased data services such as Telstra's Digital Data Network (DDN), BT's 'Kilostream' service, and so on. The term DCE (Data Circuit-terminating Equipment) is used to refer to the device that provides access to the leased line. It is the point at which a customer connects a DTE (Data Terminal Equipment) such as a computer terminal, multiplexer, front-end-processor, and so on. The DCE could be a modem, network terminating -unit (NTU) or any other device which presents a V.24, V.35 or X.21 DCE interface to the DTE.

Several leased lines can be protected by fitting more LLB units and providing either a Basic Rate or Primary Rate connection to the ISDN, depending on the number of backup links that are required.

Configuration

The configuration of the LLB facility depends on the operation of the particular pieces of equipment. The network manager needs to be aware of the behaviour of both the DTEs and the DCEs during both normal conditions and during fault conditions. The following discussion describes the configuration of the LLB facility, using OmniVision, and gives guidelines for some anticipated situations. The configuration of the top port of the LLB submodule controls the switch-over between the DCE and the ISDN rate adaptor.

The configuration of the bottom port controls the ISDN Data Rate Adaptor that is used during a backup call. It must be configured with the Hotline phone number that is to be used for making backup calls and the required backup data rate, which may be different to the data rate of the leased line. The configuration of the bottom port is similar to normal DLM operation.

V.24 and V.35 Interfaces

There are four areas of control:

- Outgoing Backup Calls
- Multiple Backup Protection
- Incoming Backup Calls
- DCE-DTE Control Signals.

Outgoing Backup Calls

This defines the conditions that constitute failure of the leased line. These conditions can take into account the state of both the DCE and the DTE.

A 'Leased Line Failed' alarm is always generated when all the failure conditions are true for the time defined by 'Alarm condition time'. After a 'Leased Line Failed' alarm is generated, a 'Leased Line OK' alarm is always generated when any of the failure conditions are not true for the time defined by 'Clear alarm condition time'.

A short 'Alarm condition time' enables rapid failure detection, whereas a long 'Clear alarm condition time' ensures stability.

The 'Auto call setup' and 'Auto call clear' checkboxes allow a network manager to enable or disable the automatic setting up and clearing of backup calls. This can be particularly useful if it is desirable for the backup call to remain active until the network manager is absolutely certain that a fault has been rectified.

If a backup call attempt fails, or if an established backup call fails, a 'Backup Call Failed' alarm is given. The ISDN call is retried after a period of 'Alarm condition time'.

The number of ISDN call retries are limited and are discussed in the next section.

Multiple Backup Protection

This allows a network manager to prevent an LLB unit from making too many leased line to ISDN switch-overs and if it occurs, to nominate to stay connected to either the leased line or ISDN. 'Too many switch-overs' is defined as being a certain amount of calls made within a certain time period.

If the 'Enable' checkbox is crossed, multiple backup protection is in operation and the drop-down boxes can be set to define 'too many switch-overs' and the nominated connection state. A failure alarm of 'Multiple Backup Failure' is given if too many switch-overs occur.

Independent of the 'Enable' checkbox, the number of ISDN calls set by the network manager also defines how many calls are attempted before ISDN retry attempts are aborted and a 'Multiple Retry Failure' alarm is given.

Incoming Backup Calls

This defines the conditions under which an incoming backup call is rejected. These conditions take into account the state of the DTE.

If any of the conditions are true and have been true for the time defined by 'Reject condition time', a 'DTE Failure' alarm is given and when an incoming call arrives, the call is rejected and a 'Call Rejected' alarm is given. If, after having accepted a backup call, any of the conditions subsequently becomes true for the time defined by 'Reject condition time', the backup call is cleared if the 'Auto clear after accept' checkbox is crossed and a 'DTE Failed' alarm is then given.

After a 'DTE Failure' alarm has been given, all the conditions must be false for the period defined by 'Accept condition time' before incoming calls can be accepted.

DCE-DTE Control Signals

This allows a network manager to decide whether the CTS, DSR, and CD signals that are presented to the DTE are the real signals from the DCE (or the ISDN rate adaptor during backup) or are set to the 'on' state.

This feature is useful for certain types of DTE (for example, some kinds of front-end processors) which do not like to see these signals change to the 'off' state.

Impact of Timers

The 'Alarm condition time' of an LLB unit should be set greater than the answering end's LLB unit 'Reject condition time'. This is so that, if the need be, the answering end has enough time to reject a backup call made after its DTE failed.

The 'Clear Alarm condition time' of an LLB unit should be set less than the answering end's LLB unit 'Accept condition time'. This ensures that when a DTE becomes OK, the answering end rejects any backup call retries. As a general rule, the LLB unit should be quick to Alarm and slow to clear alarms. This ensures that faults are quickly detected, but cautiously cleared.

Default Configuration

The default settings for LLB have been selected to suit common situations. The only thing that should need to be changed is the 'Alarm condition time'. It should be set to a different value at each end so that a backup call collision does not occur when both ends simultaneously detect a fault condition. This also allows a network manager to nominate which end usually makes the backup call.

The default configuration means that a backup call is made automatically if the following conditions are all true (for 15 seconds).

RxD (from DCE)	= all 1s
CTS (from DCE)	= OFF
CD (from DCE)	= OFF
Test Indicator (from DCE)	= OFF
DTR (from DTE)	= ON

The backup call is also made if either the Tx or Rx clock signals from the DCE fail for 15 seconds. The call is cleared automatically if the failure condition goes away for more than 25 seconds.

If a backup call fails, there are up to six call retries at 15 second intervals. If the leased line fails six times in one hour, the backup call is not released automatically.

The answering end rejects the call if any one of the following are true (for 10 seconds):

DTR (from DTE)	= OFF
Test Indicator (from DCE)	= ON

The call is accepted independently of the reject conditions if either the Tx or Rx clock signals from the DCE fail for 10 seconds. The answering end automatically clears the backup call if these conditions become not true for 30 seconds.

Configuration for 56 or 64 kbit/s

At low speeds a DCE is usually able to sense a fault in the link because the signal between the DCEs usually contains a framing signal that can be used to monitor the quality of the link. At higher speeds this is not necessarily the case because there may not be any framing signal used between the DCEs at these rates. This means that the CTS, DSR and CD signals from the DCE are unreliable for indicating failure of the link and should be ignored. The received data is the only end-to-end signal that can be used to sense link failure. For this reason the default configuration at 56 and 64 kbit/s uses only the data and clocks for deciding to make a backup call.

If there are no transitions of the received data signal, then the backup call is made. In this situation it is very important that the answering end be able to reject the backup call because it may be that the DTE at that end is not producing data transitions and is therefore the cause of the apparently faulty link. In this case the link has not really failed at all and a backup is not required. If this condition persists, there are backup call retries at 15 second intervals that are rejected at the answering end until a 'Multiple Retry Failure' alarm is given. The various criteria available for rejecting a backup call are given on page 4-125.

The backup call should be ideally configured to operate at 48 kbit/s so that failure of the backup link can be detected by the V.110 framing that is used to carry user data over the 64 kbit/s ISDN B-channel.

X.21 Interfaces

The control panel for X.21 differs from that of V.24 and V.35 because of the differing signals, however the same rationale for X.21 low and high data rates' configuration is applicable.

Manual Backup Calls

Manual backup calls can be set up or released using OmniVision. See your *OmniVision User Manual* for details.

If manual calls are used for testing the backup path, then the automatic call controls ('Auto call setup', 'Auto call clear', and 'Auto clear after accept') should be switched off, otherwise the backup call is automatically cleared if there is no existing fault condition.

Default settings for Leased Line Backup Configuration

V.24 or V.35 up to 48 kbit/s

Alarm condition time default

15 seconds.

Changing Default Setting for Alarm Condition Time

Enter any value of time within the range 1 to 255 seconds.

Outgoing Backup Call Alarm Criteria defaults

Alarm criteria	Setting
RxD	All 1s
DSR	don't care
TI	OFF
CTS	OFF
TxD	don't care
RTS	don't care
DTR	ON
CD	OFF

Changing default setting for Outgoing Backup Calls

An alarm is generated if all conditions specified are true for the Alarm Condition Time. Enter your alternative conditions in the selection boxes: RxD, DSR, TI, CTS, TxD, RTS, DTR, CD.

Incoming Backup Call Reject Criteria defaults

Alarm criteria	Setting
RTS	don't care
TI	ON
TxD	don't care
DTR	OFF

Changing default setting for Incoming Backup Calls

Incoming calls are rejected if any of the criteria you nominate are satisfied for the Reject Condition Time. Incoming calls are cleared if a backup call is made. Enter your criteria in the selection boxes: RTS, TI, TxD, DTR.

Reject condition time default

15 seconds.

Changing default setting for Reject condition time

Enter any value of time within the range 1 to 255 seconds.

Backup operation defaults

Operation	Setting
Auto call setup	Enabled
Auto call clear	Enabled

Changing default setting for Backup operation

Disable **Auto call setup** if you do not require an alarm to automatically trigger a backup call.

Disable **Auto call clear** if you do not want the backup call to be cleared as soon as any of the alarm conditions no longer hold.

DCE-DTE Ctrl Signals defaults

Signal	Setting
Pass from DCE-DTE	Enabled
Force DSR, DCD, CTS on	Disabled

Changing default setting for DCE-DTE Ctrl Signals

You have the option of forcing on the values of DCD, DSR, CTS. By default, the DCE control signal is passed transparently from the DCE to the DTE, through the Leased Line Backup interface.

Select **Force DSR, DCD, CTS on** when these control leads are assumed to be always on by the DTE.

Incoming operation default

Operation	Setting
Auto clear after accept	Enabled

Changing default setting for Auto clear after accept

Disable the **Auto clear after accept** if you do not require an accepted call to be automatically cleared when the conditions causing a rejection hold.

V.24 or V.35 at 56 or 64 kbit/s**Alarm condition time default**

15 seconds.

Changing default setting for Alarm condition time

Enter any value of time within the range 1 to 255 seconds.

Outgoing Backup Calls Alarm Criteria defaults

Alarm criteria	Setting
RxD	All 0s or 1s
DSR	don't care
TI	OFF
CTS	don't care
TxD	don't care
RTS	don't care
DTR	ON
CD	don't care

Changing default setting for Outgoing Backup Calls

An alarm is generated if all conditions specified are true for the Alarm Condition Time. Enter your alternative conditions in the selection boxes: RxD, DSR, TI, CTS, TxD, RTS, DTR, CD.

Incoming Backup Call Reject Criteria defaults

Alarm criteria	Setting
RTS	don't care
TI	ON
TxD	all 0s or 1s
DTR	OFF

Changing default setting for Incoming Backup Calls

A call is rejected if any condition specified is true. Enter your alternative conditions in any of the selection boxes: RTS, TI, TxD, DTR.

Reject condition time default

15 seconds.

Changing default setting for Reject condition time

Enter any value of time within the range 1 to 255 seconds.

Backup operation defaults

Operation	Setting
Auto call setup	Enabled
Auto call clear	Enabled

Changing default setting for Backup operation

Disable **Auto call setup** if you do not want an alarm call to automatically trigger a backup call.

Disable **Auto call clear** if you do not require the backup call to be cleared as soon as any of the alarm conditions no longer hold.

DCE-DTE Ctrl Signals defaults

Signal	Setting
Pass from DCE-DTE	Enabled
Force DSR, DCD, CTS on	Disabled

Changing default setting for DCE-DTE Ctrl Signals

You have the option of forcing on the values of DCD, DSR, CTS. By default, the DCE control signal is passed transparently from the DCE to the DTE, through the Leased Line Backup interface.

Select **Force DSR, DCD, CTS on** when these control leads are assumed to be always on by the DTE.

Incoming operation default

Operation	Setting
Auto clear after accept	Enabled

Changing default setting for Auto clear after accept

Disable the **Auto clear after accept** if you do not require an incoming call to be automatically accepted when the conditions causing its rejection no longer hold, or if you do not want the call to be cleared whenever the clear/reject criteria no longer hold.

X.21 up to 48 kbit/s**Alarm condition time default**

15 seconds.

Changing default setting for alarm condition time

Enter any value of time within the range 1 to 255 seconds.

Outgoing Backup Calls Alarm Criteria defaults

Alarm criteria	Setting
R	All 1s
I	OFF
T	don't care
C	ON

Changing default setting for Outgoing Backup Calls

An alarm is generated if all conditions specified are true for the Alarm Condition Time. Enter your alternative conditions in the selection boxes: R, I, T, C.

Incoming Backup Calls Reject Criteria defaults

Alarm criteria	Setting
T	don't care
C	OFF

Changing default setting for Incoming Backup Calls

A call is rejected or cleared if any of the conditions you select are true for the Reject Condition Time. Enter your criteria in the selection boxes: T, C.

Reject condition time default

15 seconds.

Changing default setting for Reject condition time

Enter any value of time within the range 1 to 255 seconds.

Backup operation defaults

Operation	Setting
Auto call setup	Enabled
Auto call clear	Enabled

Changing default setting for Backup operation

Disable **Auto call setup** if you do not require an alarm to automatically trigger a backup call.

Disable **Auto call clear** if you do not require the backup call to be cleared as soon as any of the alarm conditions no longer hold.

DCE-DTE Ctrl Signals defaults

Signal	Setting
Pass from DCE-DTE	Enabled
Force C on	Disabled

Changing default setting for DCE-DTE Ctrl Signals

You have the option of forcing on the value of I. By default, the DCE control signal is passed transparently from the DCE to the DTE, through the Leased Line Backup interface.

In situations where the I level cannot be reliably detected or passed to the DTE, you have the option of forcing I on.

Incoming operation default

Operation	Setting
Auto clear after accept	Enabled

Changing default setting for Auto clear after accept

Disable the **Auto clear after accept** if you do not require an incoming call to be automatically accepted and an alarm cleared when the conditions causing its rejection no longer hold.

X.21 56 and 64 kbit/s**Alarm condition time default**

15 seconds.

Changing default setting for Alarm condition time

Enter any value of time within the range 1 to 255 seconds.

Outgoing Backup Calls Alarm Criteria defaults

Alarm criteria	Setting
R	All 0s or all 1s
I	don't care
T	don't care
C	ON

Changing default setting for Outgoing Backup Calls

An alarm is generated if all conditions specified are true for the Alarm Condition Time. Enter your alternative conditions in the selection boxes: R, I, T, C.

Incoming Backup Call Reject Criteria defaults

Alarm criteria	Setting
T	All 0s or all 1s
C	Off

Changing default setting for Incoming Backup Calls

A call is rejected or cleared if any of the conditions you select are true for the Reject Condition Time. Enter your criteria in the selection boxes: T, C.

Reject condition time default

10 seconds.

Changing default setting for Reject condition time

Enter any value of time within the range 1 to 255 seconds.

Backup operation defaults

Operation	Setting
Auto call setup	Enabled
Auto call clear	Enabled

Changing default setting for Backup operation

Disable **Auto call setup** if you do not require an alarm call to automatically trigger a backup call.

Disable **Auto call clear** if you do not want the backup call to be cleared as soon as any of the alarm conditions no longer hold.

DCE-DTE Ctrl Signals defaults

Signal	Setting
Pass from DCE-DTE	Enabled
Force C on	Disabled

Changing default setting for DCE-DTE Ctrl Signals

You have the option of forcing on the value of I. By default, the DCE control signal is passed transparently from the DCE to the DTE, through the Leased Line Backup interface.

In situations where the I level cannot be reliably detected or passed to the DTE, you have the option of forcing I on.

Incoming operation default

Operation	Setting
Auto clear after accept	Enabled

Changing default setting for Auto clear after accept

Disable the **Auto clear after accept** if you do not require an incoming call to be automatically accepted and an alarm cleared when the conditions causing its rejection no longer hold.

Analog Line Exchange Module (ALEM)

The Analog Line Exchange Module (ALEM) is an analog voice-frequency line module that provides an interface to 2-wire analog systems. It contains four individual ports, each of which digitally encodes voice signals onto a 64 kbit/s B-channel.

The ALEM is a loop-in, ring-out interface which emulates some of the features of a 2-wire PSTN connection. It provides:

- battery feed (48 volts)
- on-hook and off-hook detection
- ring signal (with programmable duration and duty cycle)
- dual tone multi-frequency (DTMF) or pulse (decadic) dialling detection.

The ALEM can be used for either switched calls or semipermanent connections. When connected across an ISDN to an ALPM, the two modules can simulate a tie line.

See *Appendix D* for information on operating analog interfaces.

Configuration

Line Number	Enter the line number for the port. Each of the four ports of this module has its own line number. You must enter a line number if the line is to receive incoming calls directly. However, if the line is a member of a lineset or a Virtual Line, no number is required. The line number will be affected when a base number has been configured. See <i>Parameters</i> , page 2-5 for more information.
Subaddress	Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.
Port ID	Enter the Port ID if required. The default setting for this parameter is 0. See <i>Parameters</i> , page 2-5 for more information. It is not mandatory to enter anything in this parameter.
Port Name	Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics provided by the ISDN for incoming and outgoing calls. The default setting is *speech* which requests an analog line, echo suppression and service tones. Other settings may be necessary in special applications. The following table shows the default settings.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Speech

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults are enabled for incoming calls and disabled for outgoing calls. For further information about the default settings, refer to *Parameters* page 2-7.

ISDN Interface — High Layer Compatibility

	Parameter	Default
CCITT (ITU-T) Standards	Incoming	Telephony and Group 2/3 Facsimile
	Outgoing	None
Telstra Standards (where available)	Incoming	Group 1 - 3 Facsimile
	Outgoing	None

Changing High Layer Compatibility defaults

It is not usual to change the High Layer Compatibility settings for an ALEM line. However, you can change them to accept different services from a single line. See *Parameters* page 2-10 for a full description of the available parameters.

ISDN Interface — Call Establishment Modes

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Label	—
Hotline	Enable	Disabled
	Number	—
	XSPC	Disabled
	Subaddress	—
Call Control	Auto Answer	Disabled
Line facilities	Tone suppress	Disabled
Signalling	National Significant Remote Number	—
	End-to-End	Disabled
	OPS (Unint)	Disabled
	Unack	Disabled
PABX Recall	Enable	Disabled

Changing Call Establishment Modes defaults

Semipermanent (where available) Select **Enable** and enter the semipermanent label to set up a semipermanent connection from an ALEM line. You should also disable Hotline calls.

Note *The semipermanent label is supplied by your service provider.*

Hotline Enter the Hotline number and select **Enable** to enable en bloc dialling from an ALEM line. Enter the Subaddress, if required. You should also disable semipermanent connections.

Select **XSPC** to set up a Switched Semipermanent Connection.

Signalling Enter the **National Significant Remote Number** for the remote device.

Select **End-to-End** to enable the out-of-band signalling to be associated with the call, and to prevent the connected customer equipment from affecting the call state of the ISDN call. Proprietary signalling is used to carry information down the D-channel or inband on a B-channel using the Inband Signalling feature. When enabled, **OPS (Unint)** and **UnAck** boxes are available to be selected. This is used for tie line applications such as SPCs or XSPCs. For more information on End-to-end signalling, refer to *Appendix E*.

Call Control You can select **Auto Answer** for Virtual Lines. This can also be used in conjunction with a squawk box.

PABX Recall Select **Enable** to activate the timed break detection settings for PABX recall. See *ISDN Internal Control Defaults* later in this section for further information.

Line facilities When **Tone Suppress** is enabled, the ALEM will cut the dial tone when the first digit is received. This feature is particularly useful during heavy load conditions on the Resource Manager or Control Module when the disconnection of the dial tone tends to be sluggish.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E164
	Hotline	E164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC (where available)

Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID

Do not select **CLIP** (CLI Presentation).

If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

Note *If you request AOC or CLIR for a PRA, but do not subscribe to them, the call attempt may be rejected by the ISDN.*

CLIVE List Assignment

Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format

Select the appropriate ITU-T (CCITT) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

Teleservice Telephony.

ISDN Internal Control defaults

	Parameter	Default
Restriction	None	Selected
Dial Type	Both	Selected
Codec – Relative level (dBr)	line -> CPE	-9 (dBr) 129 (value)
	line <- CPE	+3 (dBr) 187 (value)
Codec – Hybrid Balance	Line length	Short
	Impedance	Complex
	R0	5
	R1	105
	R2	23
Max. PABX Recall time	—	Short
Service Tones	Ring	Selected
	Period	4 on, 2 off, 4 on, 20 off (x 100 ms)
Outgoing TE	Any	Selected

Changing ISDN Internal Control defaults

- Restriction** Select **Disable** to prevent calls from being made to or from this line.
Select **In Only** or **Out Only** to allow calls in the selected direction only.
- Dial Type** Select **Pulse** signalling if a pulse dialling telephone set is used.
Select **DTMF** signalling if a push-button DTMF dialling telephone is used.
Select **Both** to use **Pulse** and **DTMF** signalling as appropriate. The ALEM automatically detects which type is used for a call.
If you select **None** all dialling will be ignored.
- Codec** This controls the signal level of calls and echo cancellation. It should only be changed by your approved maintainer.
- PABX** Select **Long** to set timed break detection of up to 750 ms.
Select **Other** to define the maximum timed break detection period yourself. This should not be greater than 1280 ms.
- Service Tones** Select cadence values to set the cadence of the Ring signal generated by the ALEM.
- Outgoing TE** Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all of the module's lines.

See *Analog line module applications* on page 5-78 for some examples on using the ALEM.

Analog Line Exchange Module-2 (ALEM-2)

The Analog Line Exchange Module -2 (ALEM-2) is a line module for interfacing to two wire analog transmission lines (300 - 3400Hz bandwidth). This module provides four independent configurable two wire analog interface ports that emulate the functions of an analog exchange circuit. Each port may also be configured as a 'direct dial in' (DDI) interface allowing the termination of DDI PSTN lines at the customer premises.

The ALEM-2 provides four two-wire analog interface ports that support the following features:

- Normal/reverse battery feed
- Line loop/unloop detection
- Programmable hookflash detection
- Ring signal generation with programmable cadence
- Pulse or DTMF dial detection
- Back busy signal generation
- Programmable signal protocol
- Programmable termination impedance
- Programmable balance impedance
- Programmable input/output relative levels
- Programmable A-law or μ -law PCM voice encoding.

The ALEM-2 supports the following two-wire signal protocols:

- Loop call/unguarded clear (loop in/ring out)
- Loop call/guarded clear
- Loop call/disconnect clear
- Earth call
- Direct Dial In (DDI).

As part of the system, the ALEM-2 has the capabilities of:

- making on demand ISDN switched calls through Hotline
- setting up XSPCs and SPCs
- providing D-channel end-to-end signalling for XSPC and SPC calls (where available).

Configuration

Line Number Enter the number for the port. Each of the four ports of this module has its own line number.

You must enter a line number if the line is to receive incoming calls (from the backplane of the chassis) directly. However, if the line is a member of a lineset or a Virtual Line, no number is required. The line number will be affected when a base number has been configured.

Note *If a TE is present and a base number has been defined, the line number matched will be the incoming called party number - base number.*

Subaddress Enter the subaddress if required. This parameter is optional. See *Parameters*, page 2-5, for more information.

Port ID Enter the Port ID if required. The default setting for this parameter is 0. This parameter is optional. See *Parameters*, page 2-5, for more information.

Port Name Enter the Port Name if required. This parameter is optional. See *Parameters*, page 2-5, for more information.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics provided by the ISDN for incoming and outgoing calls. The default setting is *speech* which requests an analog line, echo suppression and service tones. Other settings may be necessary in special applications. The following table shows the default settings.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Speech

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. By default Low Layer Compatibility is enabled for incoming calls and disabled for outgoing calls. For more information about the default settings, refer to *Parameters* on page 2-7.

ISDN Interface — High Layer Compatibility

The setting for these parameters should be left as defaults. However, changes can be made to them to accept different services from a single line.

	Parameter	Default
CCITT (ITU-T) Standards	Incoming	Telephony and Group 2/3 Facsimile
	Outgoing	None
Telstra Standards (Australia only)	Incoming	Group 1 - 3 Facsimile
	Outgoing	None

Changing High Layer Compatibility defaults

It is not usual to change the High Layer Compatibility settings for an ALEM-2 line. However, you can change them to accept different services from a single line. See *Parameters* page 2-10 for a full description of the available parameters.

ISDN Interface — Call Establishment Modes

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Label	—
Hotline	Enable	Disabled
	Number	—
	XSPC	Disabled
	Subaddress	—
Call Control	Auto Answer	Disabled
	Release on Progress	Disabled
Line facilities	Tone Suppress	Disabled
Signalling	National Significant Remote Number	—
	End-to-End	Disabled
	Interpreted	Disabled
	Unack	Disabled

Changing Call Establishment Modes defaults

Semipermanent (where available)

Select **Enable** and enter the semipermanent label to set up a semipermanent connection from an ALEM-2 line. You should also disable **Hotline** calls.

Note *The semipermanent label is supplied by your service provider.*

- Hotline** Enter the **Hotline number** and select **Enable** to enable en bloc dialling from an ALEM-2 line. Enter the **Subaddress**, if required. You should also disable semipermanent connections.
- Select **XSPC** to set up a Switched Semipermanent Connection if you need to establish a connection whenever a call fails.
- Signalling** Enter the **National Significant Remote Number** for the remote device.
- Select **End-to-End** to enable the out-of-band signalling to be associated with the call. This prevents the connected customer equipment from affecting the call state of the ISDN calls. Proprietary signalling is used to carry information down the D-channel or inband on a B-channel using the Inband Signalling feature. When enabled, **Interpreted** and **UnAck** boxes are available to be selected. This is used for tie line applications such as SPCs or XSPCs.
- When Interpreted and UnAck are **not** enabled, UDAS Acknowledged End-to-End transfer mechanism is used.
- If Interpreted mode is enabled, IDAS End-to-End signalling is used instead of UDAS. If UnAck mode is enabled, UDAS unacknowledged End-to-End signalling is used instead of IDAS.
- For more information on End-to-End signalling, refer to *Appendix E*.
- Call Control** Select **Auto Answer**, if incoming calls from the backplane are to be automatically answered.
- Enabling **Release on Progress** releases a call even if a Progress Indicator is sent. In this instance, any clearing tones are not heard.
- Line Facilities** When **Tone Suppress** is enabled, the ALEM-2 will cut the dial tone when the first digit is received. This feature is particularly useful during heavy load conditions on the Resource Manager or Control Module when the disconnection of the dial tone tends to be sluggish.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC (where available) Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID Do not select **CLIP** (CLI Presentation).

If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

Note *If you request AOC or CLIR for a PRA, but do not subscribe to them, the call attempt may be rejected by the ISDN.*

CLIVE List Assignment Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format Select the appropriate ITU-T (CCITT) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

Teleservice Telephony.

ISDN Internal Control defaults

	Parameter	Default
Restriction	None	Selected
Dial Type	Both	Selected
Codec – Relative level (dBr)	line -> CPE	-9 (dBr) 129 (value)
	line <- CPE	+3 (dBr) 187 (value)
Codec – Hybrid Balance	Line length	Short
	Impedance	Complex
	R0	5
	R1	105
	R2	23
Signals	See below	Dependent on selected signalling
Outgoing TE	Any	Selected

Changing ISDN Internal Control defaults

Restriction Select **Disable** to prevent calls from being made to or from the line. When selected, the back busy signal type chosen in **Custom Signalling** will be presented to the line. Select **In Only** or **Out Only** to allow calls in the selected directions.

Dial Type Select the dial type.

Codec This controls the signal level of calls and echo cancellation. **It should only be changed by the approved maintainers.**

When the **Set Default Values** button is selected, the codec settings are set to their default values. These values are recommended for most applications. However, if required, the **Relative Level** and the **Hybrid Balance** can be changed.

The **Relative Level** setting from **Digital to Analog** allows you to select or enter the relative level value for audio transmission from the ALEM-2 port to the attached equipment. Likewise, the **Relative Level** setting from **Analog to Digital** allows you to select or enter the relative level value for audio transmission from the attached equipment to the ALEM-2 port.

Refer to the *Installation Manual* for information on relative level settings depending on installation conditions.

The **Hybrid Balance** setting types provide two default sets of echo cancellation register coefficient values. In general, Type 1 values should be selected when the impedance of the line or equipment attached to the ALEM-2 port is the same as the ALEM-2 port sign-on impedance. Type 2 Hybrid Balance may be selected for an alternative line/equipment impedance.

Australian complex

Type 1 sets the echo coefficients to balance the conditions when the line or remote party impedance is an Australian complex impedance. Type 2 balances against a line of 600 Ω impedance.

UK termination

If the port signs on as a UK1 Termination, Type 1 sets the echo coefficients to balance the conditions when the line or remote party impedance is a UK1 impedance. Type 2 balances against a line of UK2 impedance. On the other hand, if the port signs on as a UK2 Termination, Type 1 sets the echo coefficients to balance the conditions when the line or remote party impedance is a UK2 impedance. Type 2 balances against a line of UK1 impedance.

Note *UK2 Termination is not supported from Network Release A onwards.*

Signals This provides a number of 2-wire signalling protocols to choose from. Selecting a signalling option would default all the necessary signalling associated with that option in the **Customise Signalling Dialog**. For most applications, custom configuration is not required.

Signalling Disable ignores all events (loop, unloop, dialling and so on) taking place at the port when selected. Call initiation and termination can only be done via an NMS.

AUSTRALIAN Ring In /Loop Out or UK Loop Calling /Unguarded Clearing

This is the simplest of the signalling protocols and is used for interworking with simple attachments such as telephones. Equipment attached to an ALEM-2 port may generate a call by looping the line. A call is terminated by the equipment unlooping the line. Calls received from a network by the ALEM-2 are indicated to the attached equipment by ring signal.

Loop Calling /Guarded Clearing

This is a more sophisticated signalling protocol, that allows the ALEM-2 to indicate to attached equipment when its port is ready to accept a new call after completion of the last call. This indication is provided by the application of a guarded clear signal (removal of battery feed from both A and B wires) for approximately 400 milliseconds at the end of a call.

Loop Calling /Disconnect Clearing

This is similar to **loop call/guarded clearing** except that the ALEM-2 signals the completion of the last call by application of a disconnect clear signal (removal of battery feed from the A wire only) for approximately one second at the end of a call.

Earth Calling

This requires equipment attached to an ALEM-2 port to initiate a call by momentarily grounding the B wire (for between 50 to 400 ms) before looping the line. The ALEM-2 provides indication of an incoming call from a network by application of ring signal to the attached equipment. Such a call may be answered by the attached equipment looping the line.

Loop In Exchange Line (AUS Indial /UK DDI)

This form of signalling is used when it is required to connect the ALEM-2 to a PSTN and receive incoming calls with called party address information. Calls are received from the analog network in the form of a line loop, along with called party address information in either pulse or DTMF dial form. The ALEM-2 port cannot generate outgoing calls to the network when operating in this mode.

Warning

The ALEM-2 is not permitted to be connected to PSTN lines in the UK.

Custom Signalling

This allows more detailed signalling to be configured. This is only recommended when specialised signals are needed. **It should only be carried out by the approved maintainers.**

Call Request Signals

This selects the type of seize signal to be detected in initiating a call. These options are exclusively selected:

- **Ignore Signal** — ignores all activity at the analog port.
- **Loop Signal** — detects a loop applied to the analog line side as a call initiation.
- **Earth Signal** — detects an earth followed by loop to the analog line side as a call initiation.
- **Loop in Exch Line (Indial/DDI)** — this is the same as **Loop Signal**. It is for PSTN call initiation. No calls will be presented to the analog line side.

Clearing Signals Response

This selects the type of clearing phase to be taken in terminating a call. These options are exclusively selected:

- **Unguarded Return to Idle** — returns the port to an idle state after detecting the analog line side is unlooped.
- **Guarded Clearing** — the ALEM-2 port will respond to a call termination by removing battery feed from the line for 400 ms before returning to the idle line state.
- **Disconnect Clearing** — this is the same as **Guarded Clearing** but removes the battery feed from the line for one second in response to a call termination.

Battery Reversal

- **Reverse Battery on Loop** — this only applies to incoming calls. Detection of a loop on the analog line side while ring current is being applied will cause the ALEM-2 to change the analog line side battery feed from normal to reversed.
- **Reverse Battery on Answer** — this only applies to outgoing calls. On receipt of a CONNECT from the digital network the ALEM-2 will reverse the battery feed to indicate that the call is accepted.

Force Release Signals

This mode forces the analog line to clear by one of the following:

- does nothing when **No Signal** is selected
- open circuits for 200 ms when **Open Circuit** is selected
- reverses the battery polarity for 200 ms when **Battery Reversal** is selected.

Call Presentation Signals

- **No Signal** — enables **Auto Answer** in Call Establishment Dialog. Calls received from a network by the ALEM-2 are answered. No ring signal is indicated to the attached equipment.
- **Ring** — calls received from a network by the ALEM-2 are indicated to the attached equipment by ring signal.
- **Battery Reversal** — calls received from a network by the ALEM-2 are indicated to the attached equipment by reversing the battery polarity feed.

Line Block Signals

- **No Signal** — the ALEM-2 will not change the battery feed condition presented to the analog line side when the ISDN Internal Control '**Restriction**' configuration parameter is set to '**Disable**' or when an IDAS '**block line**' message is received when the ALEM-2 is in End-to-End Signalling Mode.
- **Back Busy (Open Circuit)** — the ALEM-2 will remove battery feed from the analog line side when the ISDN Internal Control '**Restriction**' configuration parameter is set to '**Disable**' or when an IDAS '**block line**' message is received when the ALEM-2 is in End-to-End Signalling Mode.
- **Term Back Busy (Open + 600 ohm)** — the ALEM-2 will remove battery feed but leave a 600Ω termination on the analog line side when the ISDN Internal Control '**Restriction**' configuration parameter is set to '**Disable**' or when an IDAS '**block line**' message is received when the ALEM-2 is in End-to-End Signalling Mode.

Connected (Active) Call

- **PABX Recall** — when enabled, hookflash signal is recognised.
- **PABX Recall Time** — the range of hookflash signals that ALEM-2 will recognise:
 - Short = 40 - 150 ms
 - Long = 450 - 800 ms
 - Others (ms) (40 - 150)
(100 - 350)
(250 - 550)
(450 - 800)
(700 - 1200)
- **Digit Detection and Transfer**
 - **Pulse** — select if only pulse digits are to be transferred out-of-band during an active call.
 - **DTMF** — select if only DTMF digits are to be transferred out-of-band during an active call.
 - **Both** — select this type if BOTH pulse and DTMF digits are to be transferred out-of-band during an active call.
 - **None** — select this type if NO digits are to be transferred out-of-band during an active call.

Ring Cadence

This controls the Ring signal cadence to be generated by the ALEM-2.

The default setting of Customise Signalling for each signalling option is summarised in the following table.

	Signalling schemes					
	Signal disable	RingIn/ LoopOut	Guarded clear	Disconn clear	Earth calling	Indial
Call request signals	Ignore	Loop	Loop	Loop	Earth	Loop
Clear signals response	Unguard	Unguard	Guarded	Disconn clearing	Unguard	Unguard
Battery reversal	All disable	All disable	Reverse on answer	Reverse on Answer	All disable	Reverse on Answer
Force release signals	No signal	No signal	No signal	No signal	No signal	No signal
Call Present signals	No signal	Ring	Ring	Ring	Ring	No signal
Line block signals	No signal	No signal	No signal	No signal	No signal	No signal
Connect (Active) call	None PABX disable	None PABX disable	None PABX disable	None PABX disable	None PABX disable	None PABX disable
Auto Answer	Enable	Disable	Disable	Disable	Disable	Disable

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

See *Analog line module applications* on page 5-78 for some examples of using the ALEM-2.

Analog Line Phone Module (ALPM)

The Analog Line Phone Module (ALPM) is an analog voice-frequency line module with four individual ports which digitally encode voice signals onto a 64 kbit/s B-channel. You can configure each of the four ALPM ports differently. In each case, the appropriate ISDN signalling translation is performed.

The ALPM is a ring-in, loop-out interface. It can be used for either switched indial calls or for semipermanent connections. When connected to an ALEM across an ISDN, the two modules can simulate a tie line.

The ALPM provides:

- on-hook and off-hook generation
- busy and ring back tone detection
- ring signal detection
- tone (DTMF) or pulse (decadic) dialling
- 48 volt battery presence and reversal detection.

See *Appendix D* for more information on operating analog line interfaces.

Configuration

- Line Number** Enter the line number for the port. Each of the four ports of this module has its own line number. See *Parameters*, page 2-5 for more information.
- You must enter a line number if the line is to receive incoming calls directly. However, if the line is a member of a lineset or a Virtual Line, no number is required.
- Subaddress** Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See *Parameters*, page 2-5 for more information.
- Port ID** Enter the Port ID if required. The default setting for this parameter is 0. See *Parameters*, page 2-5 for more information. It is not mandatory to enter anything in this parameter.
- Port Name** Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See *Parameters*, page 2-5 for more information.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics provided by the ISDN for incoming and outgoing calls. The default setting is *speech* which requests an analog line, echo suppression and service tones. Other settings may be necessary in special applications. The following table shows the default settings.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Speech

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults are enabled for incoming calls and disabled for outgoing calls. For further information about the default settings, refer to *Parameters* page 2-7.

ISDN Interface — High Layer Compatibility

	Parameter	Default
CCITT (ITU-T) Standards	Incoming	Telephony and Group 2/3 Facsimile
	Outgoing	None
Telstra Standards (Australia only)	Incoming	Group 1-3 Facsimile
	Outgoing	None

Changing High Layer Compatibility defaults

It is not usual to change the High Layer Compatibility settings for an ALPM line. However they can be changed if you want to access different services from a single line. See *Parameters*, page 2-10 for a full description of the available parameters.

ISDN Interface — Call Establishment Modes defaults

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Label	—
Hotline	Enable	Disabled
	Number	—
	XSPC	Disabled
	Subaddress	—
DDI (Indial)	Enable	Disabled
	Ignore first x digits	0
Active Call Tone Detection	Release on Busy Tone	Disabled
	Use all cadence values	Disabled
Call Control	Connect tone	Disabled
	Auto Answer	Enabled
	Detect PBX tones	Disabled
	Forced release	Disabled
	Variable Pre dial delay	Disabled
Signalling	National Significant Remote Number	—
	End-to-End	Disabled

Changing Call Establishment Modes defaults

Semipermanent (where available)	Select Enable and enter the semipermanent label to set up a semipermanent connection. You should ensure that Hotline is disabled.
Hotline	Enter the Hotline number and select Enable to set up a Hotline call from an ALPM line. Enter the Subaddress if required. You should also disable any semipermanent connections. Select XSPC to set up a Switched Semipermanent Connection.
DDI (Indial)	See <i>Appendix F</i> for details of Indial operation.
Signalling	Enter the National Significant Remote Number of the remote device. Select End-to-End to enable the out-of-band signalling to be associated with the call and to prevent the connected customer equipment from affecting the call state of the ISDN call. This is used for tie line applications, such as SPCs or XSPCs. For more information on End-to-end signalling, refer to <i>Appendix E</i> .
Call Control	Connect Tone is a feature provided for 'squawk box' applications. When enabled, the ALPM should automatically answer incoming calls and then provides the connect tone (two 'beeps'). The delay between looping the line and the application of the two beeps is defaulted to two seconds. This delay can be changed by changing the pre-dial delay. For the connect tone to be provided the following ALPM-2 configuration must also be configured: Dial Type set to DTMF, Dialling disabled, Auto Answer enabled. Select Auto Answer , if incoming calls from the backplane are to be automatically answered. This should be used if the analog line does not provide an answer signal, for example, battery reversal. Enable Detect PBX Tones when the ALPM should respond to service tones provided at the analog line. When Detect PBX Tones is enabled, the ALPM will attempt to detect either ring or busy tone on the line after it has applied the loop and finished dialling any digits. If ring tone is detected, an alerting message is sent. If the busy tone is detected a disconnect message is sent. If no tone is detected the network will time out the call. When Detect PBX Tones is disabled, the ALPM will not attempt to detect any tones. A voice path to the caller is made so any tones or announcements provided at the analog line will be heard. Enabling Forced Release allows the ALPM to forcibly release calls initiated by the CPE connected to the ALPM. When the ALPM fails to setup the call, the CPE can be made to stop ringing into the port by looping the line and immediately proceeding to the disconnecting state. This feature is suited for applications where an ALPM port is connected to a PABX extension circuit. Select Variable Pre Dial delay to initiate a delay in sending dialling digits. This may be necessary if the CPE does not signal its readiness to accept the digits. The delay is set according to the selections made in the ISDN Internal Control/ Dialling. See page 4-160 for more information. Enabling Release on Progress releases a call even if a Progress Indicator is sent. In this instance, any clearing tones will not be heard.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC (where available) Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID Do not select **CLIP** (CLI Presentation).
If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

Note *If you request AOC or CLIR for a PRA, but do not subscribe to them, the call attempt may be rejected by the ISDN.*

CLIVE List Assignment Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format Select the appropriate ITU-T (CCITT) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

Teleservice Telephony.

ISDN Internal Control defaults

The following table shows the ISDN Internal Control defaults. You can change some of the parameters to meet your requirements.

	Parameter	Default
Restriction	None	Selected
Dial Type	DTMF	Selected
Codec – Relative level (dBr)	line -> CPE	-9 (dBr) 169 (value)
	line <- CPE	+3 (dBr) 195 (value)
Codec – Hybrid Balance	Line length	Short
	Impedance	Complex
	R0	69
	R1	12
	R2	55
Dialling	Short (1.2 sec)	Selected
	Minimum (0.4 sec)	Not selected
	Other (1.2 to 11 sec)	Not selected
Service Tones	Ring	Period 4 on, 2 off, 4 on, 20 off (x 100 ms)
	Busy	Period 80, 80, 80, 80
Outgoing TE	Any	Selected

Changing ISDN Internal Control defaults

Restriction Select the required Restriction type.

Note *If you select Disable, the line will not work.*

Codec Accept the defaults for all Codec parameters. This controls the signal level of calls and echo cancellation and should only be changed by your service provider.

Dialling Select the dialling delay after looping the line. The default is **Short** (1.2 secs). Other available settings are **Minimum** (0.4 secs) or any **Other** setting in the range 1.2 to 11 seconds. These parameters are only available if the Call Control **Variable Pre Dial Delay** checkbox is enabled. See page 4-150 for more information.

Service Tones Select **Ring** to display the ring cadence values, then choose **Period** to set the cadence values which are used for the detection of the ring tone. Select **Busy** to display the busy cadence values, then choose **Period** to set the cadence values which are used for the detection of the busy tone.

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

See *Analog line module applications* on page 5-78 for some examples on using the ALPM.



Analog Line Phone Module -2 (ALPM-2)

The ALPM-2 is an analog voice-frequency line module that can connect to two-wire analog systems. The module digitally encodes voice signals into a μ -law or A-law 64 kbit/s signal according to G.711 and emulates features of an analog two-wire telephone.

- The ALPM-2 supports four independently configurable, polarity-independent, loop-out/ring-in two-wire analog interfaces. Each port provides the following features:
- Line loop/unloop generation
- Dial tone detection
- Busy and ring back tone detection
- Tone (DTMF) or pulse (decadic) dialling
- Ring signal detection
- Bi directional polarity battery feed detection
- Link selectable termination impedances
- Adjustable input/output relative levels and hybrid balance
- Programmable A-law or μ -law PCM voice encoding.

Configuration

Line Number Enter the line number for the port. Each of the four ports of this module has its own line number. See *Parameters*, page 2-5 for more information.

You must enter a line number if the line is to receive incoming calls directly. However, if the line is a member of a lineset or a Virtual Line, no number is required.

Note *If a TE is present and a base number has been defined, the line number matched will be the incoming called party number - base number.*

Subaddress Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See *Parameters*, page 2-5 for more information.

Port ID Enter the Port ID if required. The default setting for this parameter is 0. See *Parameters*, page 2-5 for more information. It is not mandatory to enter anything in this parameter.

Port Name Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See *Parameters*, page 2-5 for more information.

The Port Name field can be used to provide security access to the Virtual eXtension feature. Up to eight digits can be entered in the following range: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, #. The stored digits are used to provide password access for a remote Virtual eXtension subscriber. See the *Virtual eXtension* application on page 5-86 for further details.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics provided by the ISDN for incoming and outgoing calls. The default setting is *Speech* which requests an analog line, echo suppression and service tones. Other settings may be necessary in special applications. The following table shows the default settings.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Speech

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults are enabled for incoming calls and disabled for outgoing calls. For further information about the default settings, refer to *Parameters* page 2-7.

ISDN Interface — High Layer Compatibility

	Parameter	Default
CCITT (ITU-T) Standards	Incoming	Telephony and Group 2/3 Facsimile
	Outgoing	None
Telstra Standards (Australia only)	Incoming	Group 1-3 Facsimile
	Outgoing	None

Changing High Layer Compatibility defaults

It is not usual to change the High Layer Compatibility settings for an ALPM-2 line. However they can be changed if you want to access different services from a single line. See *Parameters*, page 2-10 for a full description of the available parameters.

ISDN Interface — Call Establishment Modes defaults

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Label	—
Hotline	Enable	Disabled
	Number	—
	XSPC	Disabled
	Subaddress	—
DDI (Indial) (Dialling)	Enable	Disabled
	Ignore first x digits	0
Active Call Tone Detection	Release on Busy Tone	Disabled
	Use all cadence values	Disabled
Call Control	Auto Answer	Enabled
	Connect tone	Disabled
	Detect PBX tones	Disabled
	Forced Release	Disabled
	Device Progress loop	Disabled
	Release on progress	Disabled
Signalling	National Significant Remote Number	—
	End-to-End	Disabled

Changing Call Establishment Modes defaults

Semipermanent (where available) Select **Enable** and enter the semipermanent label to set up a semipermanent connection. You should ensure that **Hotline** is disabled.

Hotline Enter the Hotline number and select **Enable** to set up a Hotline call from an ALPM-2 line. Enter the Subaddress if required. You should also disable any semipermanent connections.

Select **XSPC** to set up a Switched Semipermanent Connection.

DDI (Indial) (Dialling) This is used to pass part or all of the called party Information Element received with an incoming call to the connected CPE by dialling digits. A port can be part of a lineset with a range of numbers. The **Ignore first... digits** options allows you to specify the number of digits to be ignored before dialling the connected CPE.

Signalling Select **End-to-End** to enable the out-of-band signalling to be associated with the call. This prevents the connected customer equipment from affecting the call state of the ISDN call. This is used for tie line applications, such as SPCs or XSPCs. When enabled, **Interpreted** and **UnAck** boxes are available to be selected. When Interpreted and UnAck are not enabled, UDAS Acknowledged End-to-End transfer mechanism is used. If Interpreted mode is enabled, IDAS End-to-End signalling is used. If UnAck mode is enabled, UDAS unacknowledged End-to-End signalling is used.

Enter the **National Significant Remote Number** of the remote device.

For more information on End-to-end signalling, refer to *Appendix E*.

Call Control **Connect Tone** is a feature provided for 'squawk box' applications. When enabled, the ALPM-2 should automatically answer incoming calls and then provides the connect tone (two 'beeps'). The delay between looping the line and the application of the two beeps is defaulted to two seconds. This delay can be changed by changing the pre-dial delay.

For the connect tone to be provided the following ALPM-2 configuration must also be configured: Dial Type set to **DTMF**, **Dialling** disabled, **Auto Answer** enabled.

Select **Auto Answer**, if incoming calls from the backplane are to be automatically answered. This should be used if the analog line does not provide an answer signal, for example, battery reversal.

Enable **Detect PBX Tones** when the ALPM-2 should respond to service tones provided at the analog line. When Detect PBX Tones is enabled, the ALPM-2 will attempt to detect either ring or busy tone on the line after it has applied the loop and finished dialling any digits. If ring tone is detected, an alerting message is sent. If the busy tone is detected a disconnect message is sent. If no tone is detected the network will time out the call.

When **Detect PBX Tones** is disabled, the ALPM-2 will not attempt to detect any tones. A voice path to the caller is made so any tones or announcements provided at the analog line will be heard.

Enabling **Forced Release** allows the ALPM-2 to forcibly release calls initiated by the CPE connected to the ALPM-2. When the ALPM-2 fails to setup the call, the CPE can be made to stop ringing into the port by looping the line and immediately proceeding to the disconnecting state. This feature is suited for applications where an ALPM-2 port is connected to a PABX extension circuit.

Note *The Bearer Capability of the ALPM-2 port must be set to 3.1kHz Audio for it to see the disconnect from the device.*

Enabling the **Device Progress Loop** feature causes the CPE to hear network tones/announcements and so on, that occur when a call that the CPE originates is unsuccessful. Normally, unsuccessful calls originating from the CPE will not be looped.

Enabling **Release on Progress** releases a call even if a Progress Indicator is sent. In this instance, any clearing tones will not be heard.

Active Call Tone Detection **Release on Busy Tone** is only used in tie line applications. It specifies the action to be taken when a PABX line is busy. If the CPE goes busy when a call is active, the call is released.

Up to four cadence values are used to define a service tone. When **Use all cadence Values** is selected, all cadence values are used in detection. If not, only the first two cadence values are used.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC (where available) Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID Do not select **CLIP** (CLI Presentation).
If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

Note *If you request AOC or CLIR for a PRA, but do not subscribe to them, the call attempt may be rejected by the ISDN.*

CLIVE List Assignment Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format Select the appropriate ITU-T (CCITT) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

Teleservice Telephony.

ISDN Interface — ISDN Internal Control defaults

The following table shows the ISDN Internal Control defaults. You can change some of the parameters to meet your requirements.

	Parameter	Default
Restriction	None	Selected
Dial Type	DTMF	Selected
Codec – Relative level (dBr)	line -> CPE	-9 (dBr) 169 (value)
	line <- CPE	+3 (dBr) 195 (value)
Codec – Hybrid Balance	Line length	Short
	Impedance	Complex
	R0	69
	R1	12
Service tones	Ring	Period 4 on, 2 off, 4 on, 20 off (x 100 ms)
	Busy	Period 80, 80, 80, 80 ms
PABX	See below	Dependent on selection
Signals	See below	Dependent on selection
Outgoing TE	Any	Selected

Changing ISDN Internal Control defaults

Restriction Select the required Restriction type.

Note *If you select Disable, the line will not work.*

Dial type Select the dial type.

Codec This controls the signal level of calls and echo cancellation and should only be changed by approved maintainers.

The **Set Default Values** button defaults the codec setting to its factory settings. These values are recommended for most applications. If required, the settings can be changed.

Relative Level Settings

Relative Level setting from **Digital to Analog** allows the user to select or enter the relative level value for audio transmission from the ALPM-2 port to the attached analog equipment. The **Relative Level** setting from **Analog to Digital** allows the user to select or enter the relative level value for audio transmission from the attached analogue equipment to the ALPM-2 port.

Refer to the *Installation Manual* for information on relative level settings depending on installation conditions.

Hybrid Balance Settings

Hybrid Balance settings provide balancing of the termination impedance for two line impedances: **Type 1** and **Type 2**. An approved maintainer can also modify values in the hybrid balance registers.

Australian termination

If the port signs on as a 600 Ω Termination, **Type 1** sets the echo coefficients to balance the conditions when the line or remote party impedance is a 600 Ω impedance. **Type 2** balances against a line of Australian Complex impedance.

If the port signs on as an Australian Complex Termination, **Type 1** sets the echo coefficients to balance the conditions when the line or remote party impedance is an Australian Complex impedance. **Type 2** balances against a line of 600 Ω impedance.

UK termination

If the port signs on as a UK1 Termination, **Type 1** sets the echo coefficients to balance the conditions when the line or remote party impedance is an UK1 impedance. **Type 2** balances against a line of UK2 impedance.

If the port signs on as a UK2 Termination, **Type 1** sets the echo coefficients to balance the conditions when the line or remote party impedance is an UK2 impedance. **Type 2** balances against a line of UK1 impedance.

Note *UK1 termination impedance is 370 Ω + 620 Ω // 310nF. UK2 termination impedance is 300 Ω + 1K Ω // 220nF.*

PABX This allows you to select the action on receiving a timed break request and to set up the Virtual eXtension feature.

You can configure the Timed Break period by selecting **Use break time of n (x10)ms**. This will apply a looped disconnect of the configured duration to an analog line in response to a user-to-user timed break (hookflash) request received from the far end of a connection. Alternatively, you can accept the timed break duration coded into the user-to-user message by leaving the default **Use break time received in message** selected.

We recommend that you configure the timed break to suit your PABX, for example in Australia this would typically be 100 ms, in Europe 90 ms, and the USA 600 ms.

During a call, a timed break of the configured duration can be applied to an analog line in response to the ALPM-2 receiving an inband sequence of DTMF digits that match a user-configured string. This feature allows remote users of the Virtual eXtension to dial digits to request a timed break. The DTMF digits should be entered in the **DTMF Digits to interpret as a Timed Break request**. Up to four digits can be entered and they should be in the following range: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, #.

A disconnect can also be requested in a similar manner. The DTMF digits should be entered in the **DTMF Digits to interpret as a Disconnect request** field. Up to four digits can be entered and they should be in the following range: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, #.

See the *Virtual eXtension* application on page 5-86 for further information.

Signals This button configures the two-wire signalling protocol. You can choose one of the following five standard signalling protocols or select **Custom**:

- Loop In Exchange Line (Indial/DDI)
- Ring In/Loop Out
- Loop Calling/Guarded Clear
- Loop Calling/Disconnect Clear
- Loop Calling/Unguarded Clear.

Selecting a standard signalling protocol defaults all the necessary signalling parameters (as seen in the Custom dialog) as well as Call Establishment Modes' **Call Control** and **Active Call Tone Detection** options, and the **Dial Type** to DTMF.

Where the battery polarity is used in a signalling protocol, the ALPM-2 must have the idle line state referenced to the normal battery polarity. When you know that the line is idle, you can press the **Auto Reference** button. This will set the idle reference battery polarity in the ALPM-2. This need only happen when a new line is connected to the ALPM-2.

If **Custom** is selected, the following items can be configured:

Dialling

- **Wait before dialling** controls the amount of time that the ALPM-2 waits after looping a line before it starts dialling digits. Wait for dial tone can also be selected where dialling will only commence after dial tone is detected.
- The **Interdigit Pause** between digits while dialling can be adjusted. The default DTMF pause is 100ms and the default Decadic pause is 800ms.

Seize/Hold

Seize Time defines the period for when a line is first looped (seized) That a high current loop is applied. After the seize period, the normal current loop (hold) is applied and the line is ready for voice path connection. A Seize Time of 'none' can be selected where the line is only looped with the hold current.

Disconnect Calls which originated from device

- The **Device Disconnects** selection defines the way in which a call originating from the device will be unlooped from the CPE when the device disconnects.
- The **Port Disconnects** selection defines the way in which a call originating from the device will be unlooped from the CPE when the CPE disconnects.

Disconnect Calls which originated from CPE Connected to ALPM Port

- The **Device Disconnects** selection defines the way in which a call originating from the CPE will be unlooped from the CPE when the device disconnects.
- The **Port Disconnects** selection defines the way in which a call originating from the CPE will be unlooped from the CPE when the CPE disconnects.

Line Fail Monitoring

- The **Monitoring Interval** specifies the interval for which a 'Failed' or 'Ok' condition must be met before the condition is detected. You can also turn off this feature here.
- **Monitoring for Battery Removed** is mandatory when monitoring is active. Monitoring for **Battery Normal In Low Current State** and **Battery Reversed In Unloop State** can also be monitored when this is enabled.

Call Blocking

This feature specifies the duration before other calls can be made by the CPE after a disconnect sequence has been completed. Incoming calls from the device are still accepted during this period.

Invert Line Polarity Sense

When this box is checked, the normal idle line battery polarity reference in the ALPM-2 is inverted. The **Auto Reference** button may also be used.

Service Tones Select **Ring** to display the ring cadence values, then choose **Period** to set the cadence values which are used for the detection of the ring tone.

Select **Busy** to display the busy cadence values, then choose **Period** to set the cadence values which are used for the detection of the busy tone.

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

See *Analog Line module applications* on page 5-78 for some examples on using the ALPM-2.



E&M Line Module (EMM and EMM-2)

This section applies to both versions of the EMM module. Configuration is the same for both except where indicated.

The E&M Line Module is an analog voice frequency line module for interfacing to four-wire, single E&M analog systems (that is, one pair for Tx, one pair for Rx, one wire for E and one wire for M). It contains four individual interfaces which digitally encode voice signals onto a 64 kbit/s B-channel.

The E&M Line Module is used mainly for tie line simulation, but it can also be used for switched ISDN calls.

The EMM(-1) supports only Type 5 (C2) signalling in both earth on idle and earth off idle modes

The EMM-2 also supports Type 5 (C2) but supports the following additional signalling schemes:

- Type 1
- Type 2
- Type 3.

The module uses decadic dialling on E&M leads and DTMF dialling in-band on the B-channel. It can also operate in transparent end-to-end signalling mode. This is where E&M line status is transferred end-to-end to another E&M, regardless of the protocol used.

Earth-on-idle or Earth-off-idle is supported as a hardware link configuration option.

See *Appendix D* for more information on operating analog interfaces.

Configuration

Line Number	Enter the line number (allocated by your service provider) for the port. Each of the four ports on this module has its own line number. See <i>Parameters</i> , page 2-5 for more information. For outgoing calls, or if the line is a member of a Virtual Line or lineset, this parameter can be left blank.
Subaddress	Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See <i>Parameters</i> , page 2-5 for more information.
Port ID	Enter the Port ID if required. The default setting for this parameter is 0. See <i>Parameters</i> , page 2-5 for more information. It is not mandatory to enter anything in this parameter.
Port Name	Enter a Port Name if required. It is not mandatory to enter anything in this parameter.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. The following table shows the default settings. Accept the defaults for all the parameters.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Speech
Octet 5	Layer 1	μ-law when country code =1
		A-law for all other country codes

Note *Octet 5 is mandatory if Octet 3 Transfer is 'Speech' or '3.1 kHz Audio'.*

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. The defaults are enabled for incoming calls and disabled for outgoing calls. For further information about the default settings, refer to *Parameters*, page 2-7.

ISDN Interface — High Layer Compatibility

The default settings for all the parameters are shown in the following table. Accept the defaults for all the parameters. For more information see *Parameters* page 2-10.

	Parameter	Default
CCITT (ITU-T) Standards	Incoming	Telephony/Group 2/3 Facsimile
	Outgoing	None
Telecom Standards (Australia)	Incoming	Group 1-3 Facsimile
	Outgoing	None

Call Establishment Modes

The following table shows the Call Establishment Modes defaults. You can change the settings in any of the parameters to correspond with the other end of the connection.

	Parameter	Default
Semi permanent (Where available)	Enable	Disabled
	Label	—
Subrate Virtual Line member or manually connected (where applicable)	Yes	Disabled
	Label	—
Hotline	Enable	Disabled
	Number	—
	XSPC	Disabled
	Subaddress	—
DDI (Indial)	Enable	Disabled
	Ignore first	0 digits
Back Busy	Enable	Enabled
Signalling	National Significant Remote Number	—
	End-to-End	Enabled
	Interpreted	Disabled
	UnAck	Disabled
Call Control	Auto Answer	Enabled
	End-to-end Answer Transfer	Disabled
	DTMF indial	Disabled
	Release on Progress	Disabled
Call to PABX (Applicable to EMM-2 only)	Wait for dial tone	Disabled
	Wink start	Disabled
	Delay dial	Disabled
Call from PABX (Applicable to EMM-2 only)	Wink start	Disabled
	Delay dial	Disabled

Changing Call Establishment Modes defaults

Semipermanent (Where available)

Enter the semipermanent number and select **Enable**. You should also disable Hotline calls.

Subrate Virtual Line member or manually connected (where available)

Select **Yes** if the line is a part of a Subrate Virtual Line or is manually connected. Enter the appropriate label.

- Hotline** Enter the Hotline number and the subaddress number, if required. Select **Enable**.
Select **XSPC** to enable a Switched Semipermanent Call. This is only available with a Hotline setup.
- DDI (Indial)** Select **Enable** and enter the number of digits to be ignored.
For example, if the call is for extension 23 of the telephone number 8096923, the number of digits to ignore is five. The remaining digits are decadically dialled into the terminal equipment.
- Back busy** Select **Enable** to present a busy signal to the PABX when the tieline is not active.
- Signalling** Enter the **National Significant Remote Number** for the remote device.
Where available, select **End-to-End** to operate in a leased line environment, such as a semipermanent connection. This establishes a signalling channel to the other end.
If you select **End-to-End** signalling, you can select **Interpreted**. Interpreted signalling applies the C2 protocol to the E&M lines and transfers the signalling via IDAS. The remote end can be an E1M, ALEM-2, ALPM-2 or EMM. If **Interpreted** is not selected, only an EMM can be at the other end and the E state is reflected on the M LED at the remote end.
The uninterpreted signal transfer is acknowledged by default. To save packets the acknowledgment may be disabled by selecting **UnAck**.
For more information on End-to-end signalling, refer to *Appendix E*.
- Call control** Select **End-to-end answer transfer** to provide an end-to-end answering mechanism to support some inter-PABX operations. It provides for the answering of an ISDN switched call followed by the end-to-end transfer of an Answer signal from the called terminal equipment back to the calling line. This cannot be used when End-to-end signalling is enabled.
Enable **DTMF indial** to select DTMF dialling instead of pulse dialling.
Enabling **Release on Progress** releases a call even if a Progress Indicator is sent. In this instance, any clearing tones are not heard.
- Call to PABX (EMM-2 only)** The default signalling protocol used when the EMM-2 seizes the PABX is 'immediate dial'. In this state, no dialling control signal is used. The EMM-2 waits for 1.2 seconds after sending a seize signal before dialling the first digit.
You can apply alternative protocols by selecting the appropriate check box.
If you select **Wink Start**, the EMM-2 sends an 'active' signal with a defined duration (wink) after it has received a seize signal. This indicates that it is prepared to receive dialling information.
If you select **Delay Dial**, the EMM-2 applies an 'active' signal until it is ready to receive dialling information. It then reapplies an 'idle' signal to indicate that it is prepared to receive dialling information.
You can select **Wait for Dial Tone** to work in conjunction with the three protocols explained above, so that as well as those protocol conditions, the DDI dialling to the PABX is delayed until the EMM-2 recognises dial tone from the PABX.

Note *Glare Resolution is not supported.*

**Call from PABX
(EMM-2 only)**

The default signalling protocol used when the PABX seizes the EMM-2 is 'immediate dial'. In this state, no dialling control signal is used. The PABX waits for a specified period after sending a seize signal before dialling the first digit.

You can apply alternative protocols by selecting the appropriate check box.

If you select **Wink Start**, the PABX sends an 'active' signal with a defined duration (wink) after it has received a seize signal. This indicates that it is prepared to receive dialling information.

If you select **Delay Dial**, the PABX applies an 'active' signal until it is ready to receive dialling information. The terminating switching system then reapplies an 'idle' signal to indicate that it is prepared to receive dialling information.

Note *Glare Resolution is not supported.*

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (Where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults**AOC
(Where available)**

Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via Call Log within PM. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID

Do not select **CLIP** (CLI Presentation).

If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

Note *If you request AOC or CLIR for a PRI, but do not subscribe to them, the call attempt may be rejected by the ISDN.*

-
- CLIVE List Assignment** Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.
- Numbering format** Select the appropriate ITU-T (CCITT) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

- Teleservice (Where available)** None.

Changing ISDN Internal Control defaults

- Restriction** Select **Disable** from the list to prevent calls from being made to or from this line. Select **In Only** or **Out Only** to allow calls in the selected direction only.
- Codec** This parameter controls the signal level of calls and echo cancellation. Accept the defaults. **It should only be changed by your approved maintainer.**
- Outgoing TE** Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

COMBO Module

The COMBO Module provides a means of connecting both analog and digital devices to a single card, and multiplexing them across a digital link to other devices. The voice compression capability is used to compress the analog voice signals into digital signals. These signals are then multiplexed with any data onto a common channel, thus providing tie lines between members of the same product family.

The data interfaces on the COMBO typically provide low data rate terminal links to computers and inter-computer links.

Each Analog Line Interface Module (ALIM) port can be independently configured to provide any of the following signalling schemes:

- Austel's Ring In/Loop Out (ALIM-EXCH) — connects as exchange equipment.
- Austel's extension line (ALIM-EXCH).

These signalling schemes can be used to provide either dialled-up voice circuits or permanent voice circuits.

The COMBO can be used across Subrate and TimeFrame Virtual Lines between nodes in a network.

The COMBO supports a single Voice Compression submodule on the main board. The COMBO supports the Dual Channel G.723.1 Compressor (VC G.7231-2) and the High Density Voice Compression Module (HD-VCM).

These submodules use the MP-MLQ (Multi-Pulse Maximum Likelihood Quantization) speech compression technique. They provide a G3 fax relay capability in addition to enhanced voice compression at the following rates:

- 8kbit/s MP-MLQ
- 16kbit/s MP-MLQ.

The HD-VCM also provides a Modem Relay capability.

The voice compression modules compress 64 kbit/s A-law or μ -law voice signals to either 8 or 16 kbit/s for transmission over a common channel. The receiving end expands the voice sub-channels back to 64 kbit/s. A combination of voice and data sub-channels can be transmitted over a common channel via the Subrate data switch on the COMBO.

Note *The voice compression algorithm of a COMBO with VC-G7231-2 is compatible with an MFCM with MLQ-CMP submodules, or an LDCM with VF-CMP submodules. The voice compression algorithm of a COMBO with HD-VCM is only compatible with an MFCM with HD-VCM submodules.*

Analog line interface configuration

Line Number Select and enter the number for the desired port. Each of the ports of this module has its own line number.

You must enter a line number if the line is to receive incoming calls (from the backplane) directly. However, if the line is a member of a lineset or a Virtual Line, no number is required. The line number will be affected when a base number has been configured.

Note *If a TE is present and a base number has been defined, the line number matched will be the incoming called party number - base number.*

Subaddress Enter the subaddress if required. This parameter is optional. See *Parameters*, page 2-5, for more information.

Port ID Enter the Port ID if required. The default setting for this parameter is 0. This parameter is optional. See *Parameters*, page 2-5, for more information.

Port Name Enter the Port Name if required. This parameter is optional. See *Parameters*, page 2-5, for more information.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics provided by the ISDN for incoming and outgoing calls. The default setting is *speech* which requests an analog line, echo suppression and service tones. Other settings may be necessary in special applications. The following table shows the default settings.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Speech

ISDN Interface — Low Layer Compatibility

The settings for these parameters are determined by the capability of the equipment used. By default Low Layer Compatibility is enabled for incoming calls and disabled for outgoing calls. For more information about the default settings, refer to *Parameters* on page 2-7.

ISDN Interface — High Layer Compatibility

The setting for these parameters should be left as defaults. However, changes can be made to them to accept different services from a single line.

	Parameter	Default
CCITT (ITU-T) Standards	Incoming	Telephony and Group 2/3 Facsimile
	Outgoing	None
Telstra Standards (Australia only)	Incoming	Group 1 - 3 Facsimile
	Outgoing	None

Changing High Layer Compatibility defaults

It is not usual to change the High Layer Compatibility settings for an ALIM line. However, you can change them to accept different services from a single line. See *Parameters* page 2-10 for a full description of the available parameters.

ISDN Interface — Call Establishment Modes

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Label	—
Hotline	Enable	Disabled
	Number	—
	XSPC	Disabled
	Subaddress	—
Call Control	Auto Answer	Disabled
	Release on Progress	Disabled
Line facilities	Tone Suppress	Disabled
Signalling	National Significant Remote Number	—
	End-to-End	Disabled
	Interpreted	Disabled
	Unack	Disabled

Changing Call Establishment Modes defaults

Semipermanent (where available)

Select **Enable** and enter the semipermanent label to set up a semipermanent connection from an ALIM line. You should also disable **Hotline** calls.

Note *The semipermanent label is supplied by your service provider.*

- Hotline** Enter the **Hotline number** and select **Enable** to enable en bloc dialling from an ALIM line. Enter the **Subaddress**, if required. You should also disable semipermanent connections.
- Select **XSPC** to set up a Switched Semipermanent Connection if you need to establish a connection whenever a call fails.
- Signalling** Enter the **National Significant Remote Number** for the remote device.
- Select **End-to-End** to enable the out-of-band signalling to be associated with the call. This prevents the connected customer equipment from affecting the call state of the ISDN calls. Proprietary signalling is used to carry information down the D-channel or inband on a B-channel using the Inband Signalling feature. When enabled, **Interpreted** and **UnAck** boxes are available to be selected. This is used for tie line applications such as SPCs or XSPCs.
- When Interpreted and UnAck are **not** enabled, UDAS Acknowledged End-to-End transfer mechanism is used.
- If Interpreted mode is enabled, IDAS End-to-End signalling is used instead of UDAS. If UnAck mode is enabled, UDAS unacknowledged End-to-End signalling is used instead of IDAS.
- For more information on End-to-End signalling, refer to *Appendix E*.
- Call Control** Select **Auto Answer**, if incoming calls from the backplane are to be automatically answered.
- Enabling **Release on Progress** releases a call even if a Progress Indicator is sent. In this instance, any clearing tones are not heard.
- Line Facilities** When **Tone Suppress** is enabled, the ALIM will cut the dial tone when the first digit is received. This feature is particularly useful during heavy load conditions on the Resource Manager when the disconnection of the dial tone tends to be sluggish.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC (where available) Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID Do not select **CLIP** (CLI Presentation).

If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

Note *If you request AOC or CLIR for a PRA, but do not subscribe to them, the call attempt may be rejected by the ISDN.*

CLIVE List Assignment Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format Select the appropriate ITU-T (CCITT) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults

Teleservice Telephony.

ISDN Internal Control defaults

	Parameter	Default
Restriction	None	Selected
Dial Type	Both	Selected
Codec – Relative level (dBr)	line -> CPE	-9 (dBr) 129 (value)
	line <- CPE	+3 (dBr) 187 (value)
Codec – Hybrid Balance	Line length	Short
	Impedance	Complex
	R0	5
	R1	105
	R2	23
Signals	See below	Dependent on selected signalling
Outgoing TE	Any	Selected

Changing ISDN Internal Control defaults

Restriction Select **Disable** to prevent calls from being made to or from the line. When selected, the back busy signal type chosen in **Custom Signalling** will be presented to the line. Select **In Only** or **Out Only** to allow calls in the selected directions.

Dial Type Select the dial type.

Codec This controls the signal level of calls and echo cancellation. **It should only be changed by the approved maintainers.**

When the **Set Default Values** button is selected, the codec settings are set to their default values. These values are recommended for most applications. However, if required, the **Relative Level** and the **Hybrid Balance** can be changed.

The **Relative Level** setting from **Digital to Analog** allows you to select or enter the relative level value for audio transmission from the ALIM port to the attached equipment. Likewise, the **Relative Level** setting from **Analog to Digital** allows you to select or enter the relative level value for audio transmission from the attached equipment to the ALIM port.

Refer to the *Installation Manual* for information on relative level settings depending on installation conditions.

The **Hybrid Balance** setting types provide two default sets of echo cancellation register coefficient values. In general, Type 1 values should be selected when the impedance of the line or equipment attached to the ALIM port is the same as the ALIM port sign-on impedance. Type 2 Hybrid Balance may be selected for an alternative line/equipment impedance.

Australian termination

Type 1 sets the echo coefficients to balance the conditions when the line or remote party impedance is an Australian complex impedance. Type 2 balances against a line of 600 Ω impedance.

UK termination

If the port signs on as a UK1 Termination, Type 1 sets the echo coefficients to balance the conditions when the line or remote party impedance is a UK1 impedance. Type 2 balances against a line of UK2 impedance. On the other hand, if the port signs on as a UK2 Termination, Type 1 sets the echo coefficients to balance the conditions when the line or remote party impedance is a UK2 impedance. Type 2 balances against a line of UK1 impedance.

Signals This provides a number of 2-wire signalling protocols to choose from. Selecting a signalling option would default all the necessary signalling associated with that option in the **Customise Signalling Dialog**. For most applications, custom configuration is not required.

Signalling Disable ignores all events (loop, unloop, dialling and so on) taking place at the port when selected. Call initiation and termination can only be done via an NMS.

AUSTRALIAN Ring In /Loop Out

This is the simplest of the signalling protocols and is used for interworking with simple attachments such as telephones. Equipment attached to a COMBO ALIM port may generate a call by looping the line. A call is terminated by the equipment unlooping the line. Calls received from a network by the ALIM are indicated to the attached equipment by ring signal.

Loop Calling /Guarded Clearing

This is a more sophisticated signalling protocol, that allows the ALIM to indicate to attached equipment when its port is ready to accept a new call after completion of the last call. This indication is provided by the application of a guarded clear signal (removal of battery feed from both A and B wires) for approximately 400 milliseconds at the end of a call.

Loop Calling /Disconnect Clearing

This is similar to **loop call/guarded clearing** except that the ALIM signals the completion of the last call by application of a disconnect clear signal (removal of battery feed from the A wire only) for approximately one second at the end of a call.

Earth Calling

This requires equipment attached to a COMBO ALIM port to initiate a call by momentarily grounding the B wire (for between 50 to 400 ms) before looping the line. The COMBO provides indication of an incoming call from a network by application of ring signal to the attached equipment. Such a call may be answered by the attached equipment looping the line.

Custom Signalling

This allows more detailed signalling to be configured. This is only recommended when specialised signals are needed. **It should only be carried out by the approved maintainers.**

Call Request Signals

This selects the type of seize signal to be detected in initiating a call. These options are exclusively selected:

- **Ignore Signal** — ignores all activity at the analog port.
- **Loop Signal** — detects a loop applied to the analog line side as a call initiation.
- **Earth Signal** — detects an earth followed by loop to the analog line side as a call initiation.

Clearing Signals Response

This selects the type of clearing phase to be taken in terminating a call. These options are exclusively selected:

- **Unguarded Return to Idle** — returns the port to an idle state after detecting the analog line side is unlooped.
- **Guarded Clearing** — the ALIM port will respond to a call termination by removing battery feed from the line for 400 ms before returning to the idle line state.
- **Disconnect Clearing** — this is the same as **Guarded Clearing** but removes the battery feed from the line for one second in response to a call termination.

Battery Reversal

- **Reverse Battery on Loop** — this only applies to incoming calls. Detection of a loop on the analog line side while ring current is being applied will cause the ALIM to change the analog line side battery feed from normal to reversed.
- **Reverse Battery on Answer** — this only applies to outgoing calls. On receipt of a CONNECT from the digital network the ALIM will reverse the battery feed to indicate that the call is accepted.

Force Release Signals

This mode forces the analog line to clear by one of the following:

- does nothing when **No Signal** is selected
- open circuits for 200 ms when **Open Circuit** is selected
- reverses the battery polarity for 200 ms when **Battery Reversal** is selected.

Call Presentation Signals

- **No Signal** — enables **Auto Answer** in Call Establishment Dialog. Calls received from a network by the COMBO are answered. No ring signal is indicated to the attached equipment.
- **Ring** — calls received from a network by the COMBO are indicated to the attached equipment by ring signal.
- **Battery Reversal** — calls received from a network by the COMBO are indicated to the attached equipment by reversing the battery polarity feed.

Line Block Signals

- **No Signal** — the COMBO ALIM will not change the battery feed condition presented to the analog line side when the ISDN Internal Control '**Restriction**' configuration parameter is set to '**Disable**' or when an IDAS '**block line**' message is received when the COMBO is in End-to-End Signalling Mode.
- **Back Busy (Open Circuit)** — the COMBO ALIM will remove battery feed from the analog line side when the ISDN Internal Control '**Restriction**' configuration parameter is set to '**Disable**' or when an IDAS '**block line**' message is received when the COMBO is in End-to-End Signalling Mode.
- **Term Back Busy (Open + 600 ohm)** — the COMBO ALIM will remove battery feed but leave a 600Ω termination on the analog line side when the ISDN Internal Control '**Restriction**' configuration parameter is set to '**Disable**' or when an IDAS '**block line**' message is received when the COMBO is in End-to-End Signalling Mode.

Connected (Active) Call

- **PABX Recall** — when enabled, hookflash signal is recognised.
- **PABX Recall Time** — the range of hookflash signals that ALEM-2 will recognise:
 - Short = 40 - 150 ms
 - Long = 450 - 800 ms
 - Others (ms) (40 - 150)
(100 - 350)
(250 - 550)
(450 - 800)
(700 - 1200)
- **Digit Detection and Transfer**
 - **Pulse** — select if only pulse digits are to be transferred out-of-band during an active call.
 - **DTMF** — select if only DTMF digits are to be transferred out-of-band during an active call.
 - **Both** — select this type if BOTH pulse and DTMF digits are to be transferred out-of-band during an active call.
 - **None** — select this type if NO digits are to be transferred out-of-band during an active call.

Ring Cadence

This controls the Ring signal cadence to be generated by the COMBO ALIM.

The default setting of Customise Signalling for each signalling option is summarised in the following table.

	Signalling schemes				
	Signal disable	RingIn/ LoopOut	Guarded clear	Disconn clear	Earth calling
Call request signals	Ignore	Loop	Loop	Loop	Earth
Clear signals response	Unguard	Unguard	Guarded	Disconn clearing	Unguard
Battery reversal	All disable	All disable	Reverse on answer	Reverse on Answer	All disable
Force release signals	No signal	No signal	No signal	No signal	No signal
Call Present signals	No signal	Ring	Ring	Ring	Ring
Line block signals	No signal	No signal	No signal	No signal	No signal
Connect (Active) call	None PABX disable	None PABX disable	None PABX disable	None PABX disable	None PABX disable
Auto Answer	Enable	Disable	Disable	Disable	Disable

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting affects all the module's lines.

See *Analog line module applications* on page 5-78 for some examples of using the COMBO ALIM.

Digital interface configuration

Port Number Enter the Port number for this line. See *Parameters*, page 2-5 for more information.

Subaddress Enter the Subaddress if required. It is not mandatory to enter anything in this parameter. See *Parameters*, page 2-5 for more information.

Port ID Enter the Port ID if required. The default setting for this parameter is 0. See *Parameters*, page 2-5 for more information. It is not mandatory to enter anything in this parameter.

Port Name Enter the Port Name if required. It is not mandatory to enter anything in this parameter. See *Parameters*, page 2-5 and *TimeFrame* on page 3-56 for more information.

Port Setup

Interface Mode Configures the COMBO port to act as DTE or DCE. The default is DCE. In DTE mode, the COMBO is capable of synchronising its clock with the DCE-supplied clock, to supply the Reference Master Clock.

Rate Adaption Allows the Rate Adaption method to be selected. See page 2-8 for further information. The type of rate adaption selected will affect the data rates that are available on the port.

- Port Speed** This allows the selection of the following speeds:
- **CCITT Rate Adapted** – 600, 1200, 2400, 4800, 9600, 19200, 38400 48000, 56000, 64000 (synchronous only)
 - **Non-CCITT Rate Adapted** – n x 8kbit/s up to 256 kbit/s.

Options

- Outgoing TE** Select the required TE for the port. If you select **Preferred** or **Exclusive**, you must then select the TE from the list of available TEs.
- Don't use as clock reference** Enable this if you do not want the COMBO to supply the reference clock. This option is only available for ports acting as DTEs.

Call Establishment Modes

Click on this to open the Call Establishment Modes dialog.

ISDN Interface — Call Establishment Modes

This information is used to control how the COMBO establishes outgoing calls and how it accepts incoming calls. See the sections on the *SDLM V.24*, *SDLM X.21* and *SDLM V.35* for detailed information on Call Establishment Modes for each type of DTE.

Before an outgoing call can be made using Hotline mode, the Hotline number and the Hotline subaddress must be entered if required. Hotline mode must then be enabled.

For lead call control, the DTR Call option must be enabled for V.24 and V.35 interfaces. The C Call option must be enabled for X.21 interfaces. All other options must be disabled with the exception of Auto Answer which may be either enabled or disabled.

A DTE with synchronous X.21 interface can set up, answer or clear a switched call by exchanging signals with the COMBO port. When using this X.21 call control, the X.21 option must be enabled, CCITT rate adaption must be selected, and all other Call Control options must be disabled. This allows the attached DTE to communicate with the COMBO according to the X.21 call establishment protocol recommended by the ITU-T (CCITT).

	Parameter	Default
Semipermanent (where available)	Enable	Disabled
	Backup	Disabled
	Label	—
Member of Subrate Virtual Line or manually connected	Yes	Disabled
Hotline	Enable	Disabled
	Number	—
	Don't increment Hotline No.	Disabled
	XSPC	Disabled
	Subaddress	—
Active Call Facilities	Don't release call if V.110 frame loss detected	Disabled
Call Control	Auto Answer	Disabled
	DTR or C Call	Enabled
	V.25 bis	Disabled
	X.21	Disabled
	DTR/C Blocking	Disabled
Control Leads	Local Only	Enabled
	Transferred Inband	Disabled

Changing Call Establishment Modes defaults

See the sections on the *SDLM V.24*, *SDLM X.21* and *SDLM V.35* for detailed information on changing call establishment modes for each type of interface.

Semipermanent (where available)

Enter the semipermanent number in the Label parameter and select **Enable**.

Note *Semipermanent circuits are not supported on COMBO Version 5.0 and onwards.*

Hotline

Enter the Hotline number and the subaddress number, if required, and select **Enable**. If a subaddress is not used, the Hotline number for subsequent (64kbit/s) lines is automatically incremented, based on the Hotline number configured. The automatic Hotline number increment can be disabled by checking the **Don't increment Hotline No.** checkbox.

Select **XSPC** to enable a Switched Semipermanent Connection.

Call Control If you are using CCITT rate adaption, select **V.25 bis** to use Cisco V.25 bis (software release version 10) protocol for call control between the DTE and the COMBO. This is available over X.21, V.35 and V.24 synchronous line interfaces. Selecting V.25 bis allows DTEs to make outgoing calls and accept incoming calls through the COMBO.

Advanced

Click on the Advanced button to open a Line Configuration dialog. All configuration data entered in this dialog is applied individually to every line that is a member of the Wideband port.

ISDN Interface — Bearer Capability

The settings for these parameters determine the bearer transfer characteristics of the ISDN for outgoing calls. This information element is used by the ISDN to determine whether or not interworking equipment is necessary. It does not affect the characteristics of the digital interfaces. Usually, only octets 3 and 4 are required. The following table shows the default settings. If the other octets are required they should be set to the same values as the Low Layer Compatibility information element. The default values are suitable for almost every application.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest.Digital
Octet 4	Mode	Circuit
	Rate	64 kbit/s

ISDN Interface — Low Layer Compatibility

The Low Layer Compatibility information element is transmitted when setting up an outgoing call, and optionally is used to check incoming calls. It is used to configure the data rate and asynchronous data format. The default values are shown below for Low Layer Compatibility at speeds less than 64 kbit/s where V.110/X.30 rate adaption is required.

	Parameter	Default
Octet 3	Standard	CCITT (ITU-T)
	Transfer	Unrest. Digital
Octet 4	Mode	Circuit
	Rate	64 kbit/s
Octet 5	Layer 1	CCITT (ITU-T) rate adaption

The settings for these parameters are determined by the capability of the equipment used. The settings are applied to incoming calls only. For further information about the default settings, refer to *Parameters* page 2-7.

ISDN Interface — High Layer Compatibility defaults

	Parameter	Default
Call Direction	Incoming	None
	Outgoing	None

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults**AOC
(where available)**

Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID

Do not select **CLIP** (CLI Presentation).

If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

**CLIVE List
Assignment**

Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format

Select the appropriate CCITT (ITU-T) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

ISDN Interface defaults**Teleservice
(where available)**

None.

Changing ISDN Interface defaults**Teleservice
(where available)**

Select **Timelink** if you have subscribed to this service.

ISDN Internal Control defaults

	V.24	V.35	X.21
Restriction	None	None	None
Operation	DCE	DCE	DCE
Synch mode	Synch	—	—
Tx Clock	DCE	DCE	—
Outgoing TE	Any	Any	Any

Changing ISDN Internal Control defaults

- Restriction** Select **Disable** to prevent calls being made to or from this line. Select **In Only** or **Out Only** to allow calls in the selected direction only.
- Tx Clock** Select either **DCE Tx Clock** or **DTE Tx Clock** for the V.24 and V.35 interfaces. The receive clock is always sourced by the DCE. These clock source rules are true when the interface is configured for either DTE or DCE operation.
- Outgoing TE** Select the required TE. Any of the IPMT, BRMT or ISO-BCAM modules may be selected.

Configuring COMBO with VC-G7231-2

This section details the available parameters for the COMBO with the VC-G7231-2 compression submodule.

Mode options

- Echo cancellation** This feature removes echo from the line by electrically cancelling the echoes up to 16ms long. The echo canceller complies with ITU-T recommendation G.165.
- Enhanced Compander (NRC and above)** Enabling this feature downloads a later version of software to the compander DSP. The new software improves the performance of DTMF dialling for those applications that require DTMF. The module should be reset after enabling or subsequently disabling this feature.

Note *All MFCMs (with MLQ-CMP) and COMBO modules (with VC-G7231-2) in the same network should have the feature all enabled or all disabled. A mixture of settings may cause network dialling problems.*

Audio coding type

Select the encoding type — A-law or μ -law.

Configuring COMBO with HD-VCM

RM

This section details the available parameters for a COMBO fitted with the HD-VCM submodule.

Options

Note *We recommend that the following parameters are left at their default settings. The defaults will suit most applications in which the COMBO with HD-VCM is used.*

Echo cancellation	This feature removes echo from the line by electrically cancelling the echoes up to 25ms long. The echo canceller complies with ITU-T recommendation G.165. The default is enabled.
Silence Compression	If this parameter is enabled (default is disabled), speech silence periods are not sent as data. In this way, the number of data packets transmitted are reduced considerably. In packet-oriented networks, the bandwidth requirements for voice calls is substantially reduced. We recommend that this parameter is enabled for packet-oriented networks.
High Pass Filter	When enabled (the default), the filter cancels noises at the coder input. We recommend this is left enabled for most applications. When multiple tandem (more than one voice compressor in the same call path) is expected to be used in the channel, high pass filtering can cause quality degradation. When this is the case, the filter can be disabled.
Post Filter	When the post filter is enabled (the default), a filter is applied after the decoder to improve voice quality. We recommend this is left enabled for most applications. When multiple tandem (more than one voice compressor in the same call path) is expected to be used in the channel, sequential post filtering can cause quality degradation. When this is the case, the filter can be disabled.

Audio coding type

Select the encoding type — A-law or μ -law.

Algorithm

In this release, only MP-MLQ is supported. OmniVision displays the different types of voice compression algorithms as:

- **MP-MLQ(VFM2)** – used on the VC-G7231-2 submodule, and;
- **G.723.1 (6.3k)** – used on the HD-VCM.

You should select **G.723.1 (6.3k)** as the algorithm type for the HD-VCM.

Note *Although the algorithm is the same as that used for COMBOs fitted with the VC-G7231-2 submodule, its implementation is different. The HD-VCM is therefore not compatible with the MLQ-CMP on MFCMs, the VF-CMP on LDCMs, or the VC-G231-2 on the COMBO.*

Tips on Configuring Modem Relaying

To set up Modem Relaying over the COMBO with HD-VCM, use the following instructions.

Setting up the Terminal software

- 1 Ensure that you have serial communication (Terminal) software (e.g. Telix, Procomm or Hyperterminal) installed on the PC or workstation.
- 2 Connect the PC or Workstation's serial port to the modem using a suitable modem cable with all the pins connected.
- 3 Set up a terminal (DTE) speed of 57600 bit/s, No Parity, eight Data bits and one stop bit, or what ever suits the modem being used.
- 4 Configure the terminal application Flow Control method to either **Hardware** or **CTS/RTS** only.

Configuring the Modem

At this stage in the set up, the modem should also be connected to a telephone wall socket or a suitable module (for example, an ALEM) using a telephone line cable.

- 1 Type `AT <enter>` to determine if the terminal is communicating with the modem. If the modem receives the command, it responds with `OK`.
- 2 Once you have established communications between the terminal and the modem, restore the factory settings with the command: `AT&F0` or `AT&F1`.
- 3 To have the connection parameters displayed when the modem makes a successful call, type: `ATS95=63`.
- 4 To configure hardware flow control, type: `AT&k3`.

Making a Call

To make a call, type: `ATD<number>` where `<number>` is the telephone number of the modem on the far end of the connection.

If the call establishes a connection with the far-end modem, you should get the modem displaying the following information on the terminal screen. This assumes that the connection has a 16 kbit/s bandwidth running modem relaying at 14.4 kbit/s:

```
CARRIER 14400
PROTOCOL: LAP-M
COMPRESSION: V.42BIS
CONNECT 14400/ARQ
```

Note *The maximum modem speed supported is dependent on the bandwidth of the call, For example:*

- *For 8 kbit/s calls, the maximum fax and modem relay speed is 4800 bit/s.*
 - *For 16 kbit/s calls, the maximum fax and modem relay speed is 14400 bit/s.*
-

Forcing Modem Relaying Connection at slower speeds

You can limit the fax and modem speeds as follows:

- 1 Before making the call, set up the modem (as described in *Configuring the Modem* above) with the speed set to 4800 bit/s, and then type:

```
AT+ms=10,0,4800,4800
```

or (depending on modem);

```
AT+ms=10,0,4800,4800,1,0,33600
```

Note *Some modems have a variation of this command. Use: `AT+ms=?` to find out more about your particular modem.*

Diagnostics and monitoring

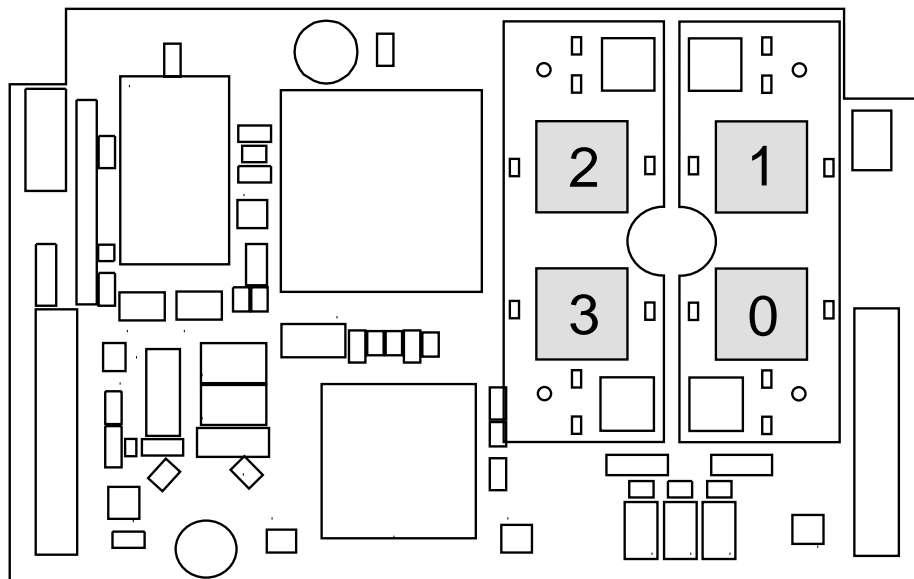
OmniVision can be used to monitor the status of each voice channel on the COMBO used with the HD-VCM. The channels on each HD-VCM submodule are mapped to a line diagnostic dialog in OmniVision.

Card States

Selecting **Diagnostics** opens the **Choose Line** dialog which displays the logical number of channels on the COMBO. Coloured indicators against each channel shows its current status. See the *OmniVision User Manual* for information about the relevance of each colour.

Choose the channel that you want to diagnose, and a **Diagnostics** dialog is displayed. The **Card States** information shows the **Physical Position** of the channel. This is displayed as Module Number, Chip Number and Channel Number to assist you in identifying a fault. For example, **Module:0, Chip:1, Channel:4** indicates the channel is on Compander Chip 1 of submodule 0.

The following diagram shows the positions of each compander chip in relation to the HD-VCM submodule.



Call information

In addition to the normal call information available through the diagnostics facility, the COMBO with HD-VCM has a 'Ping' feature. To view the status of the Ping, select the **Call Information** button in the **Diagnostics** dialog. The Ping read-out in the **Call Information** display shows the end-to-end packet propagation delay in milliseconds. If no time is shown, the line may be faulty. See the *OmniVision User Manual* for more details on call information.

General configuration

As with the SRMM and VCM, the Subrate multiplexing function of this module can be used to transit-switch Data Leased Lines and Voice Tie Line calls. In this case, you may need to change some of the Call Establishment Modes settings to accommodate the type of connection and signalling used, and set up an internal connection within the device.

Most of the options available on the COMBO are selected through the Virtual Line dialog box on the Resource Manager or Control Module. This is selected through the RM or CM using OmniVision.

A Virtual Line is a user-defined line; that is, it is not a physical line in the device. So you can define a Virtual Line as a combination of physical lines, then reference this Virtual Line when you want to make a call. You must select the type of Virtual Line you want to create. Whenever you are using a COMBO, you must select a TimeFrame Virtual Line or a Subrate Virtual Line, as the COMBO typically combines circuits of less than 64 kbit/s to make up a total of 64 kbit/s.

Choosing elements for the Subrate Virtual Line

Enter a name for the line and select New VL. In the Define Subrate Virtual Line dialog box, select the physical lines that are to make up the Virtual Line. Search through the Available Lines list. For the COMBO you usually select at least two voice ports, and then some additional combination of voice and data ports to make up the 64 kbit/s. Select the particular voice port you want and choose **Add**. The line appears on the left-hand side under Virtual Line Members.

Selecting voice/data rates for a Subrate Virtual Line

To specify the bandwidth (how many kbit/s) of the 64 kbit/s you want allocated to that physical port, click on the boxes next to the selected ports shown in the lefthand side. Each column represents 8 kbit/s of the total bandwidth. With a COMBO, any single voice port can be given 8 kbit/s or 16 kbit/s of the total 64 kbit/s.

Note *It is very important that the Subrate Virtual Line at the far end has the same columns selected for the physical ports that want to communicate using the Virtual Line. The columns selected define which portion of the 64 kbit/s Virtual Line gets sent to which physical port.*

Next, select the type of compression. The compression mode available on the COMBO with the VC-G7231-2 is **MP-MLQ (VFM2)**. If your COMBO is fitted with the HD-VCM the compression mode is **G.723.1 (6.3k)**.

Note *It is very important that the algorithm you select matches the algorithm you select for the same columns in the Virtual Line at the other end of the call.*

You can use any combination of data rates inside a single Virtual Line, as long as you have enough COMBOs or MFCMs in the device to accommodate them.

Note *No checks are made on whether there are enough MFCM or COMBO resources to handle all Virtual Lines being connected simultaneously. This is because the resources are dynamically allocated (like PABX outgoing circuits) on a per call basis. This provides flexibility in the operation of your network.*

Configuring the Subrate Virtual Line

Having selected the elements you want included in your 64 kbit/s Virtual Line, you must now configure it.

See *Virtual Lines* on page 3-21 for further details. Use OmniVision to set the VL's Line Number, Hotline Number, SPC Number, Auto Answer mode and so on.

The **Line Number** should be set to the line number you want the distant party to call to 'ring' the Virtual Line. The **End-to-End Signalling Mode** of the Virtual Line should be selected. This permits the call signalling information, originating from either a data or voice port to be passed from end to end.

After configuring one side of the connection, the process must be repeated for the other side. You must logon to the device at the other side of the link and repeat the configuration process.

Configuring the TimeFrame Virtual Line

This section cover the various aspects of choosing elements and configuring a TimeFrame Virtual Line.

There is no configuration action required in order to choose the COMBO for use with a TimeFrame Virtual Line. The TimeFrame Virtual Line negotiates with the far end to determine which type of voice compression algorithm to use for each voice call.

If an MFCM or COMBO fitted with HD-VCMs is available at both ends of a call, **G.723.1 (6.3k)** is used. If an MFCM with MP-MLQ submodules, or a COMBO with VC-G7231-2 submodules is available at both ends, then **MP-MLQ (VFM2)** is chosen. If both types are available at each end, the TimeFrame Virtual Line selects **G.723.1 (6.3k)** as its first choice, and only reverts to **MP-MLQ (VFM2)** when no **G.723.1 (6.3k)** companders are free.

Note *Only 8 kbit/s Subrate is supported on a TimeFrame Virtual Line.*

For details on configuring the TimeFrame Virtual Line, see *TimeFrame Virtual Line* on page 3-50.



Low Delay CELP Module (LDCM)

The Low Delay CELP Module (LDCM) compresses standard 64 kbit/s A-law encoded voice/fax signals and switches them onto a common channel. The LDCM uses one of the following techniques depending to its model type:

- Code Excited Linear Prediction (CELP) coding technique for voice compression
- MP-MLQ (Multi-Pulse Maximum Likelihood Quantization) technique for voice and fax compression.

LDCM incorporating CELP

There are four variations of the LDCM CELP: LDCM-1/2, LDCM-2/4, LDCM -3/6 and LDCM-4/8. The last digit specifies the number of compander (compressor/expander) functions provided on each module type. An LDCM-1/2 has two companders, and LDCM-2/4 has four companders, and so on.

The LDCM can have up to eight DSPs (Digital Signal Processors). Two CELP DSPs are resident on the LDCM board and others can be added by fitting CELP submodules (two DSPs per submodule). The CELP DSP submodule supports the following voice compression rates:

- 8 kbit/s CELP
- 16 kbit/s CELP
- 16 kbit/s Low Delay CELP.

The 16 kbit/s LDCELP voice compression is fully compliant with ITU-T recommendation G.728. Private networks using this feature can be connected to the Public Telecommunications Network (PTN).

The 8 kbit/s and 16 kbit/s CELP voice compression employ proprietary algorithms. These features can only be used in Private Networks (PN).

LDCM incorporating the voice/fax submodule

There are three models of the LDCM with the voice/fax submodule. These are referred to as LDCM1/VF2, LDCM1/VF4 and LDCM1/VF6. The last digit specifies the number of Voice/Fax compander (compressor/expander) functions provided on each module type. As the LDCM already has two on board CELP companders, LDCM1/VF4 signifies that it has an additional four MP-MLQ companders fitted.

The Voice /Fax submodule provides a G3 fax detection/relay capability in addition to enhanced voice compression at the following rates:

- 8kbit/s MP-MLQ
- 16 kbit/s MP-MLQ.

The 6.3 Kbit/s MP-MLQ algorithm provides 8 kbit/s and 16 kbit/s channel voice compression and is compliant with ITU-T recommendation G.723.1.

See *Appendix K* for a list of fax machines that have been tested with the LDCM.

Operation

The LDCM allows multiple voice sub-channels to be transmitted over a single 64 kbit/s channel. The receiving end then expands the voice sub-channels back to 64 kbit/s A-law. A combination of voice and data sub-channels can also be transmitted over a common channel via the Subrate data switch. The LDCM always combines sub-channels to produce a total of 64 kbit/s.

Voice and/or Fax signals access the LDCM through the standard interface modules used for voice applications, such as the Analog Line Exchange Module (ALEM), Analog Line Phone Module (ALPM), E & M Module (EMM) or the 2.048 Mbit/s Digital Module NT (E1MN). The signals are directed via the internal buses to the LDCM.

Echo suppression

This feature removes echo from the line by allowing speech in one direction at a time. Since there is no voice path in the return direction, no echo is heard. When both parties in the call talk at the same time, this feature is disabled during that period allowing two way speech communication. This will allow echo. However, since both parties are talking, the echo will not be noticed.

Gain Control for CELP

A gain control is provided on both the input and output side of the 8K CELP, 16K CELP and 16K Low Delay CELP lines. This feature can provide an improvement in the quality of the voice by presenting a level to the CELP line which is within the optimum range. When the level is too high, both distortion and echo are increased. The gain setting affects the overall volume of the voice and echo. We recommend that you use this feature only when the relative level can not be adjusted on third-party systems.

For information on using the LDCM for various types of applications see page 5-47 of the *Applications* section.

Configuration

As with the SRMM, the Subrate multiplexing function of this module can be used to transit-switch Data Leased Lines and Voice Tie Lines. In this case, you may need to change some of the Call Establishment Modes settings to accommodate the type of connection and signalling used, and set up an internal connection within the device.

Most of the options available on the LDCM are selected through OmniVision's *Virtual Line* facility. See page 3-21 for more information.

A Virtual Line is a user-defined line; that is, it is not a physical line. So you can define a Virtual Line as a combination of physical lines, then reference this Virtual Line when you want to make a call. You must select the type of Virtual Line you want to create. Whenever you are using an LDCM, you must select a TimeFrame or a Subrate Virtual Line, as the LDCM always combines circuits of less than 64 kbit/s to make up a total of 64 kbit/s.

Choosing elements for the Subrate Virtual Line

Enter a name for the line and select New VL. In the Define Virtual Line dialog box select the physical lines that are to make up the Virtual Line. Search through the Available Lines list. For the LDCM you usually select at least two voice ports, and then some additional combination of voice and data ports to make up the 64 kbit/s. Select the particular voice port you want and choose **Add**. The line appears on the left-hand side under Virtual Line Members.

Selecting voice/data rates

To specify the bandwidth (how many kbit/s) of the 64 kbit/s you want allocated to that physical port, click on the boxes next to the selected ports shown in the lefthand side. Each column represents 8 kbit/s of the total bandwidth. With an LDCM, any single voice port can be given 8 kbit/s or 16 kbit/s of the total 64 kbit/s.

Note *It is very important that the Subrate Virtual Line at the far end has the same columns selected for the physical ports that want to communicate using the Virtual Line. The columns selected define which portion of the 64 kbit/s Virtual Line gets sent to which physical port.*

Next, select the type of compression. The compression modes available on the LDCM (CELP and MP-MLQ) are as shown in the table below.

Compression Rate	LDCELP	Proprietary (CELP)	Voice/Fax (MP-MLQ)
8 kbit/s		LDCM	LDCM-VF
16 kbits/s	LDCM	LDCM	LDCM-VF
24 kbits/s			
32 kbits/s			

For the LDCM, you can choose either CELP, LDCELP or MP-MLQ (VFM2) (depending on the LDCM model and the data rate chosen).

Note *It is very important that the algorithm you select matches the algorithm you select for the same columns in the Virtual Line at the other end of the call.*

You can use any combination of algorithms and data rates inside a single Virtual Line, as long as you have enough LDCMs in the chassis to accommodate them.

Note *No checks are made on whether there are enough LDCMs to handle all Virtual Lines being connected simultaneously. This is because the LDCMs are dynamically allocated (like PABX outgoing circuits) on a per call basis. This provides flexibility in the operation of your network.*

Configuring the Subrate Virtual Line

Having selected the elements you want included in your 64 kbit/s Virtual Line, you must now configure it.

The Virtual Line now appears like any other type of 64 kbit/s line. You can set its Line Number, Hotline Number, SPC Number, Auto Answer Mode, and so on.

The Line Number should be set to the line number you want the distant party to call to 'ring' the Virtual Line. Given that you usually include some voice circuits in the Virtual Line, the End-to-End Signalling Mode of the Virtual Line should be selected. This permits the call signalling information to be passed from end to end, thus allowing the telephone to ring when the corresponding handset is lifted at the other end of this link.

After configuring one side of the connection, the process must be repeated for the other side. You must logon to the device at the other side of the link and repeat the configuration process.

Configuring the TimeFrame Virtual Line

There is no configuration required in order to choose the voice compression algorithm for a TimeFrame Virtual Line. It will always use the 8 kbit/s MP-MLQ algorithm.

Refer to *TimeFrame Virtual Line* on page 3-50 for details on configuring a TimeFrame Virtual Line.

Mode options

The LDCM has three mode options for echo control:

- Echo cancellation — the default
- Echo suppression — select this to enable echo suppression parameters
- No echo processing.

Configuring echo suppression (CELP only)

Suppressor Cut-Out Time	This defines the amount of time that the backward path is disabled after a party finishes talking. This is required since the echo is delayed relative to the speech and the backward path must not be enabled until the echo has gone. It is adjustable in the range of 0 to 150 ms. The default is 0.
Suppressor Cut-In Level	This defines the level at which the suppressor detects speech from one of the two parties. It is adjustable in the range of 0-15. The default is 0.

Relative level options (CELP only)

Pre-Algorithm Input gain	The input gain can be adjusted from 9dB to -12dB.
Pre-Algorithm Output gain	The output gain can be adjusted from 12dB to -9dB.



Multi Function Compression Module (MFCM)

The Multi Function Compression Module is designed to compress standard 64 kbit/s A-law or μ -law encoded voice channels. A number of compressed voice channels can then be switched onto a single common channel for onward transmission across a network. The MFCM achieves the compression by the use of companders mounted on submodules. The MFCM can have up to four submodules fitted. These can be either:

- MP-MLQ Compander submodules (MLQ-CMP), or,
- High Density Voice Compression Modules (HD-VCM)

Note *A mixture of MLQ-CMP and HD-VCM submodules cannot be supported on the same MFCM.*

An MFCM populated with four MLQ-CMP compander submodules can provide compression for up to 8 voice channels. An MFCM fitted with four HD-VCM-15 compander submodules provides compression for up to 60 voice channels.

Operation

The MFCM at one end of a network compresses multiple voice subchannels. An MFCM at the receiving end then expands the voice subchannels back to 64 kbit/s A-law or μ -law. A combination of voice and data subchannels can also be compressed by the on-board substrate data switches. The MFCM combines subchannels to produce a single channel of 64 kbit/s, even if the total bandwidth of the subchannels is less than 64 kbit/s.

Voice and/or Fax signals (and Modem signals with HD-VCM) access the MFCM through the standard interface modules used for voice applications, such as the Analog Line Exchange Module (ALEM-2), Analog Line Phone Module-2 (ALPM-2), E & M Module (EMM) or the 2.048 Mbit/s Digital Module NT (E1MN). The signals are directed via the internal buses in the chassis to the MFCM.

For information on using the MFCM for various types of applications see page 5-47 of the *Applications* section.

See *Appendix K* for a list of fax machines that have been tested with the MFCM.

Configuring MFCM with MLQ-CMP

RM CM

This section details the available parameters for an MFCM fitted with the MLQ-CMP submodule.

Mode options

Echo cancellation	This feature removes echo from the line by electrically cancelling the echoes up to 16ms long. The echo canceller complies with ITU-T recommendation G.165.
Enhanced Compander (NRC and above)	Enabling this feature downloads a later version of software to the compander DSP. The new software improves the performance of DTMF dialling for those applications that require DTMF. The module should be reset after enabling or subsequently disabling this feature.

Note *All MFCMs (with MLQ-CMP) and COMBO modules in the same network should have the feature all enabled or all disabled. A mixture of settings may cause network dialling problems.*

Audio coding type

Select the encoding type — A-law or μ -law.

Configuring MFCM with HD-VCM

RM

This section details the available parameters for an MFCM fitted with the HD-VCM submodule.

Options

Note *We recommend that the following parameters are left at their default settings. The defaults will suit most applications in which the MFCM with HD-VCM is used.*

Echo cancellation	This feature removes echo from the line by electrically cancelling the echoes up to 25ms long. The echo canceller complies with ITU-T recommendation G.165. The default is enabled.
Silence Compression	If this parameter is enabled (default is disabled), speech silence periods are not sent as data. In this way, the number of data packets transmitted are reduced considerably. In packet-oriented networks, the bandwidth requirements for voice calls is substantially reduced. We recommend that this parameter is enabled for packet-oriented networks.
High Pass Filter	When enabled (the default), the filter allows high-frequency components of the voice channel to pass. When disabled, the high-frequency components are suppressed. Suppressing high-frequencies may help to improve voice reception in a noisy environment.
Post Filter	When the post filter is enabled (the default), a filter is applied after the decoder to improve voice quality. We recommend this is left enabled for most applications. The filter can, however, alter the properties of the received voice. Where this is not desirable, the filter can be disabled.

Audio coding type

Select the encoding type — A-law or μ -law.

Algorithm

In this release, only MP-MLQ is supported. OmniVision displays the different types of voice compression algorithms as:

- **MP-MLQ(VFM2)** – used on the MLQ-CMP and VF-CMP submodules, and;
- **G.723.1 (6.3k)** – used on the HD-VCM.

You should select **G.723.1 (6.3k)** as the algorithm type for the HD-VCM.

Note *Although the algorithm is the same as that used for MFCMs fitted with the MLQ-CMP submodule, and the LDCM fitted with the VF-CMP submodule, its implementation is different. The HD-VCM is therefore not compatible with the MLQ-CMP or VF-CMP.*

Tips on Configuring Modem Relaying

To set up Modem Relaying over the MFCM with HD-VCM, use the following instructions.

Setting up the Terminal software

- 1 Ensure that you have serial communication (Terminal) software (e.g. Telix, Procomm or Hyperterminal) installed on the PC or workstation.
- 2 Connect the PC or Workstation's serial port to the modem using a suitable modem cable with all the pins connected.
- 3 Set up a terminal (DTE) speed of 57600 bit/s, No Parity, eight Data bits and one stop bit, or what ever suits the modem being used.
- 4 Configure the terminal application Flow Control method to either **Hardware** or **CTS/RTS** only.

Configuring the Modem

At this stage in the set up, the modem should also be connected to a telephone wall socket or a suitable module (for example, an ALEM) using a telephone line cable.

- 1 Type `AT <enter>` to determine if the terminal is communicating with the modem. If the modem receives the command, it responds with `OK`.
- 2 Once you have established communications between the terminal and the modem, restore the factory settings with the command: `AT&F0` or `AT&F1`.
- 3 To have the connection parameters displayed when the modem makes a successful call, type: `ATS95=63`.
- 4 To configure hardware flow control, type: `AT&k3`.

Making a Call

To make a call, type: `ATD<number>` where `<number>` is the telephone number of the modem on the far end of the connection.

If the call establishes a connection with the far-end modem, you should get the modem displaying the following information on the terminal screen. This assumes that the connection has a 16 kbit/s bandwidth running modem relaying at 14.4 kbit/s:

```
CARRIER 14400
PROTOCOL: LAP-M
COMPRESSION: V.42BIS
CONNECT 14400/ARQ
```

Note *The maximum modem speed supported is dependent on the bandwidth of the call, For example:*

- *For 8 kbit/s calls, the maximum fax and modem relay speed is 4800 bit/s.*
 - *For 16 kbit/s calls, the maximum fax and modem relay speed is 14400 bit/s.*
-

Forcing Modem Relaying Connection at slower speeds

You can limit the fax and modem speeds as follows:

- 1 Before making the call, set up the modem (as described in *Configuring the Modem* above) with the speed set to 4800 bit/s, and then type:

```
AT+ms=10,0,4800,4800
```

or (depending on modem);

```
AT+ms=10,0,4800,4800,1,0,33600
```

Note *Some modems have a variation of this command. Use: `AT+ms=?` to find out more about your particular modem.*

Diagnostics and monitoring

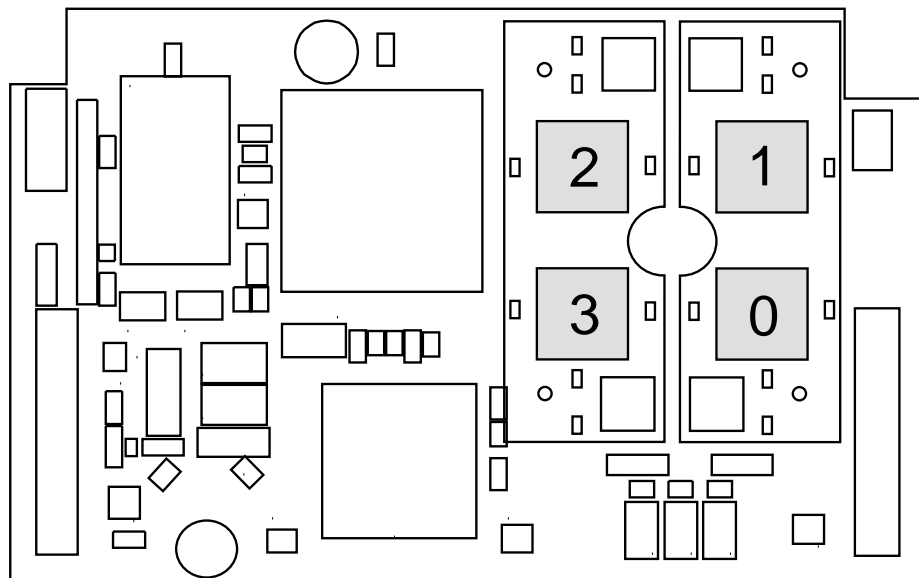
OmniVision can be used to monitor the status of each voice channel on the MFCM used with the HD-VCM. The channels on each HD-VCM submodule are mapped to a line diagnostic dialog in OmniVision.

Card States

Selecting **Diagnostics** opens the **Choose Line** dialog which displays the logical number of channels on the MFCM. Coloured indicators against each channel shows its current status. See the *OmniVision User Manual* for information about the relevance of each colour.

Choose the channel that you want to diagnose, and a **Diagnostics** dialog is displayed. The **Card States** information shows the **Physical Position** of the channel. This is displayed as Module Number, Chip Number and Channel Number to assist you in identifying a fault. For example, **Module:0, Chip:1, Channel:4** indicates the channel is on Compander Chip 1 of submodule 0.

The following diagram shows the positions of each compander chip in relation to the HD-VCM submodule. Refer to the *MFCM* section in the *J1000/J5000 Installation Manual* for information on submodule positions.



Call information

In addition to the normal call information available through the diagnostics facility, the MFCM with HD-VCM has a 'Ping' feature. To view the status of the Ping, select the **Call Information** button in the **Diagnostics** dialog. The Ping read-out in the **Call Information** display shows the end-to-end packet propagation delay in milliseconds. If no time is shown, the line may be faulty. See the *OmniVision User Manual* for more details on call information.

General configuration

As with the SRMM and VCM, the Subrate multiplexing function of this module can be used to transit-switch Data Leased Lines and Voice Tie Line calls. In this case, you may need to change some of the Call Establishment Modes settings to accommodate the type of connection and signalling used, and set up an internal connection within the device.

Most of the options available on the MFCM are selected through the Virtual Line dialog box on the Resource Manager or Control Module. This is selected through the RM or CM using OmniVision.

A Virtual Line is a user-defined line; that is, it is not a physical line in the device. So you can define a Virtual Line as a combination of physical lines, then reference this Virtual Line when you want to make a call. You must select the type of Virtual Line you want to create. Whenever you are using an MFCM, you must select a TimeFrame Virtual Line or a Subrate Virtual Line, as the MFCM typically combines circuits of less than 64 kbit/s to make up a total of 64 kbit/s.

Choosing elements for the Subrate Virtual Line

Enter a name for the line and select New VL. In the Define Subrate Virtual Line dialog box, select the physical lines that are to make up the Virtual Line. Search through the Available Lines list. For the MFCM you usually select at least two voice ports, and then some additional combination of voice and data ports to make up the 64 kbit/s. Select the particular voice port you want and choose **Add**. The line appears on the left-hand side under Virtual Line Members.

Selecting voice/data rates for a Subrate Virtual Line

To specify the bandwidth (how many kbit/s) of the 64 kbit/s you want allocated to that physical port, click on the boxes next to the selected ports shown in the lefthand side. Each column represents 8 kbit/s of the total bandwidth. With an MFCM, any single voice port can be given 8 kbit/s or 16 kbit/s of the total 64 kbit/s.

Note *It is very important that the Subrate Virtual Line at the far end has the same columns selected for the physical ports that want to communicate using the Virtual Line. The columns selected define which portion of the 64 kbit/s Virtual Line gets sent to which physical port.*

Next, select the type of compression. The compression mode available on the MFCM with the MLQ-CMP is **MP-MLQ (VFM2)**. If your MFCM is fitted with the HD-VCM the compression mode is **G.723.1 (6.3k)**.

Note *It is very important that the algorithm you select matches the algorithm you select for the same columns in the Virtual Line at the other end of the call.*

You can use any combination of data rates inside a single Virtual Line, as long as you have enough MFCMs in the device to accommodate them.

Note *No checks are made on whether there are enough MFCM resources to handle all Virtual Lines being connected simultaneously. This is because the resources are dynamically allocated (like PABX outgoing circuits) on a per call basis. This provides flexibility in the operation of your network.*

Configuring the Subrate Virtual Line

Having selected the elements you want included in your 64 kbit/s Virtual Line, you must now configure it.

See *Virtual Lines* on page 3-21 for further details. Use OmniVision to set the VL's Line Number, Hotline Number, SPC Number, Auto Answer mode and so on.

The **Line Number** should be set to the line number you want the distant party to call to 'ring' the Virtual Line. The **End-to-End Signalling Mode** of the Virtual Line should be selected. This permits the call signalling information, originating from either a data or voice port to be passed from end to end.

After configuring one side of the connection, the process must be repeated for the other side. You must logon to the device at the other side of the link and repeat the configuration process.

Configuring the TimeFrame Virtual Line

This section cover the various aspects of choosing elements and configuring a TimeFrame Virtual Line.

There is no configuration action required in order to choose the MFCM for use with a TimeFrame Virtual Line. The TimeFrame Virtual Line negotiates with the far end to determine which type of voice compression algorithm to use for each voice call.

If an MFCM fitted with HD-VCMs is available at both ends of a call, **G.723.1 (6.3k)** is used. If an MFCM with MP-MLQ submodules is available at both ends, then **MP-MLQ (VFM2)** is chosen. If both types are available at each end, the TimeFrame Virtual Line selects **G.723.1 (6.3k)** as its first choice, and only reverts to **MP-MLQ (VFM2)** when no **G.723.1 (6.3k)** companders are free.

Note *Only 8 kbit/s Subrate is supported on a TimeFrame Virtual Line.*

For details on configuring the TimeFrame Virtual Line, see *TimeFrame Virtual Line* on page 3-50.



Subrate Switch Multiplexer Module (SRMM)

The Subrate Switch Multiplexer Module (SRMM) provides a Subrate switching facility which enables a number of data leased lines to be directed to their destination via a single 64 kbit/s B-channel. Up to 64 half duplex sub-channels or 32 full duplex sub-channels can be switched using an SRMM.

This module is used as a resource by the Resource Manager or Control Module and, as such, has no configurable parameters. However, it can also be used to transit switch data leased lines and voice tie lines. You may need to change some of the call establishment mode settings to accommodate the type of connection and signalling used and any internal connections. Refer to the *OmniVision User Manual* for further instructions.

Configuration

Configuration is only required if an SRMM is terminating a call which is transit switched. In this case the Line number, Low Layer Compatibility and Call Control parameters should be configured and all other options left at the default value. See *Parameters* for more information.



Voice Compression Module (VCM)

The Voice Compression Module (VCM) can perform two separate functions: voice compression and Subrate multiplexing.

The voice compression function allows multiple fixed voice channels (tie lines) to use one or more 64 kbit/s B-channels and decompresses (expands) them at the other end of the system via another VCM.

The VCM also provides a Subrate data switching function, combining voice and/or data onto a common channel.

This module is used as a resource by the Resource Manager or Control Module and, as such, has no configurable parameters. However, as with the SRMM, MFCM and LDCM, the Subrate multiplexing function of this module can also be used to transit switch data leased lines and voice tie lines. In this case, you may need to change some of the Call Establishment Modes settings to accommodate the type of connection and signalling used, and set up an internal connection within the device. Refer to the *OmniVision User Manual* for further instructions.

Auto mute

The VCM supports the generation of compressed service tones.

When the VCM receives a constant bit pattern, the ADPCM generates a screeching sound that is unpleasant to the person listening to the line. Normally the VCM overcomes this by detecting a burst of constant bit pattern and then muting the channel.

This feature can be disabled for switches, such as MD110s, that provide a constant bit pattern before sending DTMF tones. If Auto Mute is enabled (default) the VCM may prevent the beginning of the DTMF tones from being passed. In this case the Auto Mute should be disabled.

To disable Auto Mute, click on the top section of the VCM, and check the 'Disable Auto Mute' checkbox.

B-channel Aggregation Module AS4064 (BCAM)

The B-channel Aggregation Module (BCAM) supports user speeds ranging from 63 kbit/s to 504 kbit/s across the ISDN. It performs all the channel synchronisation and sequence alignment necessary to aggregate the required B-channels. We refer to this as Channel Aggregation, but it may also be called Inverse Multiplexing, ISDN Super Rate Adaption or Reverse Multiplexing. The BCAM aggregation method conforms to procedures and modes defined by AS4064.

The BCAM is a DCE (Data Circuit Equipment) as defined by ITU-T (CCITT) recommendations. It can have either an X.21 or a V.35 interface on the front panel and can be used as a resource module by an E1M.

Configuration

BCAM (Set) Number	By default, all lines in a BCAM are allocated the same number. However, it is sometimes necessary to number all lines individually. For example, in the case of multiple BRAs when each line has a separate number.
	To give all of the BCAM lines the same number, enter the number allocated by your service provider. If you wish to number the lines individually, leave this parameter blank and refer to <i>Line configuration</i> on page 2-5 of the <i>Parameters</i> section for further information.
	If you leave this parameter blank, the module can still make calls, but cannot receive them.
R6 Compatible	(Default - Cleared). Select this option to allow calls to be established between Release 6 and later release BCAMs.
Delay X.21 connect	(Default - Cleared). Select this option so that the data rate presented to the user interface is as configured before and after calls. The user rate is 64 kbits/s until one second after the BCAM has synchronised following call establishment. The user rate then changes to the configured or negotiated value.
Don't send LCC	(Default - Cleared). Calls initiated from the BCAM include LLC information in the setup messages, even when 'Inband' is selected. Select this option to prevent the LLC from being included. This should not be selected if 'Inband' is not selected.

Configuration defaults

The table below shows the default settings for BCAM Configuration — Port Configuration, Data Source and Call Establishment.

	Parameter	Default
Port Configuration	Channel Rate	64 kbit/s
	Negotiable	Cleared
	Inband	Cleared
	User Rate No. of Channels	384 kbit/s 7
Data Source	Front Panel interface	Selected
	E1 Module in...	Cleared
	Card Slots/WAN ports	—
	1st Channel	—
Call Establishment	Auto Answer	Selected
	XSPC	Cleared
	Multiple Hotline To	—
	Fast Dial	Cleared
	Interface lead controlled	Cleared
	Accept Broadcast Calls	Cleared
	From Channels	Selected

Changing Port Configuration defaults

- Negotiable** Select this to enable the user rate to be negotiated.
- Inband** Select this to enable the data rate to be negotiated Inband. When Inband is enabled the BCAM uses inband negotiation according to AS 4064. When it is not enabled the BCAM uses LLC.
- User Rate** Select the required rate from the displayed list. (The number of channels is set automatically.)

Changing Data Source defaults

- Front Panel Interface** Clear this to allow an E1M to be used as the data source in another slot.
- E1 Module In** Select this if the data source is an E1M. If you select this you must also select the card slot of the E1M that is to be used as the data source.
- IRM** Select this if the data source is an IRM module. If you select this you must also select the IRM port that is to be used as the data source.
- Card Slot** Select the card slot of the E1M that is to be used as the data source.
- 1st Channel** This parameter is available if you select **E1 Module In**. You must select the first channel. The next seven channels will be allocated sequentially from that channel to the required user rate.

Changing Call Establishment defaults

Fast Dial	Not available.
Multiple Hotline To	Select this and enter a Hotline number to which all the channels are to be directed. You cannot select Multiple Hotline if you have set up individual channels.
From Channels	Clear this parameter if you entered a BCAM number. If you wish to set an individual number for each line, accept the default (selected). Then select the line number and enter the required information for each line.
Auto Answer	Select this to allow incoming calls to be answered without reference to line interface status.
XSPC	Select this to set up a Switched Semipermanent Connection if required.
Call Control	Select this to enable a DTE to use the X.21 protocol to set up, answer and clear calls. If you select this, you must disable all other options in the Call Establishment dialog box.

Note *Only basic X.21 call establishment and clearing procedures are supported. The enhanced subaddressing and facility registration procedures are not provided.*

C/DTR	Select this to allow calls to be controlled by interface control leads. You can only select this if you selected Front Panel Interface above.
Accept Broadcast Calls	Select this to accept broadcast calls, that is, receive data only.
Channel setup	
Line Number	Enter the line number for the port. Each of the four ports of this module has its own line number. You must enter a line number if the line is to receive incoming calls directly. However, if the line is a member of a lineset or a Virtual Line, no number is required. The line number will be affected when a base number has been configured. See <i>Parameters</i> , page 2-5 for more information.
Subaddress	Enter the subaddress if required. It is not mandatory to enter anything in this parameter.
Port ID	Not applicable to BCAM.
Port Name	Enter the Port Name if required. It is not mandatory to enter anything in this parameter.

ISDN Interface — Call Establishment Modes

The following table shows Call Establishment Modes default settings for each channel.

	Parameter	Default
Semipermanent (where available)	Enable	Cleared
	Label	—
Hotline	Enable	Cleared
	Number	—
	XSPC	Cleared
	Subaddress	—
Call Control	Auto Answer	Selected
Signalling	Remote number	—
	End-to-End	Cleared

Changing Call Establishment Modes defaults

The settings for these parameters control the way in which calls are made. The Call Control and Signalling parameters should not be changed. However, you can set up the following call types:

**Semipermanent
(where available)**

Enter the semipermanent number in the Label parameter and select **Enable**. You should also clear the Hotline checkbox.

Hotline

Enter the Hotline number for the channel and enter a subaddress, if required. Select **Enable**. You should also clear the Semipermanent checkbox.

Select **XSPC** to enable a Switched Semipermanent Connection if required. You can only select this if a Hotline is enabled.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC (where available) Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID Do not select **CLIP** (CLI Presentation).

If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

Note *If you request AOC or CLIR for a PRA, but do not subscribe to them, the call attempt may be rejected by the ISDN.*

CLIVE List Assignment Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format Select the appropriate ITU-T (CCITT) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

Internal Control defaults.

Outgoing TE Any.

Changing ISDN Internal Control defaults

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting is effective on a line by line basis.

B-channel Aggregation Module ISO-13871 (BCAM-ISO)

The BCAM-ISO module transports high-speed serial data across an ISDN using multiple low-speed network bearer channels. The aggregation method conforms to the procedures and modes defined by ISO-13871.

The BCAM-ISO behaves as a DCE (data circuit terminating equipment) as defined in the recommendations of the ITU-T (formerly CCITT). The module can source user data either from an internal interface to the chassis backplane, or from the front panel interface connector.

The internal interface allows other modules, such as an E1M or an IRM, in the same chassis to access the aggregation services of the BCAM-ISO. The internal interface is referred to as the E1M interface. The E1M interface supports aggregation modes 0, 1, 3 and Transparent with a maximum user data rate of 1024 kbit/s.

The BCAM-ISO front panel interface is fitted with either an X.21 or a V.35 connector. The front panel interface supports ISO 13871 aggregation modes 0, 1, 2, 3 and Transparent with a maximum user data rate of 1024 kbit/s.

Configuration

Number (Lineset) By default, all lines in an BCAM-ISO are allocated the same number. However, it is sometimes necessary to number all lines individually. For example, in the case of multiple BRAs when each line has a separate number.

To give all of the BCAM-ISO lines the same number, enter the number allocated by your service provider. If you wish to number the lines individually, leave this parameter blank and refer to *Line configuration* on page 2-5 of the *Parameters* section for further information.

If you leave this parameter blank, the module can still make calls, but cannot receive them.

Configuration defaults

The table below shows the default settings for BCAM-ISO Configuration — Call Rate, Data Source and Call Control, and Called Party Number.

	Parameter	Default
Call Rate	Bearer Rate	64 kbit/s
	User Rate No. of Channels	512kbit/s 9
	Mode	3
	Non Negotiable	Cleared
	Calling endpoint	Cleared
Data Source and Call Control	None (Front panel)	Selected
	Call Control (Front panel)	Cleared
	C/DTR (Front panel)	Cleared
	Fast Dial (Front panel)	Cleared
	X.21	Cleared
	V.25bis	Cleared
	Auto Answer	Cleared
	XSPC	Cleared
	Switched (Backplane)	Cleared
	Manual (Backplane)	Cleared
Called Party Number	Multiple Hotline	Cleared
	From Each Line	Selected

Changing Call Rate defaults

Channel Rate	Selects the network channel rate. You can select only 64 kbit/s.
User Rate	Selects the required rate from the displayed list. (The number of channels is set automatically.) The VX should be configured for more than 16 channels to enable calling at the maximum rates. The number of channels in the list is the number of network channels required for the call.
Mode	Select the required ISO aggregation mode from the list. Mode 0: Initial parameter negotiation is performed, after which the call enters the transparent mode. Channel alignment is performed by the attached user equipment. Mode 1: Data rates are multiples of the bearer channel rate. After initial parameter negotiation and channel alignment, the framing is stripped. The user data rate is the full available bandwidth. However, errors that disturb channel alignment cannot be automatically detected. Mode 2: Data rates are multiples of 63/64 of the bearer channel rate. Parameter negotiation, channel alignment, and framing are enabled. Framing provides a continuous monitor of channel alignment and bit-error rate. The framing occupies 1.66% of the aggregated bandwidth. Mode 3: Data rates are multiples of 8 kbit/s. Parameter negotiation, channel alignment, and framing are enabled. Framing provides a continuous monitor of channel alignment and bit-error rate. The framing overhead is provided by adding bandwidth to the call, thereby preserving the full user data rate. Transparent mode: The aggregated channels are 'cut-through' to the attached application. Initial parameter negotiation, channel phase alignment and framing are disabled. This mode is useful when delay equalisation is performed by the attached terminal equipment (for example, video codecs), or where the remote equipment is not an aggregator.
<hr/> <p>Note <i>The BCAM-ISO module supports ISO 13871 aggregation modes 0, 1 and 2, and a subset of Mode 3. Mode 3 aggregations only supports Nx64k rates (not Nx8k). The E1M interface does not support Mode 2 aggregation. Transparent mode aggregation is supported on all user interfaces.</i></p> <hr/>	
Non Negotiable	Select this to prevent the User Rate from being negotiated.
Calling endpoint	When SPC or NMS manually-connected channels are present on Line 1 of the BCAM-ISO, the call setup is automatic. One endpoint in the call must be nominated as the calling endpoint. Enable if this BCAM-ISO is the calling endpoint.

Aggregation Specials

The Aggregation Specials controls special aggregation features of the BCAM-ISO. The following options are available:

Extend Protocol Timers	This option allows the timer expiry limits of the ISO-13871 protocol to be increased by a fixed, and where required, channel-dependent manner. This option can be used to help establish aggregated calls over long delay or international circuits. Select the number of additional protocol timer seconds from the displayed list. The default value is 0 seconds.
-------------------------------	---

Allow Channel Limiting This option applies to answering endpoints only. Answering endpoints normally accept the data rate and mode requested by the calling endpoint (except when the **Non Negotiable** configuration item is enabled). The option allows an answering endpoint to limit the number of user channels, but still negotiate the call mode.

The option can be used to control call congestion at answering endpoints. To enable this option, select **Allow Channel Limiting** and choose a channel limit value from the displayed list.

Allow Transparent Mode By default, when a call is made to, or received from, a remote endpoint that does not respond with ISO-13871 aggregation signalling, the call would normally fail. If this option is selected, the channels are instead “cut-through” to the attached equipment using transparent mode. This mode is useful when delay equalisation is performed by the attached equipment, or the remote equipment is not an aggregator.

Allow Remote Test Loop Applies to ISO-13871 Modes 2 and 3 only. Select this option to allow the module to loopback the call data when requested by the remote aggregator.

Enable Self-Call Testing Select this option to allow the endpoint to call a test loop (for example, the device test number). This option is used for test purposes. Regardless of whether this option is enabled, the endpoint can still answer calls.

Delayed Call Connect After call rate negotiation, the DCE clock normally rises from the call control rate (64 kbit/s) to the negotiated call rate. When this option is selected, the initial channel is instead placed in transparent mode for a short period (1 second), to allow the DTEs to exchange information at the call control rate. When the clear channel period expires, the DCE clock rises to the negotiated call rate. The clear channel period can be increased with the extend protocol timers option described above.

Note *Both endpoints of the call should be configured for, or support this option. Use of this option may preclude interoperability with DTE's using X.21 call control.*

Downspeed On Connect This option is reserved for future use and is currently disabled.

Changing Data source and Call Control defaults

Auto Answer Select this to allow incoming calls to be answered without reference to line interface status. This option is ignored if the configured call control includes an answering protocol (for example, during X.21 and V.25bis call control).

XSPC Select this to set up a Switched Semipermanent Connection if required.

Front Panel

None Select this if you require no call control from the front panel.

C/DTR Select this to allow calls to be controlled by the Front panel (X.21) or DTR (V.35) control leads.

Call Control Where an X.21 interface is fitted, select this option to enable the DTE to set up, answer, and clear calls using the call control procedures defined by ITU-T X.21.

Where a V.35 interface is fitted, select this option to enable the DTE to set up, answer, and clear calls using the call control procedures defined by ITU-T V.25bis.

Fast Dial Select this option to enable call control using the Fast Dial protocol.

Note *Fast Dial is not available in all markets. Check with your supplier for applicability.*

X.21 Select this option to enable a DTE to set up, answer, and clear calls over the installed front panel user interface (X.21 or V.35) using the call control procedures defined by ITU-T X.21.

V.25bis Select this option enable a DTE to set up, answer, and clear calls over the installed front panel user interface (X.21 or V.35) using the call control procedures defined by ITU-T V.25bis.

Backplane

Switched Select this option if the data source is a Wideband Port that will make switched calls via the BCAM-ISO. Configure the Wideband Port for the same data rate as the BCAM-ISO, and select the BCAM-ISO as its **Outgoing TE**.

Enable the Call Rate **Non Negotiable** option when TimeFrame, COMBO or HSDM ports are making calls via the BCAM-ISO.

Manual Select this if the data source is a Wideband Port that will be manually connected to the BCAM-ISO. Configure the Wideband Port for the same data rate as the BCAM-ISO.

Notes

*Manual connections from Wideband Ports to the BCAM-ISO are not automatically created. You should configure these using OmniVision's **Connection Manager**. See the **OmniVision User Manual** for details.*

*To manually connect the data source, the manual connection should be to the BCAM-ISO's Interface (Port) 1, and should be configured as **Internal**.*

*Select **Manual** for the BCAM-ISO connection to an E1 Module and an IRM.*

Called Party Number setup

Multiple Hotline Select this and enter a Hotline number to which all the channels are to be directed. You cannot select Multiple Hotline if you have set up individual channels.

From Each Line Clear this parameter if you entered a BCAM-ISO (lineset) number. If you wish to set an individual number for each line, accept the default (selected). Then select the line number and enter the required information for each line.

Line configuration

Line Number Enter the line number for the port. You must enter a line number if the line is to receive incoming calls directly. However, if the line is a member of a lineset or a Virtual Line, no number is required. The line number will be affected when a base number has been configured. See *Parameters*, page 2-5 for more information.

Subaddress Enter the subaddress if required. It is not mandatory to enter anything in this parameter.

Port ID Not applicable to BCAM-ISO.

Port Name Enter the Port Name if required. It is not mandatory to enter anything in this parameter.

ISDN Interface — Call Establishment Modes

The following table shows Call Establishment Modes default settings for each channel.

	Parameter	Default
Semipermanent (where available)	Enable	Cleared
	Label	—
Subrate Virtual Line member or manually connected	Enable	Cleared
	Label	—
Hotline	Enable	Cleared
	Number	—
	XSPC	Cleared
	Subaddress	—
Call Control	Auto Answer	Cleared
Signalling	Remote number	—
	End-to-End	Cleared

Changing Call Establishment Modes defaults

The settings for these parameters control the way in which calls are made. The Call Control and Signalling parameters should not be changed. However, you can set up the following call types:

- Semipermanent (where available)** Enter the semipermanent number in the Label parameter and select **Enable**. You should also clear the Hotline checkbox.
- Member of Subrate Virtual Line or manually connected** Select **Yes** if the line is a part of a Subrate Virtual Line or manually connected. It enables data to be transferred directly onto the B-channel without the call being set up. If you do not select this, the port does not transmit or receive data until a call is received or initiated by the port.
- Hotline** Enter the Hotline number for the channel and enter a subaddress, if required. Select **Enable**. You should also clear the Semipermanent checkbox.
- Select **XSPC** to enable a Switched Semipermanent Connection if required. You can only select this if a Hotline is enabled.

ISDN Interface — ISDN Line Controls

The following table shows the ISDN Line Controls defaults.

ISDN Line Controls

	Parameter	Default
AOC (where available)	None	Selected
	Continuous or End of Call	Cleared
Calling Line ID	CLIP	Cleared
	CLIR	Cleared
Numbering format	Line No	E.164
	Hotline	E.164
	Line and Hotline Subaddress	NSAP
CLIVE List Assignment	This line checks	—
	Available CLI/Number lists	—

Changing ISDN Line Controls defaults

AOC (where available) Select either **Continuous** or **End** to log the cost of each call from this line. You can view the cost details via OmniVision's Call Log. For further details, refer to the *OmniVision User Manual*.

This is only available if you subscribe to an AOC service.

Calling Line ID Do not select **CLIP** (CLI Presentation).
If you select **CLIR** (CLI Restriction), calls from this line do the opposite of the site default. For example, if your site default presents CLI for outgoing calls and you select **CLIR**, calls from this line do not present CLI to the called party.

Note *If you request AOC or CLIR for a PRA, but do not subscribe to them, the call attempt may be rejected by the ISDN.*

CLIVE List Assignment Select this option to apply a CLI/Number Checking list to this line. See page 3-49 of the *System Configuration* section for more information.

Numbering format Select the appropriate ITU-T (CCITT) standard for Line Number, Hotline Number, and Line and Hotline Subaddress formats.

Internal Control defaults.

Restriction None.
Outgoing TE Any.

Changing ISDN Internal Control defaults

Restriction Select **Disable** from the list to prevent calls being made to or from this line. Select **In Only** or **Out Only** to allow calls in the selected direction only.

Outgoing TE Select the required TE. If you select **Preferred** or **Exclusive**, you must then select the TE from the displayed list. This setting is effective on a line by line basis.

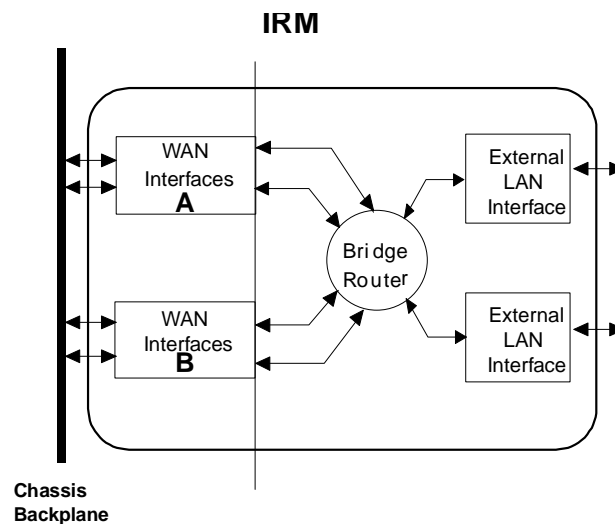
Integrated Router Module (IRM)

The Integrated Router Module (IRM) is a multiport, multiprotocol router designed to add Local Area Network (LAN) interconnection functionality. The IRM connects LANs to other remote networks via the Wide Area Network (WAN) facilities provided by the equipment.

The IRM comprises two conceptual parts:

- A Bridge/Router that supports the LAN interfaces and Bridging and Routing functions
- A line module that provides the WAN interfaces and calling functions.

It provides up to two physical LAN ports on the front panel and four WAN ports on the chassis backplane. This concept is illustrated below.



The IRM supports the following features:

- Fixed bandwidth
- On-demand calling
- Dynamic or static bandwidth management
- Automatic Backup and Restore
- Frame Relay WAN encapsulation.

Fixed bandwidth

A fixed amount of bandwidth is allocated to handle calls, regardless of traffic loads.

On-demand calling

A call is triggered when significant data is bound for an on-demand WAN port. When the call is no longer required, the call is cleared. Significant data is defined as frames routed or bridged to a single WAN port. These include Unicast frames generated by the IRM.

When no significant data (for a configured period) is bound for an on-demand WAN port, the Bridge/Router releases any active calls associated with that WAN port. While in this state, the Logical Port is not considered down and (pseudo-static) routes are maintained. The Bridge/Router is then able to route traffic to a Logical Port even though the Physical Port is disconnected. Spoofing is also provided so that host sessions are not lost.

On receipt of significant data bound for an on-demand WAN port, the Bridge/Router activates the WAN port. While this is occurring, any further traffic bound for the port is buffered until the port is activated or the buffer overflows.

When a Group A port is used in an on-demand situation, bandwidths of up to 1984kbit/s can be used over the on-demand network connections that **do not exhibit a differential delay** between bearer channels. In this case, you have the option of using either static or dynamic bandwidth management of the on-demand connection.

Note *The on-demand mode of the IRM cannot be used with the BCAM-ISO. You can use multi-linking to achieve on-demand mode speeds greater than 64 kbit/s and up to 256 kbit/s in networks that exhibit differential delay.*

Dynamic bandwidth management

Dynamic bandwidth management refers to the ability to provide automatically adjusted WAN bandwidth on the basis of preconfigured user traffic load requirements. This applies to **already** active Logical WAN connections. The mechanism does not clear or establish Logical WAN connections.

Static bandwidth management always uses the maximum configured speed (or as close as possible) as the active WAN bandwidth.

Both the Line Module and Bridge/Router Module have control over certain bandwidth management mechanism aspects.

Line Module Bandwidth Management

The Line Module dynamically manages bandwidth by adding and deleting channels of a WAN port. The bandwidth increments are therefore in bearer speed steps. Hence, bandwidth management within the Line Module domain relates only to Group A ports that **do not exhibit a differential delay** between bearer channels. It does not apply to Group B ports that only operate at a maximum of one channel.

When using Line Module dynamic bandwidth management, setting up WAN port connections result in the connection of a single channel at the bearer speed. The setups may originate from an XSPC, a user, or the Bridge/Router Module depending on how the IRM is configured. As traffic demand varies, the WAN bandwidth increases or decreases as the IRM at the calling party end connects and disconnects channels up to the maximum configured speed.

You can configure a utilisation and hold time for the up speed decision, and another utilisation and hold time for the down speed decision. If the IRM detects that the average utilisation over the hold period is above the threshold set for up speed, an additional channel is connected. Likewise for down speed, if the IRM detects that the average utilisation falls below the threshold set for down speed, a channel will be disconnected. The last channel of the WAN port is not disconnected. Either the transmitted or received data, whichever is the greater, is used for the utilisation calculation.

Where two Group A ports are Multilinked, and the dynamic management configuration on both ports are identical, the Line module will manage the bandwidth by evenly distributing the load across the pair.

When using Linesets, you can configure a trunk such that channels are dedicated for IRM traffic only. Thus you can configure committed minimum bandwidths for dynamic (and on demand) WAN ports.

The Line Module statically manages bandwidth by either:

- accepting any speed, up to the maximum configured, for incoming calls: or,
- if possible, setting up the maximum WAN speed for outgoing calls.

Bridge/Router Module bandwidth management

The Bridge/Router Module dynamically manages bandwidth by connecting and disconnecting Physical WAN ports (including Group B ports) that are multilinked within one Logical Port. The bandwidth increments are therefore in configured port WAN speed steps. This call mode is called Bandwidth on Demand.

As traffic demand varies, the WAN bandwidth increases and decreases by connecting and disconnecting WAN ports. IRMs at either end of a point-to-point link attempt to vary the Logical WAN bandwidth should they be configured to do so. The up-speed and down-speed utilisation thresholds can be configured with a common hold time. When these thresholds are out of limits for the common hold time, the IRM that detects the transmitted data utilisation connects and disconnects the WAN ports accordingly.

Dynamically managed bandwidths up to 256kbit/s are possible, without the use ISO-BCAMs, over networks that do exhibit a differential delay between 64kbit/s channels.

Backup and restore

In any calling mode the Bridge/Router Module may disconnect a WAN port connection if either:

- the Logical Port fails to come up after 40s following a call setup: or,
- if an active Logical Port goes down.

This disconnect constitutes a failure in the primary connection path for a period of 30 seconds. Within this period, any subsequent outgoing call is made using the Hotline number configured in the Line Module. Any call attempts after this period will first retry the primary path.

Frame Relay

Frame Relay is supported locally through digital interfaces, such as the QDLM and BCAM-ISO or remotely via trunk modules. It is only used with Group A WAN ports.

Notes

The IRM cannot run Frame Relay and PPP on different WAN ports. If PPP is enabled, Frame Relay will not operate.

An IRM port that is configured to run Frame Relay applies 'User-side' Frame Relay PVC procedures. The device to which the IRM port is connected needs to run 'Network-side' PVC procedures.

*If an IRM port is manually connected to a packet Port, the Packet Port must be configured to use **Flag Fill**.*

See the *IRM Router Configuration Manual* for more details.

Configuration

This section covers the basic configuration for the line module aspects of the IRM. To configure the IRM for Routing/Bridging, refer to the *IRM Router Configuration Manual*.

Configuration of the WAN ports is similar to any other line module. The WAN ports interconnect B-channels to a LAN User Interface. If multiple B-channels are configured via the WAN port, the settings of the first channel is normally applied to all other channels. However, if required, B-channels can be configured individually. See the *Parameters* section for details of line configuration.

Group configuration

The following parameters can be configured for both Group A and B. The groups differ only in their maximum speed capabilities — Group A up to 1984 kbit/s and Group B up to 64 kbit/s.

Configuration defaults

	Parameter	Default
B-Channel Rate	64 kbit/s	Selected
WAN Speed	On Demand Max	—
	Fixed User Rate	64k
Call Options	Auto Answer	Cleared
Semipermanent	SPC Enable	Cleared
	SPC Label	—
Hotline	Hotline	Cleared
	XSPC	Cleared
	Number set	—

Changing the configuration defaults

B-channel rate Only 64 kbit/s is currently available.

WAN speed The WAN speed specifies the speed at which the WAN port will be running for the Bridge/Router. Group A and B both provide the same sub speed capabilities that is, speeds less than 64kbit/s.

The sub- 64kbit/s speed capabilities include 48 kbit/s ITU-T (CCITT) framed and n x 8 kbit/s non-ITU-T (CCITT) framed speeds: 8, 16, 24, 32, 40, 48, 56 kbit/s.

The 48 kbit/s ITU-T (CCITT) WAN speed is generally used for Subrate interworking applications. A more common sub speed application involves non-ITU-T (CCITT) framed sub speeds that are used with Subrate Virtual Lines to include either compressed voice or other sub rate channels in one B-channel (see page 3-23 for more information on Subrate Virtual Lines).

Speeds greater than 64 kbit/s are incremented in multiples of 64 kbit/s. That is, one B-channel is required for each speed increment above 64kbit/s. For example, 1984 kbit/s requires the connection of 31 B-channels.

Note *The DTM can provide up to 31 full B-channels in one link. One IPM link can only provide up to 30 full B-channels.*

On Demand Max

This enables you to select a WAN speed from a set of valid speeds. The set is the same as that shown against the **Fixed User Rate**.

Static — Select this for on demand calls that are to be made at the selected speed. If the IRM fails to negotiate all of the calls required to achieve the selected speed, it accepts any calls that are successfully made.

Note *When using On-demand mode, enable **Auto Answer** at both ends of the link and use larger static bandwidths. This helps when both ends of a link require bandwidth at the same time.*

Slow Dynamic — Select this to enable calls on a single Group A WAN or two multi-linked Group A WANs to be made and cleared on the basis of traffic variation. In this mode, the IRM is slower to make and clear calls. The calling party controls the dynamic features. The slow dynamic default settings are as follows:

- Up Speed Utilisation = 75%
- Up Speed Hold Time = 12s
- Down Speed Utilisation = 30%
- Down Speed Hold Time = 60s

Pulse Dynamic — This setting has the same features as Slow Dynamic, but the IRM is faster in making and clearing calls. The pulse dynamic default settings are as follows:

- Up Speed Utilisation = 75%
- Up Speed Hold Time = 5s
- Down Speed Utilisation = 30%
- Down Speed Hold Time = 30s

Custom — This enables you to change the Threshold Utilisation and Hold Time for the Up Speed and Down Speed settings to define your own dynamic calling regime. The Custom Dynamic Setting dialog box shows the settings currently selected.

Notes *When using ISDN, you should set the parameters to delay the disconnection of active channels. This helps to minimise flag fall costs. Conversely, you should balance these costs against the cost of the timed rates when the call is active.*

When customising, be careful not to create call bouncing. This is when calls are made and cleared continuously.

Threshold utilisation is based on the greater value of either transmitted or received data.

Fixed User Rate

This shows the available WAN speed. Select this parameter to accept the shown rate. When this is selected, the Static, Slow dynamic, Pulse dynamic and Custom settings are not available.

Call options The only call option is **Auto Answer**.

When enabled, the IRM does not require the bridge/router to respond before it connects incoming calls. When disabled, the IRM does not automatically answer incoming calls but notifies the network that the bridge/router is to be alerted first. This ensures that the call is connected only if the bridge/router can use the call.

Semipermanent To set up a semipermanent connection from a WAN port, select **SPC Enable** and enter the semipermanent label issued by your service provider.

For WAN speeds higher than 64kbit/s, SPC activation is configured on an BCAM-ISO module and not the IRM. See page 4-222 for details on using the BCAM-ISO as an IRM utility.

Note *You should deactivate Hotline calls when SPCs are activated.*

Hotline To enable a WAN port to dial a Hotline number enter the **Hotline Number** and enable the **Hotline**. (Entering the **Subaddress**, if required, can be done using Edit List Configuration).

Note *You should disable SPC connections when Hotlines are activated.*

When **XSPC** is enabled, the device dials the Hotline number and continually redials if the connection fails. This is available only when Hotline is enabled.

In On-demand mode, the **Number Set** is only used when the line module must originate a call when:

- XSPC is enabled on the WAN
- a manual call is made from the Network manager
- in backup mode.

Notes *When in backup mode, the next call retry uses the **Hotline Number Set** as the backup number rather than the Bridge/Router's original calling party number.*

*If backup is not required, the **Hotline Number Set** must be the same as the Bridge/Router's calling party number.*

Outgoing TE Select the required TE for all the module's lines. If you select **Preferred** or **Exclusive**, you must then select the TE from the list of available TEs.

Port identification

A port is identified by its line number. Calls bound for an NT will be routed to an IRM WAN port when the line number is matched with the incoming called party number. Port line number configuration is achieved by assigning the appropriate IRM WAN port lines as members of the desired lineset or Virtual Line identifiers. See *Linesets* on page 3-17 and *Virtual Lines* on page 3-21 for more details.

Note *If a TE is present and a base number has been defined, the line number matched will be the incoming called party number - base number.*

As well as the line number, the following parameters can be configured on a line by line basis:

- ISDN subaddress line number suffix (optional)
- port ID label (for use in conjunction with the IPM CLI package)
- port name.

Backplane bandwidth allowance

The IRM uses backplane bandwidth dynamically. This means that backplane bandwidth is allocated as required. For example, an IRM that has all four WAN ports configured for 64kbit/s, is only allocated the four backplane timeslots that it needs.

Using the BCAM-ISO as an IRM utility

If an IRM Group A port uses a network that cannot provide a constant differential delay between B-channels, an ISO standard B-channel Aggregation Module (BCAM-ISO) is required. The BCAM-ISO performs the necessary channel sequence alignment to aggregate (referenced to time) the multiple B-channels. Other common terms used to describe this process are Inverse Multiplexing and ISDN Super Rate Adaption.

See page 4-207 for the BCAM-ISO configuration details.

When configuring the IRM and BCAM-ISO to operate together, observe the following points:

- Configure the IRM WAN speed first.
- Select **Manual Connect (Backplane)** in the BCAM-ISO configuration.
- After configuring the BCAM-ISO, ensure the status of the IRM WAN port is active. If the status is not active, configuration is not correct.
- BCAM-ISO Mode 1 has no overhead channel with the disadvantage that no data integrity checking takes place. If the data path is lost while the calls are active, the IRM cannot automatically disconnect calls. You would need to manually disconnect and reconnect the calls in this instance.
- BCAM-ISO Mode 3 provides data integrity checking but with the disadvantage of the need for an overhead channel. If the data path is lost while calls remain active, the BCAM-ISO can be configured to disconnect and re-establish the calls.
- The BCAM (AS 4064) cannot be used with the IRM.

Notes

- *Link connections that do not exhibit differential delay problems (for example, calls going through a DTM trunk or IPMs connected back to back) will not require the BCAM-ISO.*
 - *SPCs do not guarantee constant differential delay between channels. Therefore, when using SPC WAN speeds greater than 64kbit/s, the BCAM-ISO must be configured for SPCs and provide a backplane connection to the IRM.*
-

Signalling Access Module (SAM)

The Signalling Access Module (SAM) can perform the functions of an inband signalling node. An inband signalling network is created by interconnecting the inband signalling nodes of all devices in the network. The SAM is only required for Subrate Virtual Lines, and not TimeFrame VLs.

Note *The SAM is not supported by the Resource Manager (RM).*

Inband signalling provides an economical alternative to reliance on public ISDN D-channel user-to-user messaging.

Inband signalling nodes can be interconnected by any of the following types of communications links:

- packet-switched public data network via ISDN B-channel at 64 kbit/s
- 64 kbit/s Channel, via an ISDN B-channel, or via any timeslot of a G.704 system (except timeslot 0), or via an SDLM X.21 interface
- front panel X.21 DCE or DTE interface up to 64 kbit/s
- front panel synchronous V.24 DCE or DTE interface up to 38.4 kbit/s.

The number of communications links that can be connected to one device is limited only by the availability of physical connections, and other hardware limitations.

The SAM front panel can be configured as a DCE (Data Circuit Equipment) or DTE (Data Terminal Equipment) as defined by ITU-T (CCITT) recommendations. It is fitted with either an X.21, V.24 or V.35 interface on the front panel.

For further information on inband signalling, refer to page 3-43, *Inband signalling facility*.

For information on using the SAM for inband signalling see page 5-43 of the *Applications* section.

Configuration

Connecting multiple SAMs in a single node

We recommend that you connect channel B2 of the first SAM to channel B1 of the following SAM. This ensures that channel B1 of the first SAM is always connected to the Control Module inband channel, and B2 is connected to other SAMs in the chassis. The following shows the connection arrangement between a CM and three SAMs in the same node:

CM inband Ch --->B1-B2--->B1-B2--->B 1

The router table entries of the SAM can be configured as normal, but with the precaution that the table is constructed to direct those calls that do not match an entry to channel B2. This allows the unmatched calls to be routed to the remaining SAMs. The following table shows an example setup for three SAMs:

CM Router Table	93640098 0188 04-09	D1 D2 inband
SAM1	04111 042688 05-09	B3 B4 B2
SAM2	05123 055555 052-053 06-09	B3 B4 B8 B2
SAM3	08 09	B5 B2

Notes

The support of multiple SAMs in a single node may cause slight delays in the response of the inband signalling information.

*When two SAMs are used in the same chassis, incoming calls will fail if all lines on the first SAM module are busy. The incoming calls are **not** redirected to the second SAM. To avoid this, configure outgoing calls to start on the second SAM, leaving some ports free on the first SAM module for incoming calls.*

SAM Line Control

Edit Line There are no defaults on this setting.

Configure Select **Router tables**. This enables you to set up the inband signalling network for the module you have already selected.

Configure Inband Signalling Routing Table

By selecting **Add routes**, you are presented with **Configure Inband Signalling Routing Table**. This offers you the possibility of selecting Destination Address, Primary Routes, Secondary Routes, and Configured Addresses.

Each route is identified by a set of three numbers, together with a description. The first member of the set designates the module slot position, the second, the interface type and the third, the line or channel.

You should ensure that your primary route selection does not overlap with your secondary route selection, for a particular destination.

Inband Signalling Routing Table defaults

Default use D (or) B-channel	B-channel.
SAM channel idle holding time	30 secs.
Destination Address	None.
Primary Routes	None.
Secondary Routes	None.
Configured Addresses	None.

Changing Signalling Routing Table defaults

Before building a routing table, you must supply the Destination address and, if more than one address is to be routed, its range. To each address, or range of addresses, a primary route must be attached.

Default use D (or) B channel	These are relevant to the CM-SAM routing table configuration. If a called number is not listed in the routing table, the CM-SAM router makes the call directly to the destination using whichever default channel is selected.
SAM channel idle holding time	This has a range from 0 to 2550 secs. The default is 30 secs. When enabled, the channel is held open for the specified time even when no activity is detected.
Area code	Enter the Area code without the leading zero.
Transit call line number	A D-channel NMS call can be made into a router that has been programmed with this number. This is then forwarded on to a B-channel NMS call.
Destination Address	Insert the minimum destination number of the range in the Min/Only field. If there is only one destination number, enter it here. Enter the maximum number of the range in the Max field.
Primary Routes	After inserting a destination address, highlight the selected Primary Route from the list appearing in the window. Then select Add .
Secondary Routes	Highlight the selected Secondary Route from the list appearing in the window, if needed. Then select Add .
Configured Addresses	The window displays a list of configured addresses, using the Primary Route and Secondary Route settings you selected. To delete a Routing Table entry, highlight it in the Configured Addresses table, and press Delete . To eliminate the table, select Delete All .

ISDN Interface — Call Establishment Modes

The following table shows Call Establishment Modes default settings for each channel.

Call Establishment Modes defaults

	Parameter	Default
Semipermanent (where available)	Enable	Cleared
	Label	—
Subrate Virtual Line member or manually connected	Enable	Cleared
	Label	—
Hotline	Enable	Cleared
	Number	—
	XSPC	Cleared
	Subaddress	—
Call Control	Don't use for I/C manager calls	Cleared

Changing Call Establishment Modes defaults

The settings for these parameters control the way in which calls are made. You can set up the following call types:

Semipermanent (where available)	Enter the semipermanent number in the Label parameter and select Enable . You should also clear the Hotline checkbox.
Member of Subrate Virtual Line or manually connected	Select Yes if the line is a part of a Subrate Virtual Line or manually connected. Enter the number in the Label parameter. You should also clear the Hotline checkbox.
Hotline	Enter the Hotline number for the channel and enter a subaddress, if required. Select Enable . You should also clear the Semipermanent or Member of Subrate Virtual line... checkbox.
	Select XSPC to enable a Switched Semipermanent Connection if required. You can only select this if a Hotline is enabled.
Don't use for I/C manager calls	Select this to prevent the SAM channel being used for incoming OmniVision calls.

ISDN Internal Control

Restriction	Select None (default), In only, Out only or Disable.
Operation (Front panel only)	Select DCE or DCE.
Outgoing TE	Select the required TE. If you select Preferred or Exclusive , you must then select the TE from the displayed list. This setting affects all the module's lines.



Frame Switch Module (FSM)

The Frame Switch Module (FSM) is a Frame Relay switch with a built-in access router. It is available in two variants:

- FSM-15 — supports up to fifteen remote connections
- FSM-30 — supports up to thirty remote connections.

The FSM terminates remote devices to a LAN using the PPP protocol. It does this via an ethernet 10baseT port or to an external router via a Frame Relay port using X.21 or V.35 access. The Frame Relay port supports access through public networks or direct connect.

The FSM supports an external RADIUS Server for access control and accounting purposes. PAP and CHAP access control protocols are also supported.

Remote access devices can be:

- 64k ISDN service
- analog modems when used with a Digital Modem Module (DMM)
- 9600 bit/s GSM data.

The FSM is NMS-configurable using OmniVision. The router is configured directly from the front panel port or via a connected LAN.

Configuration

Call routing

Call Routing is configured in the same way as all other modules in the chassis. FSM lines can be placed in linesets or given individual line numbers. To configure linesets, see *Linesets* on page 3-17.

To configure individual line numbers select the module on the NMS front panel display.

Outgoing Calls

Outgoing calls can be made from FSM ports by configuring a Hotline number and enabling Hotline mode on the port. The call can be initiated using the NMS or by enabling XSPC mode on the FSM port.

Outgoing calls use the Bearer and LLC as configured for the port. Bearer should be set to **64K Unrestricted Digital** with no **Octet 5**. LLC should be set to **64K Unrestricted Digital** with **CCITT Rate Adaption** enabled if anything other than 64k is required. **Octet 5a** can then be programmed with the desired data rate and synchronous or asynchronous settings.

Note *In this release, modems cannot be used in outgoing calls as the DMM always has the modems in answer mode, and not originate mode.*

Manual connections

Manual connections can be made between two FSM ports or between an FSM port and another port such as that of a DLM or E1M. The connection is made using the NMS manual connection facility. In this release, manual connections are digital, that is, you cannot use modems.

The same rules apply for configuring manual connection LLC and Bearers as in *Outgoing calls* above. However, if you are making a manual connection between two FSM ports, you should programme both ports (rather than just the originator as is the case in outgoing calls). Both ports should have the same configuration settings.

FSM Line Control

FSM configuration

Port address This is a unique IP address that is assigned nominally to the LAN port, but is also used as the Router's own address for Telnet, TFTP and so on. The address is entered in the form 1.2.3.4.

The FSM does not support IP Addresses with an octet of zero (.0) – as RFC950 specifies this is an invalid address. For example, the FSM does not support the address 172.16.0.80 with subnet mask 255.255.255.0.

Route configuration The **Static Route** is entered as an IP address in the form 1.2.3.4. The default setting is 0.0.0.0 which indicates that there is no **Default Route**. If no route is entered the FSM will learn the route from another router in the network.

To configure a route, select the direction of the route, that is, **To Gateway** or **To Port** (PPP or Front panel). If using the gateway, enter an appropriate address. If using the PPP or Front panel port, enter a suitable port number.

You can configure further Static Routes by manually creating and editing a *USER.INI* file. The file, once created can be uploaded to the FSM using TFTP. Once the FSM has been restarted, the new static routes are enabled.

Enter the additional routing information in the IP Routing Section. If this does not exist, enter the following line in the *USER.INI* file:

```
[[User IP Routing = Section Start]]
```

The general syntax for a static route is:

```
target, gateway, mask, metric, port number
```

The details of this depend on whether the route is to an attached network (LAN), or to the Point-to-Point protocol (PPP) or Front panel Frame Relay port. If the route is to an attached network, set the gateway to the appropriate address and the port number to 255.

If the route is to one of the WAN links, set the gateway to 000.000.000.000 and set the port number to 1-30 for the respective PPP ports, or 31 for the Frame Relay front panel port.

Below are two example static routes:

Example 1

This routes all traffic to subnet 192.123.42.0 via gateway 192.123.43.254 with metric 1.

```
[[ User IP Routing = Section Start ]]
```

```
IP Static Route Table Entries = ;192.123.042.000, 192.123.043.254,
255.255.255.000, 01, 255
```

Notes *The line beginning 'IP Static Route' and ending ' , 255' should all be on one line.*

The semicolon (;) is positioned before the target address and all addresses are three digits with leading zeroes.

Example 2

The second routes all traffic to subnet 192.123.42.0 via PPP port 13 with metric 2:

```
[[ User IP Routing = Section Start ]]
```

```
IP Static Route Table Entries = ;192.123.042.000, 000.000.000.000,
255.255.255.000, 02, 013
```

Domain Name Server (DNS)

The DNS is not used by the FSM, but can be sent to remote dial-in users as part of the PPP negotiation. If the DNS is set to a value, other than 0.0.0.0, and a remote PPP requests a DNS, the entered address is supplied as the Primary DNS (PPP Option 129).

Note *This is not supported by all PPPs.*

PPP Idle timeout

This parameter sets an inactivity timeout after which the call is terminated. If the timeout is set to **Off**, or **On** with zero minutes, PPP sessions will continue indefinitely or until the user terminates the session. If the timeout is set to **On** with a setting of between 1 and 240 minutes, the session will terminate after that many minutes of inactivity. Inactivity is defined as no IP traffic.

IP Addressing and Routing

The WAN ports must be assigned an IP address. Each of the 31 WAN ports can run **Numbered** or **Unnumbered** links. Unnumbered is the default.

If using Numbered links, each WAN link requires its own subnet. The remote and local ends of the link are assigned addresses within that subnet, and when the port is used, the FSM creates a route for the entire subnet.

When unnumbered links are used, only the remote end of the link has an IP address. The remote link becomes a host route, using the FSM LAN port as a gateway address when the port is used. These addresses do not have to be on a unique subnet.

Select the line to be configured and make the following entries.

Line number

Enter the line number.

Typically, Bearer Capability and Low Layer Compatibility configuration is not required for the FSM. The FSM accepts a wide range of call types, and uses modems or different rate adaption schemes to match the incoming call.

Notes

If calls are being rejected due to incompatible Low Layer, disable LLC Checking for incoming calls.

*When operating over a DASS-2 network, **Telephony** must be enabled in the Higher Layer Compatibility for each FSM line being used.*

FSM port control

IP Address Enter the remote IP address. This should only be used when the remote device does not have an IP address configured, and is to be assigned an IP address by the FSM.

Note *The remote address can also be assigned from RADIUS. If you are going to use RADIUS, or the remote device already has an IP address, leave this field as 0.0.0.0.*

The FSM does not support IP Addresses with an octet of zero (.0) – as RFC950 specifies this is an invalid address. For example, the address 172.16.0.80 with subnet mask 255.255.255.0. is not supported.

To change from the default Unnumbered operation, choose PPP or Frame relay as required and select the **Setup** button. This provides access to **More port settings**.

IP

Unnumbered Link If this is selected, the Local Address field is automatically set to 0.0.0.0, and the Mask field to 255.255.255.0. These fields are disabled.

Numbered Link If you select this option, the Local Address should be set to an appropriate address. The Mask field defaults to 255.255.255.0, but can be changed to suit your requirements.

You should use the following **user.ini** configuration in the [[User IP Routing = Section Start]] section when configuring numbered WAN links:

Convert the WAN port IP address to a net route in the routing table by entering the following line in **user.ini**:

```
IP Port Remote Mask = 01, 255.255.255.0
```

where '01' is the port number and '255.255.255.0' is the mask of the WAN port.

Create a static route over a numbered WAN link, by adding the following lines to **user.ini**:

```
IP Port Unnumbered Link = 01, disabled
IP Static Route Table Entries =
;172.21.224.100,192.195.112.1,255.255.255.252,02,255
```

where the format for the first line is port, enabled/disabled and the second line is target, gateway, mask, metric, port.

Use Proxy ARP This can be used with unnumbered links. When selected, the front panel LAN port performs ARP (Address Resolution Protocol) on behalf of the WAN port. The Ethernet Address of the LAN port resolves the IP addresses of the LAN port and any WAN ports that have this proxy ARP enabled.

Enable RIP RIP can be enabled on a port by port basis. The default is off. This is recommended for WAN links.

RIP can be disabled globally by including the line `IP RIP = disabled` in the [[User IP Routing = Section Start]] section of the **user.ini** file.

Note *If a large number of routes are present in the network, and RIP is enabled, performance can be seriously degraded. This is because the routes will all be propagated down the WAN link every 30 secs.*

Multilink PPP

Multilink PPP is an extension to the PPP protocol that allows a number of PPP links to be combined to form effectively one larger link. The collection of PPP links is called a 'bundle'. Up to four **Lines** are supported in a bundle, providing a maximum data rate of 256kbit/s. A PPP bundle takes its IP addresses and routing information from the first port in the bundle. Bundles must consist of contiguous ports in the FSM.

A Multilink bundle is identified using the Multilink Endpoint Identifier. This is a unique identifier assigned to all the ports in the bundle. A number of different types of identifier are allowed. The FSM uses a 4 byte identifier made up of:

- for unnumbered operation – the port number of the start of the bundle followed by the last 3 bytes of the LAN IP address, or:
- for numbered operation – the local IP address of the bundle.

This is carried out by the FSM and does not require any configuration.

Once a bundle is created and one or more of its ports are active, IP packets are divided up and sent as fragments across the different members of the bundle and recreated at the destination.

Configure Multilink PPP

A port can be set to one of the following:

- Multilink (MP) Off
- MP Start of Bundle, or
- MP Bundle Member.

To configure a bundle, set the first (that is, the lowest port number of the bundle) to **MP Start of Bundle** and subsequent ports to **MP Bundle Member**. The bundle continues until a port is configured as **MP Start of Bundle** again or **MP Off**. Then configure the IP addressing on the **MP Start of Bundle** port.

Non-fragmented Multilink PPP packets are always sent down the member of the bundle which is marked as the 'Start of the Bundle'. This means that to get the most efficiency from the Multilink PPP link you should always make the highest speed link the 'Start of the Bundle'.

Note *If Multilink is being run across TimeFrame and a TDM link then the TDM link should always be made the 'Start of the Bundle'.*

Ports may be used for both Multilink and normal PPP operation. All ports in a bundle should have IP addresses and Authentication configured as desired. To receive a Multilink call, the caller must support the Multilink MRRU and Multilink Endpoint Discriminator options in their PPP. If a port receives a Multilink call it will operate as part of the Multilink bundle. If neither of the above options is supported, the port will revert to non-multilink operation for the duration of the call.

Note *When Multilink PPP configuration has been changed, the FSM must be restarted to incorporate the changes.*

Compression

The FSM supports two forms of compression:

- Stac LZS PPP Data Compression implementing the PPP Compression Control Protocol (CCP) and
- TCP header compression implementing Van Jacobson Compression (VJC).

Compression (CCP) If CCP is enabled on an FSM port, the FSM will attempt to negotiate compression with the remote user. The FSM only supports Stac LZS compression, but supports all the different modes such as LCB, CRC, sequence number and extended sequence number. This is compatible with Win95, Win98, WinNT, Internet Explorer Dialler and Cisco 750 series routers amongst others.

Internet Explorer Dialler enables CCP by default. Win95 needs 'Enable Software Compression' set in the Dialup Networking options.

Compression (VJC) If VJC is enabled on an FSM port, the FSM will attempt to negotiate the use of Van Jacobson TCP header compression on the link. If successful, the FSM will compress, and allow the reception of compressed headers on TCP traffic, for example, telnet, ftp and http.

Note *Where Compression is used in conjunction with Multilink PPP, all ports in a bundle should be set to the same compression options.*

In order to have compatibility with certain ISDN Terminal Adaptors, the FSM Address and Control Field Compression (ACFC), and Protocol Field Compression (PFC) are disabled. These compression fields are associated with PPP.

Frame Relay

Frame Relay DLCI See Frame Relay port on page 4-235 for more information.

Comms

Set the clock rate and DTE/DCE selection for the front panel port. **DCE** provides clock, **DTE** expects it. The clock rate should be a multiple of 64 between 64 and 2048 (Kbit/s).

RADIUS

A RADIUS server can be used for both Authentication and Accounting. A local RADIUS-style operation is also available and can be accessed using a local password file. To configure RADIUS for remote or local operation, carry out the following:

Local Select this to enable local operation. This uses the password file stored in the FSM. If the password file is encrypted, check the **Local password file is encrypted** checkbox, and type the secret in the **Secret** field.

Remote Select this to enable the remote RADIUS Server. Type its IP address in the **IP Address** field. Type the secret in the **Secret** field.

RADIUS accounting enabled Select this to enable RADIUS accounting. This sends the appropriate RADIUS accounting start and stop messages to the server. If local operation is selected, the start and stop messages are printed from the monitor port.

Login and authentication

The FSM supports bi-directional authentication for **PAP** and **CHAP**. This means that the FSM can **Authenticate** and be authenticated at the same time.

For **Login**, authentication is asymmetrical, that is, it challenges remote users. A remote user must respond accordingly. The FSM does not respond to a challenge from the remote user.

Authentication for each of the PPP ports is configured with each line configuration. The following settings are all valid:

All Authentication off (none selected)

No authentication is performed. PPP starts as soon as the port is used.

Login on, PAP and CHAP off

Only Login authentication is performed. When the port is used, the FSM sends a configured text message followed by the configured login prompt. The FSM collects the password and name and presents them for authentication (local or RADIUS). When these are accepted, PPP is started.

Note *User names less than four characters long are ignored. This allows some types of menu-expecting scripts that send a number followed by a carriage return, followed by a name and another carriage return, to operate correctly.*

Login off, PAP or CHAP on

Only PAP or CHAP authentication is performed. When the port is used, the FSM starts PPP. LCP negotiation of PAP and CHAP authentication is requested. If both of these are refused, the call is cleared. If one is accepted, the FSM will authenticate the remote user between LCP and NCP negotiation phases using whichever is accepted. If the authentication is responded to correctly, the call proceeds, otherwise it is terminated.

Login on, PAP or CHAP on

This is the most flexible configuration. PAP, CHAP or Login are acceptable. The FSM sends the login prompt as soon as the port is used. If PPP is detected, rather than a name, the FSM starts PPP and attempts to negotiate PAP or CHAP authentication.

Note *When PAP or CHAP is selected, the FSM accepts either, but will try the selected one first.*

RADIUS attributes

The FSM sends the following RADIUS attributes:

Authentication Request Using PAP or Login

- User-Name
- User-Password
- NAS-IP-Address
- NAS-Port
- Service-Type
- Framed-Protocol
- Framed-IP-Address
- Framed-IP-Netmask
- Framed-Compression
- State
- Called-Station-Id
- Calling-Station-Id

Authentication Request Using CHAP or Login

- User-Name
- CHAP-Password
- NAS-IP-Address
- NAS-Port
- Service-Type
- Framed-Protocol
- Framed-IP-Address
- Framed-IP-Netmask
- Framed-Compression
- State
- Called-Station-Id
- Calling-Station-Id
- CHAP-Challenge

Accounting Start

- User-Name
- NAS-IP-Address
- NAS-Port
- Called-Station-Id
- Calling-Station-Id
- Acct-Status-Type
- Acct-Session-Id

Accounting Stop

- User-Name
- NAS-IP-Address
- NAS-Port
- Called-Station-Id
- Calling-Station-Id
- Acct-Status-Type
- Acct-Delay-Time
- Acct-Input-Octets
- Acct-Output-Octets
- Acct-Session-Id
- Acct-Authentic
- Acct-Session-Time
- Acct-Input-Packets
- Acct-Output-Packets
- Acct-Terminate-Cause

Frame Relay port

You can configure one DLCI for the Frame Relay port. Enter the DLCI in Hexadecimal in the range 30 to 916. The clock rate and the DTE/DCE selection for the front panel port can also be configured. DCE provides clock, and DTE expects to receive clock. Clock rate should be between 64 and 2048 kbit/s in multiples of 64 kbit/s.

FSM files

The FSM has a flash memory file system with a capacity of 3.84 MB. The system uses compressed binary files. The FSM requires the following files for operation:

- **fsmvxxx.bin** — Main operational binary
- **fsm.ini** — Main default configuration file.

Note *These files and filenames should not be altered.*

The following files are optional:

- **user.ini** — Extra configuration items (in fsm.ini file format)
- **prelog.txt** — displayed before login, when ascii login is configured. This is a plain text file, up to 1K. The default is **Frame Switch Module Version x.xx**
- **login.txt** — displayed as a login prompt. the default is **login:**. The file should be on one line with no carriage return or line feed at the end
- **password.txt** — Displays as the password prompt. The default is **password:**. The file should be on one line with no carriage return or line feed at the end
- **postlog.txt** — displays after a successful login. This a plain text file up to 1k in size. The default is **Login Accepted, now starting PPP**
- **password** — This is a data file used for local password verification if the remote RADIUS is not enabled. It has the format:

```
name:password
name2:password2
```

The password can be plain text or MD5 encoded. This is configured from the RADIUS Settings dialog. If encryption is selected, the passwords are stored in this file as 16 character MD5 digest strings of the actual passwords using the RADIUS Secret.

USER.INI file

This section details the parameters than can be added to the FSM USER.INI file to change the configuration of the IP Router and PPP. Note that after changing any configuration in USER.INI, you must reset the FSM for the changes to take effect.

We advise that the USER.INI parameters listed below are changed by personnel with a good understanding of networking principles. You can use the default parameters in most cases.

Note *Lines starting with // in USER.INI are treated as comment lines and are not processed by the FSM.*

Static Routes

If you require more than one static route, the extra static routes must be added in the USER.INI file.

The syntax for static routes is:

```
IP Static Route Table Entries = ;Target, Gateway, Mask, Metric, Port
Number
```

There are two methods of entering a static route:

- Set the Gateway parameter to 000.000.000.000 and specify the correct Port Number where 000 is the Ethernet Port on the FSM, 001 to 030 are the thirty PPP WAN ports and 031 is the X.21 or V.35 port on the FSM Front Panel.
- Set the Port Number to 255 and specify a remote gateway connected to a local port.

In both cases the Metric should be the correct number of hops to the target.

The static routes must be added to the following section in USER.INI.

```
[[ User IP Routing = Section Start ]]
```

Example:

```
[[ User IP Routing = Section Start ]]
```

```
// Create a Static Route to network 172.21.224.0 using the X.21/V.35
Front
```

```
// Panel Port with a Metric of 1
```

```
IP Static Route Table Entries = ;172.21.224.0, 000.000.000.000,
255.255.255.000, 01, 031
```

```
// Create a Static Route to network 172.21.16.0 using the Ethernet Port
```

```
// with a Metric of 2
```

```
IP Static Route Table Entries = ;172.21.16.0, 000.000.000.000,
255.255.255.000, 02, 000
```

```
// Create a Static Route to network 172.21.100.0 using PPP WAN Port 1
```

```
// with a Metric of 2
```

```
IP Static Route Table Entries = ;172.21.100.0, 000.000.000.000,
255.255.255.000, 02, 001
```

```
// Create a Static Route to network 172.21.200.0 via Gateway
192.195.118.10
```

```
// with a Metric of 3
```

```
IP Static Route Table Entries = ;172.21.16.0, 192.195.118.010,
255.255.255.000, 03, 255
```

IP Port Address Masks

To set the IP Port Address Mask (subnet mask) of each IP port you must add the following line to the USER.INI file.

```
IP Port Address Mask = Port Number, Subnet Mask
```

Where the Port Number is 00 for the FSM Ethernet Port, 01 to 30 for the thirty PPP WAN Ports and 31 for the FSM X.21/V.35 Front Panel Port.

Each port has a default mask of 255.255.255.0 if no line is specified for that port in USER.INI.

The mask must be added to the following section in USER.INI.

```
[[ User IP Routing = Section Start ]]
```

Example

```
[[ User IP Routing = Section Start ]]

// Subnet Mask of 255.255.255.0 for the Ethernet Port

IP Port Address Mask = 00,255.255.255.000

// Subnet Mask of 255.255.255.252 for PPP WAN Port 1

IP Port Address Mask = 01,255.255.255.252

// Subnet Mask of 255.255.255.0 for the X.21 / V.35 Port

IP Port Address Mask = 31,255.255.255.000
```

Globally Enabling and Disabling RIP

You can globally enable or disable the Routing Information Protocol (RIP) in the FSM by using the following line in USER.INI. Note that RIP is enabled by default if the IP RIP = line is not present in USER.INI.

```
IP RIP = disabled
```

or

```
IP RIP = enabled
```

Add this line to the following section in USER.INI.

```
[[ User IP Routing = Section Start ]]
```

Example

```
[[ User IP Routing = Section Start ]]

IP RIP = disabled
```


or

```
[[ User IP Routing = Section Start ]]
```

```
IP RIP = enabled
```

PPP Maximum Number of Configuration Requests

You can add this parameter to the USER.INI to change the number of PPP configuration requests that the FSM sends before realising that there is no device responding on the other end. The call is terminated at this point. If you do not enter this line the number of attempts is defaulted to 86400.

```
PPP Maximum Number Of Configuration Requests = Attempts
```

Where `Attempts` has a valid range of 10 to 86400.

Add this line to the following section in USER.INI.

```
[[ User PPP Device Driver = Section Start ]]
```

Example:

```
[[ User PPP Device Driver = Section Start ]]
```

```
// Set the number of configuration requests to 33600
```

```
PPP Maximum Number Of Configuration Requests = 33600
```

Multilink PPP Parameters

You can use the following lines to change the parameters associated with Multilink PPP.

```
MP Small Packet Length = Length
```

This line sets the minimum packet length (in bytes) at which packet fragmentation over Multilink occurs. Valid values are 64 to 512. If you do not specify this line it uses a default value of 128.

```
MP Stall Timeout = Timeout
```

This line sets the time (in seconds) before the Multilink PPP declares a packet fragment lost and discards all received fragments of the packet. Valid values are 3 to 10. If you do not specify this line a default value of 5 seconds is used.

These parameters are added to the following section in USER.INI.

```
[[ User Multilink PPP Driver = Section Start ]]
```

Example

```
[[ User Multilink PPP Driver = Section Start ]]  
  
// Set Fragment Packet Length to 128  
  
MP Small Packet Length = 128  
  
// Set Packet Fragment Discard time to 5 seconds  
  
MP Stall Timeout = 5
```

PPP IPCP Parameters

The FSM supports the configuration of a number of PPP IPCP parameters in the USER.INI file. These are **Primary DNS** (Domain Name Server), **Secondary DNS**, **Primary NBNS** (NetBIOS Name Server) and **Secondary NBNS**.

You can configure the Primary DNS parameter using the **FSM Line Control** dialog in OmniVision. The rest of the parameters are configured in USER.INI as described below. If you do not add these parameters to USER.INI no **NBNS** or **Secondary DNS** is provided during PPP negotiation.

```
PPP Port Remote Option = Port Number,IPCP,Secondary  
DNS,131,Auto,Negotiation Not Required,Not Negotiable,IP,Secondary DNS  
IP Address
```

```
PPP Port Remote Option = Port Number,IPCP,Primary  
NBNS,130,Auto,Negotiation Not Required,Not Negotiable,IP,Primary NBNS  
IP Address
```

```
PPP Port Remote Option = Port Number,IPCP,Secondary  
NBNS,132,Auto,Negotiation Not Required,Not Negotiable,IP,Secondary  
NBNS IP Address
```

Where Port Number is 00 for PPP WAN Port 1 through to 29 for PPP WAN Port 30.

Note *You must enter the IP Addresses in this field in reverse. For example, an address of 192.195.112.13 is entered as 013.112.195.192.*

Add these parameters to the following section in USER.INI.

```
[[ User PPP Device Driver = Section Start ]]
```

Example

```
[[ User PPP Device Driver = Section Start ]]

// Set the secondary DNS as 192.195.112.8 for PPP WAN Port 1

PPP Port Remote Option = 00,IPCP,Secondary DNS,131,Auto,Negotiation
Not Required,Not Negotiable,IP,008.112.195.192

// Set the primary NBNS as 192.195.112.2 for PPP WAN Port 5

PPP Port Remote Option = 04,IPCP,Primary NBNS,130,Auto,Negotiation Not
Required,Not Negotiable,IP,002.112.195.192

// Set the secondary NBNS as 192.195.112.13 for PPP WAN Port 5

PPP Port Remote Option = 04,IPCP,Secondary NBNS,132,Auto,Negotiation
Not Required,Not Negotiable,IP,013.112.195.192
```

Telnet and Virtual Terminal

The FSM has a built in Telnet Server that allows various control and monitoring functions to be performed remotely. It can be used with most Telnet clients. To access FSM Telnet connect from a Telnet client program on a local or remote IP addressable system to the FSM's main IP address. Once connected a **login:** prompt displays. Type the name and password.

The default name and password for an unconfigured system is either:

- root for the name and Jtec for the password, or
- admin with no password <enter>.

These can be changed from within the Telnet operation.

If the name and password is accepted, the FSM:telnet> prompt is displayed. The FSM also supports the Virtual Terminal which can be accessed from OmniVision by clicking the right mouse button over the selected FSM and choosing **Virtual Terminal**.

The Virtual Terminal does not prompt for a user name and password, and goes directly to the FSM:telnet> prompt.

The following commands can be entered at the `FSM:telnet>` prompt using either the Telnet client or the Virtual Terminal:

- `dir` — Show directory of flash file system
- `ls` — Same as `dir`
- `ren oldfile.ext newfile.ext`— Rename a file in the flash file system
- `mv oldfile.ext newfile.ext` — Same as `ren`
- `del filename.ext` — Delete a file
- `rm filename.ext` — Same as `del`
- `type filename.ext` — Display a (text) file (does not support files larger than 8 Kbytes)
- `cat filename.ext` — Same as `type`
- `run [xxx.gz]`— With no following parameters, this shows the currently active program. When used with a filename (typically.gz) this sets the program that will be run next time the system reboots
- `reboot [now]`— This will cause (if the now is also given) a system reboot. This is delayed for 20 seconds to give time for the FSM to clear all calls and log them with the Accounting server
- `user [add name password]` and `user [delname]`— With no parameters, this lists currently authorised names and passwords for the Telnet server. To add a new one, use `user add name password`. To delete an unwanted one, use `user delete name`. If all users are deleted, the system will default back to the default ones. The default one can be deleted using `user delete root`
- `time` — Set or display system time. Use `time` to display it or `time hh:mm:ss (24 hour clock)` to set it
- `date` — Set or display system date. Use `date` to display it or `date dd:mm:yy (NOT mm:dd:yy)` to set it
- `who` — Lists who is currently logged in on what port since what time
- `history n` — Shows history of logins. Lists port number username, login time, duration and numbers of octets sent and received
`history 10` shows the first 10 entries in order of logoff time.
`history -42` shows the last 42 entries in reverse order of logoff time. The FSM will store the 1000 most recent sessions. This is not intended for accounting purposes and is volatile (i.e. lost when rebooted)
- `hostname` — enables you to set the hostname of the FSM.
- `naughty n` — Shows list of bad login attempts (i.e. those that fail authentication.) Lists name, port and time. The FSM will store the 1000 most recent attempts. This is volatile
- `notrace` — Turns off all tracing and reporting. This can be used after diagnostic traces have been set and ensures that performance will not be impacted by unnecessary tracing that has been inadvertently left turned on
- `router` — See Router below
- `ping <port number> <IP address to ping>` — `<port number>` is 0 for the FSM LAN port, and 1 to 30 for the FSM WAN ports. `<IP Address>` is entered in dotted decimal format
- `tftp` — See TFTP below
- `help` — List these commands
- `?` — Same as `help`

- `logout` — Leave Telnet
- `exit` — Same as `logout`.

Router

Enter Router mode by typing `router` at the `FSM:telnet>` prompt. This displays the `FSM:router>` prompt. Enter Router commands using a natural language type interface. A limited set is shown here:

<code>show stack ip route</code>	Shows routing table
<code>cache</code>	Shows routing cache
<code>protstats</code>	Shows protocol stats
<code>portstats</code>	Shows port stats
<code>arptable</code>	Shows the arp resolution table.

Diagnostics

A useful feature of the FSM is the ability to selectively trace protocol activity on any or all of the LAN or WAN ports. The following commands are available from the `FSM:router>` prompt:

```
set port ppp nn trace off
set port lan 1 trace short
set port fr1 trace medium
long
decode
```

where:

- `nn` is the PPP Port number from 1 to 30.
- `off` means disable.
- `short` means a single line display for each packet sent or received in the following format:

```
TX PPP P1 L26
```

This means that the FSM has sent a packet out of Port 1, its length is 26 bytes.

- `medium` displays the same first line and also adds the first 16 bytes in hexadecimal. the format is:

```
RX PPP P1 L26
FF 03 C0 21 01 03 00 08
```

- `long` displays the same first line but the entire packet is in hexadecimal.
- `decode` is available for PPP only. This performs a full decode of the PPP packet. For example:

```
RX PPP P1 L26
LCP Configure-Request ID=3 Len=12
Authentication-Protocol Len=5 C22305
Magic-Number Len=6 01020304
and so on...
```

Note *Be careful to turn any traces off again or performance will be degraded.*

Router commands may also be used from the front panel monitor port. This is done by prefixing the command with `ur`. For instance, to show the IP routing table, type:

```
ur show stack ip route
```

To quit Router operation and return to `FSM:telnet>`, type `exit`.

TFTP

TFTP is used to upload and download program files and configuration files. TFTP is a client application which requires a TFTP server to be available with the required files accessible. The TFTP server must be IP addressable by the FSM, and its IP address must be known. DNS lookup is not supported.

To enter TFTP mode type `tftp [aaa.bbb.ccc.ddd]` at the `FSM:telnet>` prompt. This displays the `FSM:tftp>` prompt. The following commands can be used:

- `connect aaa.bbb.ccc.ddd` — Use TFTP server at IP address `aaa.bbb.ccc.ddd`
- `get filename.ext` — Upload the file `filename.txt` from the TFTP server and store it in the FSM flash file system
- `put filename.ext` — Download the file `filename.txt` from the FSM flash file system to the TFTP server
- `quit` — Leave TFTP and return to `FSM:telnet>`.

The FSM TFTP supports a special file called `dir.txt` which can be put onto the server. This contains the current directory listing of the flash file system.

Note that all transfers are binary, ASCII transfers are not supported.

Monitor

The FSM's internal monitor can be accessed from telnet. To enter monitor mode, type `monitor` at the `FSM:telnet>` prompt. This displays the `FSM:monitor>` prompt. Using the monitor is not recommended for normal operation and is beyond the scope of this document.

- `exit` — Leave monitor and return to the `FSM:telnet>` prompt.

Finger

The FSM has a built in Finger server or daemon. It monitors the FSM LAN port address, in the same way as Telnet. It supports two types of Finger query, current online users and details of logged on users.

To find current users, enter a line similar to the following:

```
finger @fsm.something.com.au
```

This returns a list of currently online users and which port they are on. For example:

```
Username          Port
=====
chuck              1
wally              2
```

To get details of a particular user, enter:

```
finger chuck@fsm.something.com.au
```

This returns details of that user if they are logged on, for example:

```
Username  Port  Logged in at          Speed  Sent(b)  Rcvd(b)
=====
chuck     1     10:35:45, 21/01/98    33600  6034203  6050914
```

or a message to say that user is not online, for example:

```
chuck is not currently online
```

Note *The Finger function always reports the user speed as 33.6 kbit/s, regardless of the actual speed.*

Digital Modem Module (DMM)

The Digital Modem Module functions as a multi-channel, central site modem. This section refers to the following DMM models:

- DMM-15 providing up to fifteen 33.6K modem channels
- DMM-8, providing up to eight 33.6K modem channels
- DMM-56K-16, providing up to sixteen 56K modem channels
- DMM-56K-8, providing up to eight 56K modem channels.

Note *DMM-15 and DMM-8 do not support V.fast modems.*

The DMM-56K module supports V.90 (56 kbit/s) modem standards and legacy modulations such as K56Flex and V.34. For more information on V.90 technology, refer to *Appendix R*.

The DMM accepts an input from an ISDN B-channel in the form of A-law or μ -law encoded PCM data. This input originates from a remote user terminal connected to the modem. Incoming data is decoded and demodulated back to the original digital signal as an asynchronous serial stream at 57.6 kbit/s. The serial stream is then rate-adapted into a 64 kbit/s timeslot and connected to a Frame Switch Module (FSM).

In the opposite direction, the DMM receives data from the Frame Switch Module in a 64 kbit/s timeslot. The input signal is converted into a 57.6 kbit/s asynchronous stream and is then modulated and encoded into A-law or μ -law PCM data.

Note *The PCM data format used to communicate with the client modem is A-law by default. The PCM format configured in the RM overrides the default on a per call basis.*

The module provides a remote access capability only when used in conjunction with the Frame Switch Module (FSM).

Configuration

There is no configuration required for the Digital Modem Module. A DMM is automatically selected when it is required for a call. The DMM always starts training at the highest possible speed and falls back to lower speeds to compensate for the speed of the remote modem or adverse line conditions.

Applications

Introduction

This section suggests ways of applying the powerful features of the this product range to meet your needs.

The following application examples are divided into specific groups, each covering features detailed elsewhere in the manual. We suggest that you read the appropriate sections of the manual before attempting to use these examples.

The subject areas covered are as follows:

- TimeFrame Virtual Lines on page 5-3
- Subrate Virtual Lines on page 5-24
- Microplex switching on page 5-27
- Virtual eXchange on page 5-30
- Interworking including the CallBack feature on page 5-34
- Inband signalling on page 5-43
- Low Delay CELP Module (LDCM) and Multi Function Compression Module (MFCM) on page 5-47
- Digital Trunk Module (DTM) on page 5-52
- E1M-DPNSS on page 5-55
- E1M-QSIG on page 5-66
- Analog line modules on page 5-78
- Virtual eXtension on page 5-86
- Integrated Router Module (IRM) on page 5-90
- Dial Gateway (FSM and DMM) on page 5-99
- Channel Bank application (T1M, E1M-2 and EMM) on page 5-104.

TimeFrame Virtual Line applications

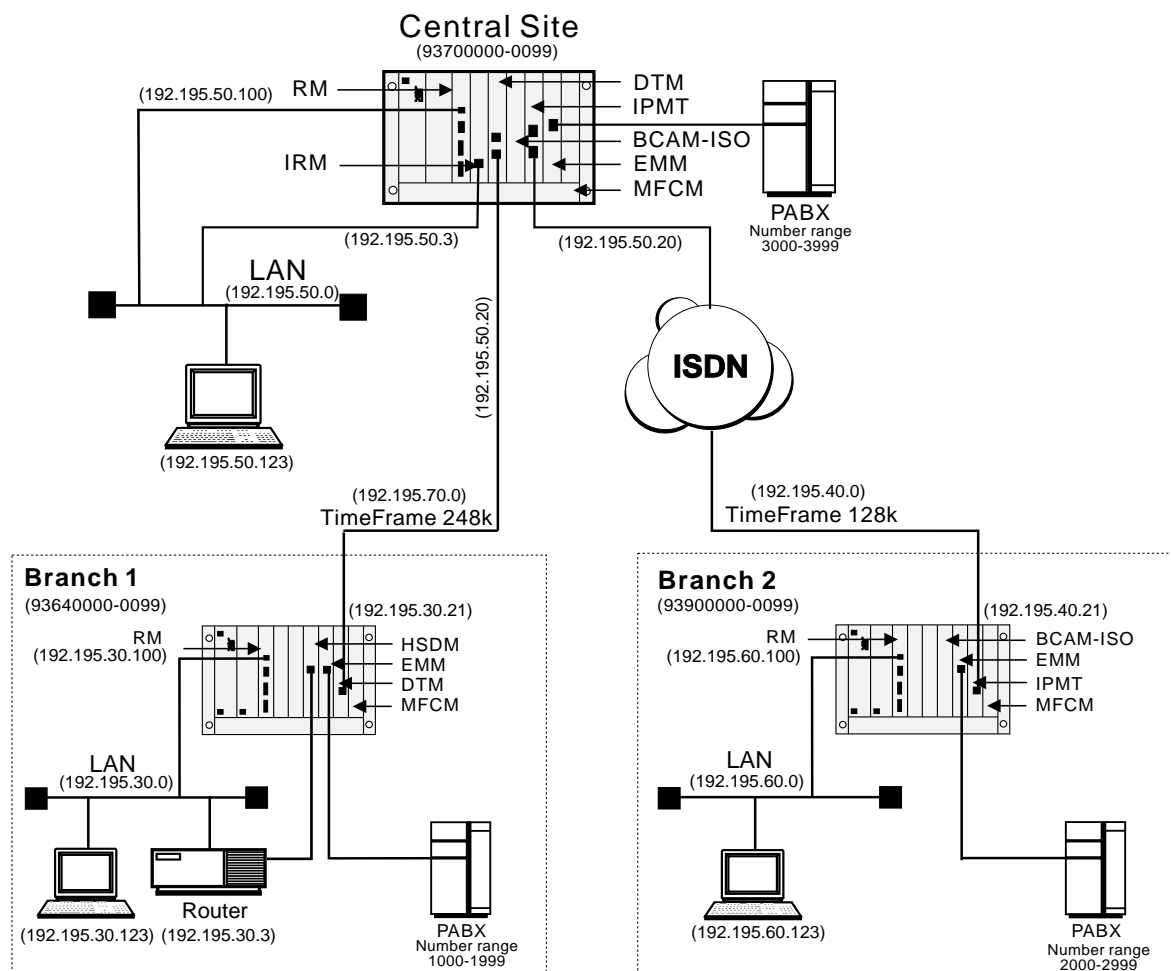
This section describes three typical TimeFrame Virtual Line applications. Example 1 shows a network of three nodes operating in a Wideband mode — one node acts as the Central node and the other two as Branch Office nodes. Example 2 shows two four slot chassis in a simple point-to-point network. Example 3 details the setting up of a Multilink TimeFrame Network.

Example 1 - Three node Wideband network

In this setup, all three nodes are connected to a PABX via an EMM to provide a voice service. Similarly each node is connected to a LAN to provide a data service. The network routing is set up as follows:

- **Central Site** — the IRM acts as the Central Site Router and is manually connected to Packet Port 2.
- **Branch 1** — uses an external router to perform routing and is connected to an X.21 port on the HSDM.
- **Branch 2** — uses the RM's IP Router.

The connection between the Central Site and Branch 1 is via a DTM, while Branch 2 and the Central Site are connected via IPMTs and BCAM-ISOs.



The following sections detail the configuration of each node using OmniVision.

General node configuration

The following table shows typical entries for management numbers, IP management addresses and so on.

Node	OmV Management No	VX Number Range		IP Management Address	Subnet Mask
		Low	High		
Branch 1	93640098	93640000	93640099	192.195.30.100	255.255.255.0
Branch 2	93900098	93900000	93900099	192.195.60.100	255.255.255.0
Central Site	93700098	93700000	93700099	192.195.50.100	255.255.255.0

DTM configuration

The following table shows the settings for the DTM modules for Branch 1 and the Central Site:

Node	Port	Trunk speed	SubB-channel speed
Branch 1	1	256k	56k
Central Site	1	256k	56k

Lineset configuration

The following table shows the entries to set up Linesets:

Node	Number Range		Members	Comments
	Low	High		
Branch 1	1000	1999	EMM Ports	PABX extensions
	2000	3999	TimeFrame Resources	Remote PABX extensions
	93700000	93700099	DTM Port 1 SubB &B	Central Site
Branch 2	2000	2999	EMM Ports	PABX extensions
	1000 3000	1999 3999	TimeFrame Resources	Remote PABX extensions
	93900070	93900070	BCAM-ISO Ch 1&2 TimeFrame VLs	Bearer from Central
Central Site	3000	3999	EMM Ports	PABX extensions
	1000	2999	TimeFrame Resources	Remote PABX extensions
	93640000	93640099	DTM Port 1 SubB & B	Branch Office 1
	93700091	93700091	BCAM-ISO Ch 1&2 TimeFrame Resources	Bearer to Branch 2

Setting up Number Checking

The following table details the settings for Number Checking:

Node	No. to check	Number range		Modification	Apply to...
		Low	High		
Branch 1	Called	2000	3999	Replace Called with 93700080	Lineset 2000-3999
Branch 2	Called	1000	1999	Replace Called with 93700091	Lineset 1000-1999
	Called	3000	3999	Replace Called with 93700091	Lineset 3000-3999
Central Site	Called	1000	1999	Replace Called with 93640091	Lineset 1000-2999
	Called	2000	2999	Replace Called with 93900070	Lineset 1000-2999

Configuring the BCAM-ISO

The following table shows the configuration for the BCAM-ISOs at Branch 2 and Central Site.

Node	Call Rate	Mode	Backplane
Branch 2	128 kbit/s	1	Switched
Central Site	128 kbit/s	1	Switched

Configuring the TimeFrame Virtual Line

The following tables show the TimeFrame VL settings. A TimeFrame VL is created in the Virtual Lines Configuration.

Branch 1

Node	Name	TimeFrame Trunk Name	Line No.	Max Port speed	Hotline enable	XSPC enable	Auto Answer
Branch 1	B1Cent	trunk1	93640091	256k	No	No	Yes
			Hotline No.	Inactivity Timeout	Outgoing TE	Min. Data Bandwidth	Don't Increment Hotline No.
			93700080	Disabled	Any	32kbit/s	Disabled

Branch 2

Node	Name	TimeFrame Trunk Name	Line No.	Max Port speed	Hotline enable	XSPC enable	Auto Answer
Branch 2	B2Cent	trunk1	-	128k	No	No	Yes
			Hotline No.	Inactivity Timeout	Outgoing TE	Min. Data Bandwidth	Don't Increment Hotline No.
			93700091	Disabled	Exclusive BCAM-ISO	24kbit/s	Enabled

Central Site

Node	Name	TimeFrame Trunk Name	Line No.	Max Port speed	Hotline enable	XSPC enable	Auto Answer
Central Site	CentB1	trunk1	93700080	256k	Yes	Yes	No
	CentB2	trunk 2	-	128k	Yes	Yes	No
			Hotline No.	Inactivity Timeout	Outgoing TE	Min. Data Bandwidth	Don't Increment Hotline No.
	CentB1	trunk1	93640091	Disabled	Any	32 kbit/s	Disabled
	CentB2	trunk 2	93900070	Disabled	Exclusive BCAM-ISO	24kbit/s	Enabled

Note *TimeFrame Virtual Line Lines are members of Linesets. No Line numbers are required.*

Configuring the Timeframe Resources

The following table shows the number of TimeFrame Resources allocated at each site. The TimeFrame Resources are configured in the Virtual Line configuration.

Node	TimeFrame Resources
Branch 1	27
Branch 2	13
Central Site	40

Configuring the Data

The external router at Branch 1 is connected to an X.21 port on an HSDM. It is configured to encapsulate all IP data in Frame Relay over the X.21 connection. Frame Relay encapsulation must be RFC1490 for the external router. The X.21 port is configured to run at 248 kbit/s.

Refer to your Router documentation for configuration of the external router.

The network IP addresses of Branch 1, Central and Branch 2 are 192.195.30.0, 192.195.50.0 and 192.195.60.0 respectively. The WAN link between Central and Branch 2 is assigned to be 192.195.40.0.

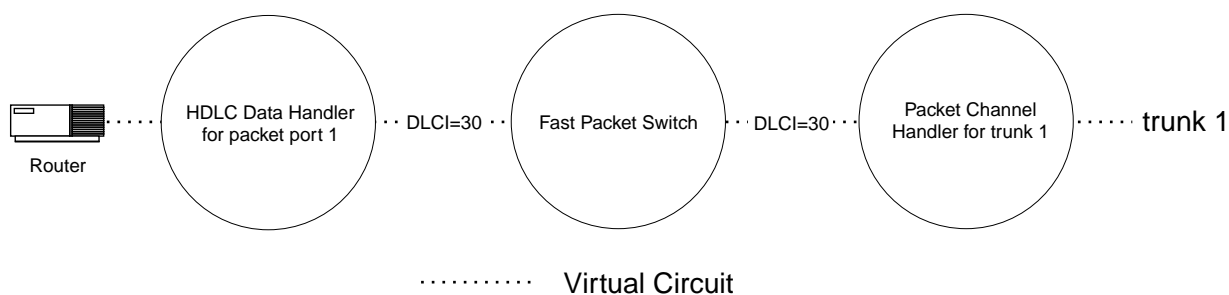
The network addresses for the WAN link between Central and Branch 1 should be configured in accordance with the instructions in the external router user manual. The diagram at the start of this section shows the IP address of each network device.

The external router at Branch 1 is configured to use DLCI=30, while the IRM at Central uses DLCI= 30 and 31 for the connections to Branch 1 and Branch 2 respectively. The IRM at Central is responsible for routing IP traffic originated from its LAN to Branch 1 and Branch 2. The data on the link between Branch 1 and Central is compressed, whilst data between Branch 2 and Central is uncompressed.

The following sections discuss the configuration of each node in detail.

Branch 1

A virtual circuit of the external router is connected to one of the Packet Ports in the HDLC Data Handler. A Packet Port can then be connected to any virtual circuit on the TimeFrame Virtual Line via the Fast Packet Switch. In this example, the external router's virtual circuit (DLCI=30) is connected to Packet Port 1 (pktport1), which is switched to trunk 1 by the Fast Packet Switch. The following diagram shows the virtual circuit connection between the external router and a TimeFrame trunk.



The HDLC Data Handler must be configured to pass Frame Relay frames from the router's virtual circuit (DLCI=30) to the Fast Packet Switch without modification. This can be achieved by setting the Packet Port's virtual circuit data type to Frame Relay (**fr**).

The data port to which the external router is connected must be manually connected to Packet Port 1 before data can be passed to the HDLC Data Handler. Using OmniVision select configuration for the port to which the external router is connected.

Once this is saved, create a manual connection between the data port and the line on the RM called **Packet Port 1**.

Use OmniVision's **Data Handling** feature to configure the Virtual Circuit for Branch 1. The following table summarises the settings:

Virtual Circuit	Interface	DLCI	Priority Data Type		
			High	Medium	Low
pktport1vc1	pktport1	don't care	fr	–	–
trunk1vc3	trunk1	30	all	–	–

The next table shows the Frame Relay Route configuration settings

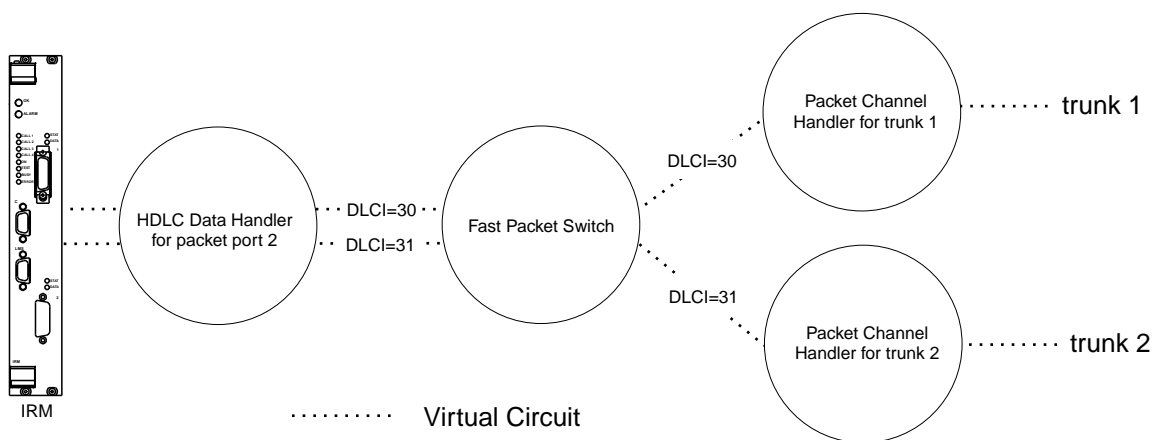
Entry	Port 1 ID	Port 1 DLCI	Port 2 ID	Port 2 DLCI	Data Compression	Data Type
1	pktport1	30	trunk1	30	Yes	all

The Packet Port does not use the configured DLCI value if its data type is set to **fr** (Frame Relay). This is because all **fr** frames originating from the external router are forwarded by the Packet Port to the Packet Switch without encapsulation or modification.

The DLCI values for the external router and trunk1 do not have to be identical. The DLCI selected for trunk1 must be unique on the trunk and must not be used for any other traffic than that originating from the external router. For example, the RM internal router must not be configured to use a DLCI value of 30 on trunk1 in addition to the external router.

Central site

Virtual circuits of the IRM at Central are assumed to be connected to pktport2. To switch these virtual circuits to their corresponding destination (i.e. trunks), the Fast Packet Switch must be set up to switch frames originating from **pktport2 DLCI 30** to **trunk1 DLCI 30**. Furthermore, frames sent from **pktport2 DLCI 31** should be switched to **trunk2 DLCI 31**. The following diagram illustrates the switching of these virtual circuits.



The data port to which the IRM is connected must be manually connected to Packet Port 2.

Use OmniVision's **Data Handling** feature to set up the Virtual Circuit and Frame Relay Route. The following table summarises the Virtual Circuit settings:

Virtual Circuit	Interface	DLCI	Priority Data Type		
			High	Medium	Low
irm30	pktport2	30	fr	–	–
irm31	pktport2	31	fr	–	–
trunk1vc3	trunk1	30	all	–	–
trunk2vc3	trunk2	31	all	–	–

The next table shows the Frame Relay Route configuration settings:

Entry	Port 1 ID	Port 1 DLCI	Port 2 ID	Port 2 DLCI	Data Compression	Data Type
1	pktport2	30	trunk1	30	Yes	all
2	pktport2	31	trunk2	31	No	all

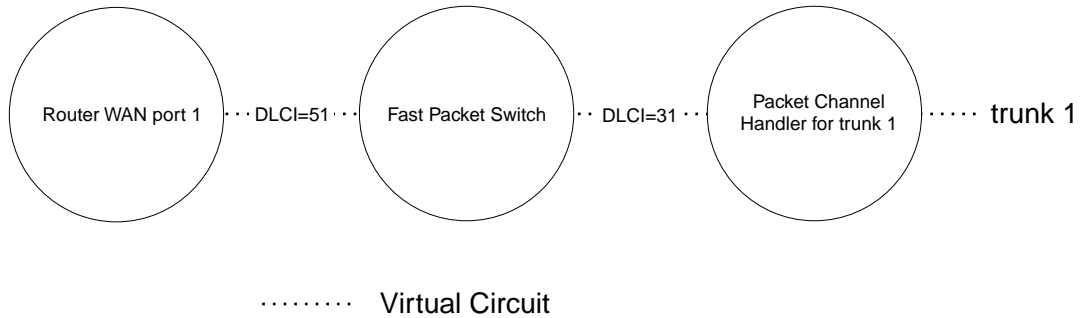
By default, the IRM enables user-side Frame Relay Permanent Virtual Circuit (PVC) management procedures on an IRM Frame Relay port. The RM Packet port to which the IRM port is connected (packet port 2) needs to perform network-side Frame Relay PVC management procedures.

Since the IRM uses ANSI Frame Relay PVC management procedures, packet port 2 needs to be configured to T1.617a. The following table summarises the Frame Relay PVC management parameters for packet port 2.

Configuration parameter	Value
Enable interface	Yes
Enable PVC management procedures	Yes
Protocol	T1.617a
Mode	Network
Error threshold	3
Monitored Events counter	4
Polling verification interval	15
Inactivity timeout	0

Branch 2

Branch 2 uses the RM's Internal Router for LAN/WAN routing. The virtual circuit connection between the Internal Router and a TimeFrame trunk is shown in the following diagram. Branch 2 is configured to use internal router WAN port 1 to route IP data to Central.



The RM's Internal IP Router has been pre-configured to use **DLCI 51 to DLCI 58**, which corresponds to WAN port 1 to 8 respectively. Traffic from a WAN port is subsequently switched to a virtual circuit on the appropriate trunk via the Fast Packet Switch. As a convention, WAN port x is switched to trunk x. For example, WAN port 1 should be switched to trunk 1.

Use OmniVision's **Data Handling** feature to configure the Virtual Circuit for Branch 2. The following table summarises the settings:

Virtual Circuit	Interface	DLCI	Priority Data Type		
			High	Medium	Low
routervc1	router	51	all	–	–
trunk1vc3	trunk1	31	all	–	–

The next table shows the Frame Relay Route configuration settings:

Entry	Port 1 ID	Port 1 DLCI	Port 2 ID	Port 2 DLCI	Data Compression	Data Type
1	router	51	trunk1	31	No	all

Configuration summary

The following tables summarise the configuration details of Branch 1, Branch 2 and the Central site.

Branch 1

RM LAN configuration

Port	Local IP address	Local subnet mask	ARP	RIP	Metric	MTU
LAN	192.195.30.100	255.255.255.0	Y	Y	1	1500

Virtual circuit configuration

Virtual circuit	Interface	DLCI	High priority Data type	Med priority Data type	Low priority Data type
pktport1vc1	pktport1	don't care	fr	—	—
trunk1vc3	trunk1	30	all	—	—

Frame Relay route configuration

Entry	Port1 ID	Port1 DLCI	Port2 ID	Port2 DLCI	Data comp.	Data type
1	pktport1	30	trunk1	30	Y	all

HSDM port 1 settings

Port Name	Speed
HSDM port 1	248kbit/s

Packet port 1 settings

Port Name	Speed	Parameter	Value
Packet port 1	248kbit/s	Enable interface	Yes
		Enable PVC management procedures	No
		Inactivity timeout	0

Manual connections

From	To
HSDM port 1:1	Packet Port 1:1
HSDM port 1:2	Packet Port 1:2
HSDM port 1:3	Packet Port 1:3
HSDM port 1:4	Packet Port 1:4

Branch 2

RM LAN/WAN configuration

Port	Local IP address	Local subnet mask	ARP	RIP	Metric	MTU	Un-numbered link	Remote IP add	Remote subnet mask
LAN	192.195.60.100	255.255.255.0	Y	Y	1	1500	—	—	—
WAN1	192.195.40.21	255.255.255.0	Y	Y	1	1522	Nn	—	—

Virtual circuit configuration

Virtual circuit	Interface	DLCI	High priority Data type	Med priority Data type	Low priority Data type
routerc1	router	51	all	—	—
trunk1vc3	trunk1	31	all	—	—

Frame Relay route configuration

Entry	Port1 ID	Port1 DLCI	Port2 ID	Port2 DLCI	Data comp.	Data type
1	router	51	trunk1	31	N	all

Central

RM LAN configuration

Port	Local IP address	Local subnet mask	ARP	RIP	Metric	MTU
LAN	192.195.50.100	255.255.255.0	Y	Y	1	1500

Packet port 2 settings

Port Name	Speed	Line Fill	Parameter	Value
Packet port 2	384kbit/s	Flag	Enable interface	Yes
			Enable PVC management procedures	No
			Protocol	T1.617a
			Mode	Network
			Error Threshold	3
			Monitored events counter	4
			Polling verification interval	15
			Inactivity timeout	0

Virtual circuit configuration

Virtual circuit	Interface	DLCI	High priority Data type	Med priority Data type	Low priority Data type
irm30	pktport2	30	fr	—	—
irm31	pktport2	31	fr		
trunk1vc3	trunk1	30	all	—	—
trunk2vc3	trunk2	31	all	—	—

Frame Relay route configuration

Entry	Port1 ID	Port1 DLCI	Port2 ID	Port2 DLCI	Data comp.	Data type
1	pktport2	30	trunk1	30	Y	all
2	pktport2	31	trunk2	31	N	all

IRM configuration

Port name	Speed
WAN A Port (0)	384kbit/s

Logical port	Physical port	IP Address	IP Address mask	IP Port routing	RIP	Management type	Channel
1	1	192.195.50.3	255.255.255.0	Enabled	Enabled	—	—
13	3	192.195.70.20	255.255.255.0	Enabled	Enabled	ansiT1-617-D	30
14	3	192.195.40.20	255.255.255.0	Enabled	Enabled	ansiT1-617-D	31

Manual connections

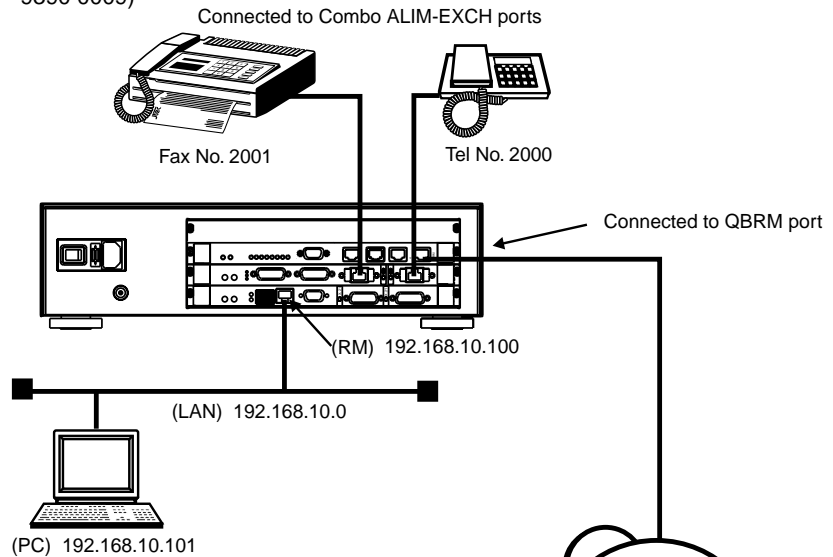
From	To
IRM WAN A-1 line 0	Packet port 2:1
IRM WAN A-1 line 1	Packet port 2:2
IRM WAN A-1 line 2	Packet port 2:3
IRM WAN A-1 line 3	Packet port 2:4
IRM WAN A-1 line 4	Packet port 2:5
IRM WAN A-1 line 5	Packet port 2:6

Example 2 — Point-to-point network

This network shows a simple point-to-point network using two four slot chassis connected to a Basic Rate ISDN service. Each node uses the internal IP router within the Resource Manager and has a telephone and a Fax connected to the ALIM-EXCH ports on a COMBO module.

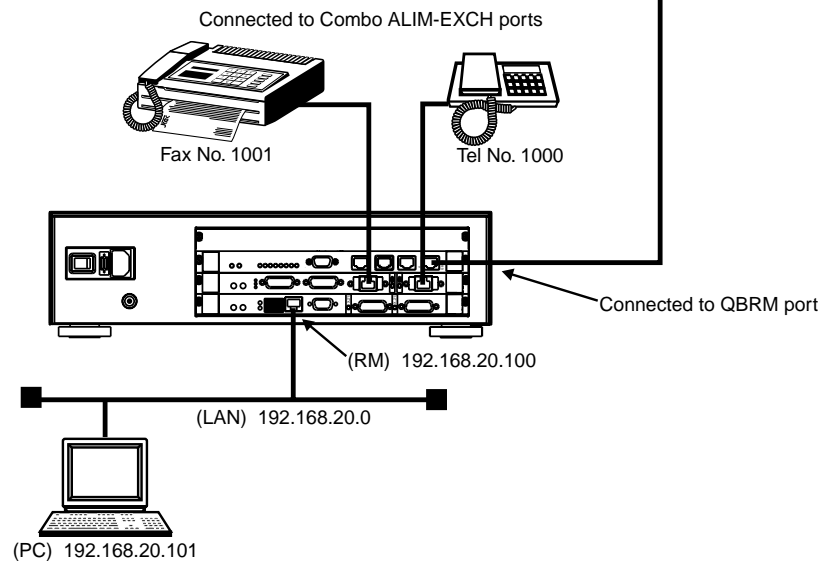
City A

(9390 0000 - 9390 0009)



City B

(9364 000 - 9364 0009)



The City A site uses XSPCs to establish the connection to City B. The following section details the configuration of each node using OmniVision and the RM Telnet Server.

Node configuration

Node	Omv Manager Number	VX Number Range		IP Management Address	Subnet Mask
		Low	High		
City A	9390 0009	9390 0000	9390 0009	192.168.10.100	255.255.255.0
City B	9364 0009	9364 0000	9364 0009	192.168.20.100	255.255.255.0

Lineset configuration

Node	Number Range		Members	Comments
	Low	High		
City A	2000	2000	ALIM Port 1	Phone extension
	2001	2001	ALIM Port 2	Fax extension
	1000	1001	TimeFrame resources	Remote extensions
	9390 0001	9390 0001	TimeFrame Virtual Line	Bearer from City B
City B	1000	1000	ALIM Port 1	Phone extension
	1001	1001	ALIM Port 2	Fax extension
	2000	2001	TimeFrame resources	Remote extensions
	9364 0001	9364 0001	TimeFrame Virtual Line	Bearer from City A

Number checking

Node	Number to check	Number Range		Modification	Apply to...
		Low	High		
City A	Called	1000	1001	Replace called with 9364 0001	Lineset 1000-1001
City B	Called	2000	2001	Replace called with 9390 0001	Lineset 2000-2001

TimeFrame Virtual Line

Node	Name	TimeFrame trunk name	Hotline No.	Max Port Speed	Hotline enable	XSPC enable	Auto Answer	Minimum Data Bandwidth	Inactivity Timer
City A	City B	trunk1	9364 0001	64 kbit/s	Yes	Yes	No	48 kbit/s	Disabled
City B	City A	trunk1	9390 0001	64 kbit/s	No	No	Yes	48 kbit/s	Disabled

TimeFrame Resources

Node	TimeFrame Resources
City A	2
City B	2

Configuring Data

The default settings for this scenario are sufficient to enable the LAN data connection to be established. The defaults are as follows:

- IP Routing
- Unnumbered link
- RIP: Enabled
- DLCI: 100 on trunk1 at both ends.

The only configuration that needs to be entered is the FR route configuration as shown below. You can also, optionally, enable or disable compression.

To alter these settings use the **Data Handler** feature in OmniVision.

Configuration summary

City A

LAN/WAN configuration

Port	Local IP Address	Local Subnet Mask	ARP	RIP	Metric	MTU	Un-numbered line	Remote IP Addr.	Remote Subnet Mask
LAN	192.168.10.100	255.255.255.0	Yes	Yes	1	1500	—	—	—
WAN1	0.0.0.0	255.255.255.0	Yes	Yes	1	1500	Yes	0.0.0.0	255.255.255.0

Virtual Circuit configuration

Virtual Circuit	Interface	DLCI	High priority data type	Medium priority data type	Low priority data type
routerc1	router	51	ip	—	—
trunk1vc1	trunk1	100	all	—	—

FR route configuration

Entry	Port 1 ID	Port 1 DLCI	Port 2 ID	Port 2 DLCI	Data compression	Data type
1	router	51	trunk1	100	No	ip

City B**LAN/WAN configuration**

Port	Local IP Address	Local Subnet Mask	ARP	RIP	Metric	MTU	Un-numbered line	Remote IP Addr.	Remote Subnet Mask
LAN	192.168.20.100	255.255.255.0	Yes	Yes	1	1500	—	—	—
WAN1	0.0.0.0	255.255.255.0	Yes	Yes	1	1500	Yes	0.0.0.0	255.255.255.0

Virtual Circuit configuration

Virtual Circuit	Interface	DLCI	High priority data type	Medium priority data type	Low priority data type
routervc1	router	51	ip	—	—
trunk1vc1	trunk1	100	all	—	—

FR route configuration

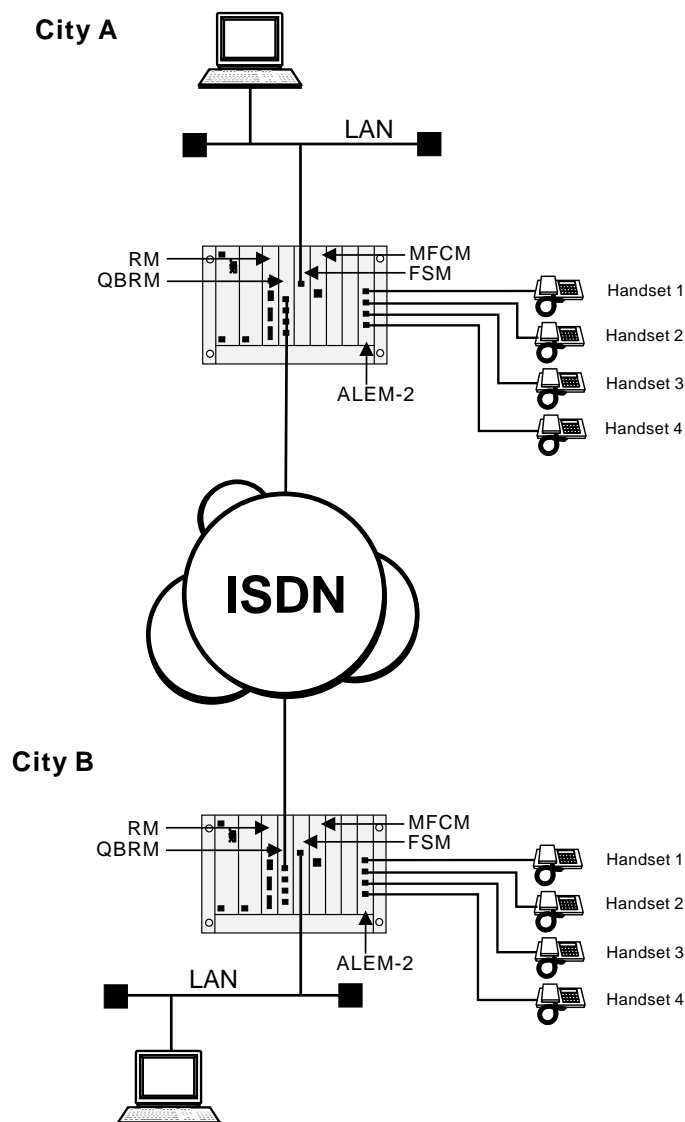
Entry	Port 1 ID	Port 1 DLCI	Port 2 ID	Port 2 DLCI	Data compression	Data type
1	router	51	trunk1	100	No	ip

Example 3 — Multilink TimeFrame Network

If your Router supports Multilink, you can use TimeFrame Virtual Lines in the Multilink network.

The following example shows a typical point-to-point network over ISDN. Here, a Frame Switch Module (FSM) is used to Multilink two TimeFrame Virtual Lines to provide an IP Network across a WAN bandwidth of 128 kbit/s. Voice services are provided by telephones connected to ALEM-2 modules.

The IP Router in the RM is not used in this example, as it does not support Multilink. In this case, the FSM Router is used, and connects to TimeFrame via two 64 kbit/s Packet Ports. The following diagram shows the network.



Optimising data throughput

To achieve the most efficient data throughput, the voice calls in this network are distributed evenly across both trunks. This is necessary because of the way in which the Multilink algorithm operates in most routers. Generally, the router expects the trunks, over which it is forming a Multilink bundle, to be at the same data rate – this is when Multilink is most efficient.

When there is a large difference in trunk bandwidth, Multilink is not as efficient because it wastes the excess bandwidth. For example, if a Multilink has two trunks at 64 kbit/s and 32 kbit/s respectively, the effective bandwidth is 64 kbit/s. This bandwidth is formed from 32 kbit/s from the first trunk (with 32 kbit/s excess wasted) and 32 kbit/s of the second trunk. If this was split into two trunks of 48 kbit/s, the effective bandwidth is 96 kbit/s, because Multilink would fully utilise both trunks and no bandwidth is lost.

The example below describes how you can configure the J5000 to provide Multilink with two equal data rate trunks. This is achieved by alternating the compressed voice circuits onto two different TimeFrame Virtual Lines. The configuration involves using a combination of Interwork Units (IWUs) and CLI/Number Checking to direct odd-numbered calls down TimeFrame VL 1, and even-numbered calls down TimeFrame VL 2.

VX Configuration

Node	Omv Manager Number	VX Number Range	
		Low	High
City A	9390 0009	9390 0000	9390 0009
City B	9364 0009	9364 0000	9364 0009

TimeFrame Virtual Line Configuration

Node	Name	Trunk name	Line No.	Hotline No.	Hotline enable	XSPC enable	Auto Answer	Port Speed	Inactivity Timeout
City A	CityB_1	trunk1	93900001	93640001	Yes	No	No	64 kbit/s	30 secs
	CityB_2	trunk2	93900002	93640002	Yes	No	No	64 kbit/s	30 secs
City B	CityA_1	trunk1	93640001	93900001	Yes	No	No	64 kbit/s	30 secs
	CityA_2	trunk2	93640002	93900002	Yes	No	No	64 kbit/s	30 secs

Note There are four **TimeFrame Resources** enabled in **TimeFrame Capacity** on each node.

Interwork Configuration

Node	Number of Interworks	Configuration
City A	1	Default configuration used for each IWU
City B	1	Default configuration used for each IWU

Lineset configuration

Node	Number Range		Members	Comments
	Low	High		
City A	1000	1003	1-1L, 1-2L, 1-3L, 1-4L	Four Left Hand Interwork Units
	2000	2003	ALEM-2 Ports 1,2,3,4	Four ALEM-2 Ports
	3000	3000	TimeFrame Resources	All four TimeFrame Resources (trunk1)
	4000	4000	TimeFrame Resources	All four TimeFrame Resources (trunk2)
City B	1000	1003	ALEM-2 Ports 1,2,3,4	Four ALEM-2 Ports
	2000	2003	1-1L, 1-2L, 1-3L, 1-4L	Four Left Hand Interwork Units
	3000	3000	TimeFrame Resources	All four TimeFrame Resources (trunk1)
	4000	4000	TimeFrame Resources	All four TimeFrame Resources (trunk2)

CLI/Number Checking

Configuration

Node	List Name	Number to Check	Apply when line is...	Modification
City A	citybtf1	Called Party No.	Call Destination	Replace Called Party No. with 93640001
	citybtf2	Called Party No.	Call Destination	Replace Called Party No. with 93640002
	prefix1	Called Party No.	Call Source	Prefix Called Party No. with 3000
	prefix2	Called Party No.	Call Source	Prefix Called Party No. with 4000
	strip	Called Party No.	Call Source	Remove first 4 digits of Called Party No.
City B	cityatf1	Called Party No.	Call Destination	Replace Called Party No. with 93900001
	cityatf2	Called Party No.	Call Destination	Replace Called Party No. with 93900002
	prefix1	Called Party No.	Call Source	Prefix Called Party No. with 3000
	prefix2	Called Party No.	Call Source	Prefix Called Party No. with 4000
	strip	Called Party No.	Call Source	Remove first 4 digits of Called Party No.

Apply to...

Node	List Name	Apply to...	Number Range		Comments
			Low	High	
City A	citybtf1	All TF Resources	3000	3000	Switches all calls starting with 3000 onto trunk1
	citybtf2	All TF Resources	4000	4000	Switches all calls starting with 4000 onto trunk2
	prefix1	1-1R, 1-3 R	1000	1003	Applied to all odd-numbered RH IWU involved in call
	prefix2	1-2R, 1-4R	1000	1003	Applied to all even-numbered RH IWU involved in call
	strip	All TF Resources	3000	4000	Strip 3000 and 4000 prefixes from received calls
City B	cityatf1	All TF Resources	3000	3000	Switches all calls starting with 3000 onto trunk1
	cityatf2	All TF Resources	4000	4000	Switches all calls starting with 4000 onto trunk2
	prefix1	1-1R, 1-3 R	2000	2003	Applied to all odd-numbered RH IWU involved in call
	prefix2	1-2R, 1-4R	2000	2003	Applied to all even-numbered RH IWU involved in call
	strip	All TF Resources	3000	4000	Strip 3000 and 4000 prefixes from received calls

Packet Ports

Node	Packet Port Number	Speed	Comment
City A	Packet Port 1	64kbit/s	All other configuration is default
	Packet Port 2	64kbit/s	All other configuration is default
City B	Packet Port 1	64kbit/s	All other configuration is default
	Packet Port 2	64kbit/s	All other configuration is default

FSM Configuration

Node	Port Address	Routing Configuration
City A	192.195.30.100	Default Route to PPP Port 1
City B	192.195.50.100	Default Route to PPP Port 1

Node	WAN Port No.	IP Address	Authentication	PPP Settings
City A	1	192.195.40.1	None	Numbered Link to 192.195.40.2 / MP Start of Bundle
	2	192.195.40.101	None	Numbered Link to 192.195.40.102 / MP Bundle Member
City B	1	192.195.40.2	None	Numbered Link to 192.195.40.1 / MP Start of Bundle
	2	192.195.40.102	None	Numbered Link to 192.195.40.101 / MP Bundle Member

Note For Multilink to take effect, the FSMs must be reset after making these configuration changes.

Manual Connections

Node	Manual Connections
City A	FSM Line 1 to Packet Port 1 FSM Line 2 to Packet Port 2
City B	FSM Line 1 to Packet Port 1 FSM Line 2 to Packet Port 2

Virtual Circuit Configuration

Node	Interface	Name	DLCI	High Priority Data	Med. Priority Data	Low Priority Data
City A	Packet Port 1	pktport1vc1	100	7e	—	—
City A	Packet Port 2	pktport2vc2	100	7e	—	—
City B	Packet Port 1	pktport1vc1	100	7e	—	—
City B	Packet Port 2	pktport2vc2	100	7e	—	—

Frame Relay Route Configuration

Node	Entry	Port 1 ID	Port 1 DLCI	Port 2 ID	Port 2 DLCI	Data Compression	Data Type
City A	1	pktport1	100	trunk1	100	No	all
	2	pktport2	100	trunk2	100	No	all
City B	1	pktport1	100	trunk1	100	No	all
	2	pktport2	100	trunk2	100	No	all

Voice operation

First call

The following steps summarise how the first voice call from City A reaches an ALEM-2 port in City B. Refer to the diagram on page 5-18.

- 1 City A ALEM-2 receives **1000** from handset.
- 2 Call Handling matches this to **LHS Interwork Unit 1**.
- 3 Number Checking prefixes **1000** with **3000** to give a number **30001000**.
- 4 Call Handling in City A matches this onto a **TimeFrame Resource** on **trunk 1**.
- 5 Call Handling in City B receives the call to **30001000** from City A.
- 6 Number Checking strips the first 4 digits from the number, leaving **1000**.

- 7 Call Handling in City B matches this onto the first available ALEM-2 port.
- 8 ALEM-2 port rings the attached handset.

Second call

The following steps summarise how the second voice call from City A reaches an ALEM-2 port in City B. Refer to the diagram on page 5-18.

- 1 City A ALEM-2 receives **1001** from handset.
- 2 Call Handling matches this to **LHS Interwork Unit 2**.
- 3 Number Checking prefixes **1001** with **4000** to give a number **40001001**.
- 4 Call Handling in City A matches this onto a **TimeFrame Resource** on **trunk 2**.
- 5 Call Handling in City B receives the call to **40001001** from City A.
- 6 Number Checking strips the first 4 digits from the number, leaving **1001**.
- 7 Call Handling in City B matches this onto the next available ALEM-2 port.
- 8 ALEM-2 port rings the attached handset.

When these two calls have been made, both TimeFrame Virtual Line 1 and TimeFrame Virtual Line 2 have a data bandwidth of 56 Kbit/s, which is maximising the data throughput for PPP.

Subrate Virtual Line applications

This section provides examples of the settings required for Access Controller (AC) and Virtual eXchange (VX) equipment to enable Virtual Line (VL) operation.

Setting up VLs in an Access Controller

When adding the various line types to a Virtual Line in a J1000 AC, use the following settings for the Call Establishment Modes of the Virtual Line and the relevant module.

Virtual Line Type	Virtual Line Setting	Module setting	
		ALEM/ALPM etc.	DLMs and B-SAM
SPC/Manual Connection operation	<input checked="" type="checkbox"/> SPC Label required for real SPCs	<input checked="" type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E National Significant Remote Number	<input checked="" type="checkbox"/> SPC
XSPC operation	<input type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E <input checked="" type="checkbox"/> Hotline (both ends) <input checked="" type="checkbox"/> XSPC (one end only) CM-SAM is optional	<input type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E	<input type="checkbox"/> SPC
On Demand operation	<input type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E <input checked="" type="checkbox"/> Hotline (both ends) CM-SAM is optional	<input type="checkbox"/> SPC <input type="checkbox"/> E2E	<input type="checkbox"/> SPC

Setting up Interworking Units as members of a Virtual Line on a VX

Use the following settings for the Call Establishment Modes of the VL and IWU when configuring a Substrate Virtual Line in a J1000/J5000 Virtual eXchange. Note that when connecting a VL to another Virtual eXchange, the settings are different to those required for connection to a J1000 Access Controller.

Virtual Line Type	Virtual Line Setting	IWU-R connected to a remote VX	IWU-R connected to a remote AC
SPC/Manual connection operation	<input checked="" type="checkbox"/> SPC Label required for real SPCs No Label for Manual connection	<input checked="" type="checkbox"/> SPC and own label <input checked="" type="checkbox"/> E2E National Significant Remote Number <input checked="" type="checkbox"/> E2E Service Tones <input checked="" type="checkbox"/> Transparent	<input checked="" type="checkbox"/> SPC and label <input checked="" type="checkbox"/> E2E National Significant Remote Number <input checked="" type="checkbox"/> E2E Service Tones <input type="checkbox"/> Transparent
XSPC operation	<input type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E <input checked="" type="checkbox"/> Hotline (both ends) <input checked="" type="checkbox"/> XSPC (one end only) CM-SAM is optional	<input checked="" type="checkbox"/> SPC ¹ <input checked="" type="checkbox"/> E2E <input checked="" type="checkbox"/> E2E Service Tones <input checked="" type="checkbox"/> Transparent	<input checked="" type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E <input checked="" type="checkbox"/> E2E Service Tones <input type="checkbox"/> Transparent
On Demand operation	<input type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E <input checked="" type="checkbox"/> Hotline (both ends) CM-SAM is optional	<input type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E <input checked="" type="checkbox"/> E2E Service Tones <input checked="" type="checkbox"/> Transparent	<input type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E <input checked="" type="checkbox"/> E2E Service Tones <input type="checkbox"/> Transparent

- 1** The SPC is set so that it does not drop the call when the last member of the Virtual Line clears.

Note We recommend that you use XSPC operation instead of manual connections for the configurations shown above.

Setting up a 64K call using Interworking Units

To set up a 64K call via an IWU, use the following Call Establishment Mode settings for the IWU:

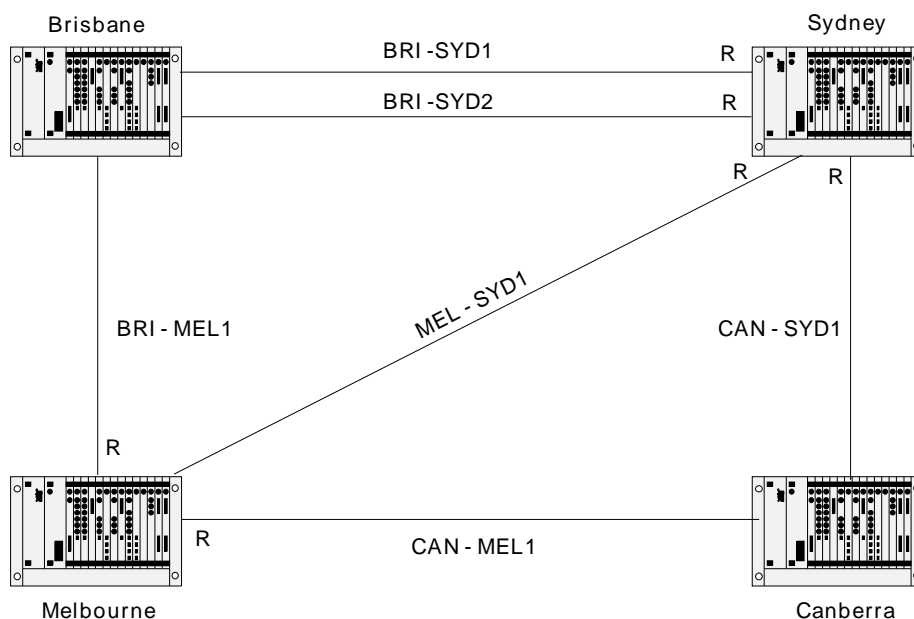
Virtual Line Type	IWU-R settings
SPC/Manual connection operation	<input checked="" type="checkbox"/> SPC and own label <input checked="" type="checkbox"/> E2E National Significant Remote Number <input checked="" type="checkbox"/> E2E Service Tones <input checked="" type="checkbox"/> Transparent (see Note)
XSPC operation	<input type="checkbox"/> SPC <input checked="" type="checkbox"/> E2E <input checked="" type="checkbox"/> E2E Service Tones <input checked="" type="checkbox"/> Transparent (see Note) <input checked="" type="checkbox"/> Hotline <input checked="" type="checkbox"/> XSPC (at one end only, Auto Answer enabled at the other end)

Note *Transparent should be disabled if the remote end is **not** an Interworking channel.*



MicropleX switching applications

MicropleX switching allows switching onto Subrate or TimeFrame Virtual Lines to be performed within a single VX. A new type of subchannel line is configured which is not connected to any physical port. This line can be configured to be Subrate voice or Subrate data. In the case of voice, an LDCM, MFCM or COMBO compander is used to connect the port. For data, an SRMM Subrate switch is used. This new line can have calls directed to it, using the normal call routing techniques such as matching line numbers and/or linesets. The following diagram shows a typical network on which to apply this form of switching.



The sample network connects sites in Brisbane, Sydney, Canberra and Melbourne. Two tie lines are provided between Sydney and Brisbane since most call traffic will flow between these two sites. Redundancy is provided by the tie lines between Sydney and Canberra and between Melbourne and Brisbane. The tie lines are established using XSPCs. Channel signalling is also used. See page 3-38 for general instructions on configuration of D- and B- channel signalling, SPC/XSPC/manual connection or demand-established operation.

The convention of having **left** Interwork Unit lines as members of linesets and **right** Interworking Unit lines as Virtual Line members is followed for this configuration. The 'R' in the above network connection diagram indicates that the Interworking Unit lines which are members of the lineset have their order reversed to minimise call collisions. See *ISDN Interworking* on page 3-26 for more information.

The table below lists the routes and the Interworking Unit lines which are members of the Virtual Line for each route. Interworking Unit A indicates the Interworking Unit lines which are members of the lineset defining the route in Node A and Interworking Unit B for Node B.

Route	Node A	Interworking Unit A	Node B	Interworking Unit B
BRI-SYD1	Brisbane	2/(1-4)R	Sydney	3/(1-4)R
BRI-SYD2	Brisbane	2/(5-8)R	Sydney	3/(5-8)R
BRI-MEL1	Brisbane	1/(1-4)R	Melbourne	3/(1-4)R
CAN-MEL1	Canberra	1/(1-4)R	Melbourne	2/(1-4)R
MEL-SYD1	Melbourne	1/(1-4)R	Sydney	1/(1-4)R
CAN-SYD1	Canberra	2/(1-4)R	Sydney	2/(1-4)R

The following table lists the lineset number ranges and line members defined in each node. The routes associated with the number ranges are also listed. The ordering of the routes in the table indicates the priority for each route. For example, a call from Sydney to Melbourne will choose the MEL-SYD1 route first, followed by the CAN-SYD1 route (Sydney - Canberra - Melbourne), and the last choices are the BRI-SYD1 and BRI-SYD2 routes (Sydney - Brisbane - Melbourne).

Node	Lineset range	Member Interworking Unit lines	Associated route
Sydney	03 - 03	1/(4-1)L (Reversed)	MEL-SYD1
		2/(4-1)L (Reversed)	CAN-SYD1
		3/(4-1)L (Reversed)	BRI-SYD1
		3/(8-5)L (Reversed)	BRI-SYD2
	026 - 026	2/(4-1)L (Reversed)	CAN-SYD1
		1/(4-1)L (Reversed)	MEL-SYD1
		3/(4-1)L (Reversed)	BRI-SYD1
		3/(8-5)L (Reversed)	BRI-SYD2
	07 - 07	3/(4-1)L (Reversed)	BRI-SYD1
		3/(8-5)L (Reversed)	BRI-SYD2
		1/(4-1)L (Reversed)	MEL-SYD1
		2/(4-1)L (Reversed)	CAN-SYD1

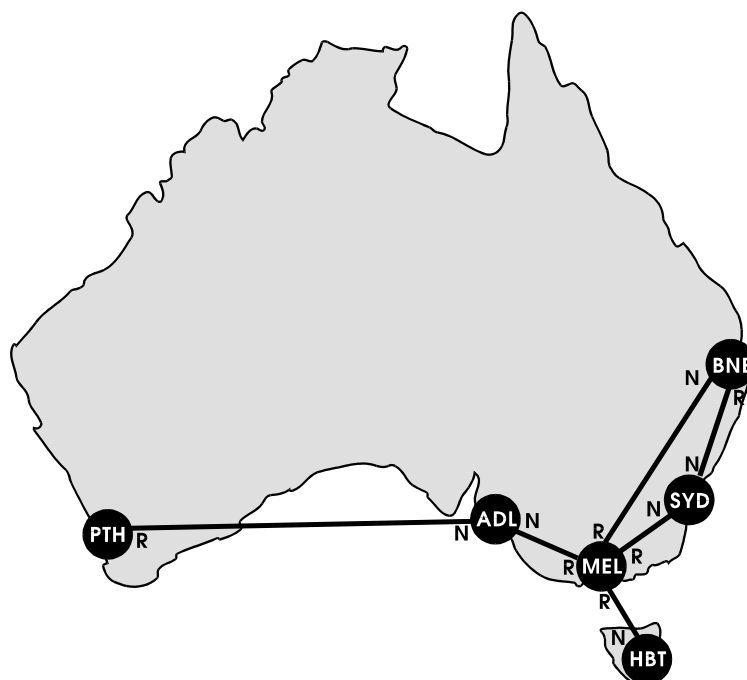
Node	Lineset range	Member Interworking Unit lines	Associated route
Melbourne	07 - 07	3/(4-1)L (Reversed)	BRI-MEL1
		1/(4-1)L (Reversed)	MEL-SYD1
		2/(4-1)L (Reversed)	CAN-MEL1
	026 - 026	2/(4-1)L (Reversed)	CAN-MEL1
		1/(4-1)L (Reversed)	MEL-SYD1
		3/(4-1)L (Reversed)	BRI-MEL1
	029 - 029	1/(1-4)L	MEL-SYD1
		2/(1-4)L	CAN-MEL1
		3/(1-4)L	BRI-MEL1
Canberra	029 - 029	2/(1-4)L	CAN-SYD1
		1/(1-4)L	CAN-MEL1
	03 - 03	1/(1-4)L	CAN-MEL1
		2/(1-4)L	CAN-SYD1
	07 - 07	2/(1-4)L	CAN-SYD1
		1/(1-4)L	CAN-MEL1
Brisbane	026 - 026	2/(1-4)L	BRI-SYD1
		2/(5-8)L	BRI-SYD2
		1/(1-4)L	BRI-MEL1
	03 - 03	1/(1-4)L	BRI-MEL1
		2/(1-4)L	BRI-SYD1
		2/(5-8)L	BRI-SYD2
	029 - 029	2/(1-4)L	BRI-SYD1
		2/(5-8)L	BRI-SYD2
		1/(1-4)L	BRI-MEL1

The VX number range is defined as the range of the PRA or BRA number range. The area code is set to the city area code without the leading 0. For this application a VX call capacity of 15 calls should be adequate.

Virtual eXchange applications

Introduction

The example network described in this section illustrates a private network connecting Australia's major cities. Each node in the network is a Virtual eXchange. The links between the nodes are 2.048 Mbit/s lease lines (for example, Telstra Megalink). Both ends of each leased line are terminated at IPMNs, such that the private network path between two VXs is implemented as an 'IPMN to leased line to IPMN' connection. The network shown below also provides redundant paths for the Sydney, Melbourne and Brisbane nodes.



The following sections describe the configuration of the network shown above. All configuration parameters are mandatory unless indicated as optional.

VX configuration

For full details on configuring the VX see page 3-15 of the *System configuration* section. To configure the VX for the example network carry out the following steps:

- 1 Enable the VX by checking the **Enable VX** checkbox in the **VX Configuration** dialog.
- 2 Set the VX call **Capacity** to an appropriate number of calls, for example 45.
- 3 Set the **VX Tones** or leave them at their default settings (optional).
- 4 Enter the **Minimum number in this device**, the **Maximum number in this device**, and the **Area code** for each node.

- 5 Configure **Call charging** (optional)
- 6 Set **Endless loop prevention** to 4. This is the longest path within this example network.

RM/CM configuration

For a fuller explanation of RM and CM configuration, see page 3-5 and page 3-3 respectively, in the *System configuration* section.

- 1 Enter the **Manager Access** number.
- 2 Enable **Resubmit calls from TE**.
- 3 Enable **Resubmit calls from NT**.
- 4 Enable **Log resubmitted calls** (optional).

Route configuration

The following table shows the configuration for the routes at each node. The numbers programmed into the linesets represent the area codes of the destinations serviced by the node. The IPMN line order for each route corresponds to the Normal (N) or Reversed (R) order indicated on the example network.

The linesets defined below provide for least-cost routing to NSW and VIC country areas, and Canberra/ACT by routing the call to the nearest capital city node. For example, if the Melbourne node receives a call for Canberra (area code 062), it will attempt to send it to the Sydney node via the Sydney private trunk (SYD). The Sydney node, upon receiving a call for Canberra, will fail to match the 062 number to a lineset or locally assigned line. Consequently, it will send the call to a TE module (IPMT), from which point the public ISDN will route the call.

For simplicity, this example assumes that charges incurred for public ISDN switching of calls to any destination within a State's boundaries are least expensive if the call is made from the capital city of that State, and not a neighbouring one. This may not necessarily be the case, as public ISDN call charging is usually based on the radial distance of the destination from the source. To fine tune the least-cost routing characteristics, different linesets with narrower number ranges would need to be configured.

The IPMNs have been assigned to the linesets in order of shortest path first. For example, the Melbourne destination linesets in the Sydney node have been configured with the direct route to Melbourne as the first member, and the indirect Brisbane route as the second member. This configuration provides the following call path preference:

- 1st Sydney—Melbourne
- 2nd Sydney—Brisbane—Melbourne.

Each lineset has **Allow TE submission when call not originating from this device fails** enabled and **Allow calls arriving on member interface to be sent back on same interface** disabled.

Configuration details table

VX Node	Destination	Linesets (min-max)	Member routes (Private line IPMNs)	IPMN line order
Sydney <i>Area codes serviced by this node begin with 02, 04 and 06.</i>	Hobart & TAS	002—002	MEL (Melbourne) BNE (Brisbane)	Normal Normal
	Melbourne	03—03	MEL (Melbourne) BNE (Brisbane)	Normal Normal
	VIC	05—05	MEL (Melbourne) BNE (Brisbane)	Normal Normal
	Brisbane & QLD	07—07	BNE (Brisbane) MEL (Melbourne)	Normal Normal
	Adelaide & SA Perth & WA	08—09	MEL (Melbourne) BNE (Brisbane)	Normal Normal
Melbourne <i>Area codes serviced by this node begin with 03 and 05</i>	Hobart & TAS	002—002	HBT (Hobart)	Reversed
	Sydney	02—02	SYD (Sydney) BNE (Brisbane)	Reversed Reversed
	NSW	04—04	SYD (Sydney) BNE (Brisbane)	Reversed Reversed
	Canberra & ACT	06—06	SYD (Sydney) BNE (Brisbane)	Reversed Reversed
	Brisbane & QLD	07—07	BNE (Brisbane) SYD (Sydney)	Reversed Reversed
	Adelaide & SA Perth & WA	08—09	ADL (Adelaide)	Reversed
Adelaide <i>Area codes serviced by this node begin with 08</i>	Hobart & TAS	002—002	MEL (Melbourne)	Normal
	Sydney Melbourne NSW VIC Canberra & ACT Brisbane & QLD	02—07	MEL (Melbourne)	Normal
	Perth & WA	09—09	PTH (Perth)	Normal
Brisbane <i>Area codes serviced by this node begin with 07</i>	Hobart & TAS	002—002	MEL (Melbourne) SYD (Sydney)	Normal Reversed
	Sydney	02—02	SYD (Sydney) MEL (Melbourne)	Reversed Normal
	Melbourne	03—03	MEL (Melbourne) SYD (Sydney)	Normal Reversed
	NSW	04—04	SYD (Sydney) MEL (Melbourne)	Reversed Normal
	VIC	05—05	MEL (Melbourne) SYD (Sydney)	Normal Reversed
	Canberra & ACT	06—06	SYD (Sydney) MEL (Melbourne)	Reversed Normal
	Adelaide & SA Perth & WA	08—09	MEL (Melbourne) SYD (Sydney)	Normal Reversed

VX Node	Destination	Linesets (min-max)	Member routes (Private line IPMNs)	IPMN line order
Perth	Hobart & TAS	002—002	ADL (Adelaide)	Reversed
<i>Area codes serviced by this node begin with 09</i>	Sydney Melbourne NSW VIC Canberra & ACT Brisbane & QLD Adelaide & SA	02—08	ADL (Adelaide)	Reversed
Hobart	Sydney Melbourne NSW VIC Canberra & ACT Brisbane & QLD Adelaide & SA	02—08	MEL (Melbourne)	Normal
<i>Area codes serviced by this node begin with 002</i>				

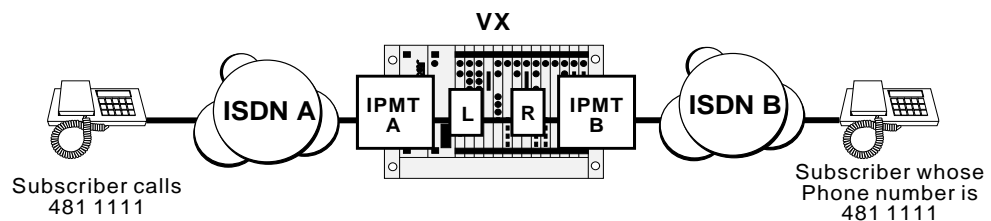
Interworking applications

This section details several scenarios for operating with the Interworking feature of this equipment. For further information on configuring Interworking see page 3-26 in the *System configuration* section.

In the following examples only the non-default settings are detailed. All other settings are assumed to be at their default value. Where any examples show the use of an IPMT, you may also use a BRMT. Similarly, where an Access Controller is specified, you may also use a VX.

Switched calls between two networks

In this application example, the called number or range of numbers are on a different network to the one originating the call.



Configuration

- 1 Install one Interwork unit.
- 2 Set Interwork unit 1 L to have:
 - **Outgoing TE** set to **Exclusive**
 - **Use TE** set to **IPMT A**.
- 3 Set Interwork 1 R to have:
 - **Outgoing TE** set to **Exclusive**
 - **Use TE** set to **IPMT B**.
- 4 Set Interwork 1, Line 1 Left to have:
 - **Line number** set to **4811111**
 - **Indial** set to **On**.
- 5 Set Interwork 1, Line 1 Right to its default configuration.

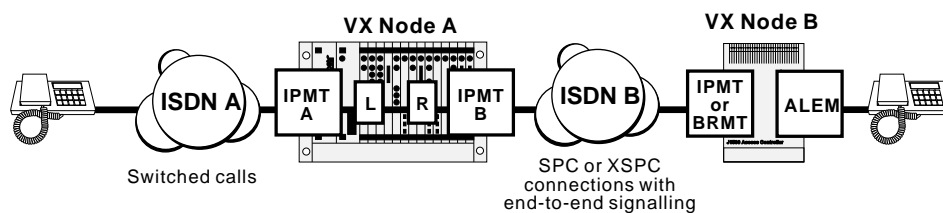
Operation

The incoming call from ISDN A, with called party number 4811111, is directed to Interwork 1, Line 1 Left. The called party number is passed to Interwork 1, Line 1 Right. This causes an outgoing Setup to be issued to ISDN B, with called party number 4811111, and the same ISDN Information Elements as the incoming call from ISDN A. All signalling, that is, Alerting, Connect, Disconnect and so on, received from the outgoing call is reflected back to the incoming call. When either call is cleared, the line is cleared.

The configuration described above only supports one call at a time. To support multiple calls, create a lineset that has one half of the Interwork unit as its members. Then set the lineset **Min** and **Max numbers** to 4811111. If you wish to support a range of phone numbers, set the lineset **Min** and **Max numbers** accordingly.

Using remote tie line to make switched calls

This application can be used for bridging two networks, where tie lines operate across one network and switched calls across the other. This application can also provide tie lines with access to local switched calls on the same network.



Note: ISDN A and ISDN B can be the same network. If this is the case, IPMT A and IPMT B can be the same ISDN interface

Configuration

VX Node A

Configure the VX as follows:

- 1 Install one Interwork unit.
- 2 If ISDN A and B are different networks, set **Outgoing TE** to **Exclusive**.
- 3 Configure Interwork 1, Line 1 Left to have:
 - **Bearer Capability** set to **Speech A-law**.
- 4 Configure Interwork 1, Line 1 Right to have:
 - **Line number** configured to **4811111**
 - **Auto answer** set to **On**
 - **End-to-end signalling** set to **On**.

VX Node B

- 1 Set the ALEM port to have:
 - **Hotline number** set to **4811111**
 - **Hotline** enabled
 - **End-to-end** set to **On**
 - **XSPC** set to **On**.

Operation

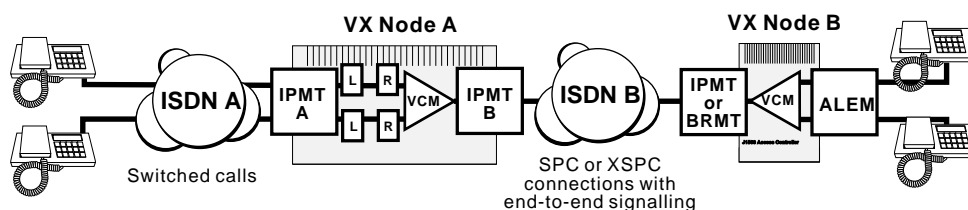
The configuration of the ALEM port in the Node B causes a call to be established between the ALEM and the Interwork 1, Line 1 Right. This call also carries the IDAS end-to-end signalling.

When the handset connected to the ALEM in the VX Node B goes off-hook, an IDAS setup message is sent to the Interwork 1, Line 1 Right of VX Node A. This is passed to the Interwork 1, Line 1 Left, which issues a setup message to ISDN A. Any signalling that is received from ISDN A is reflected back in IDAS messages to the ALEM. When digits are dialled into the ALEM, they are carried over IDAS to the Interwork 1, Line 1 Right, which then translates them into ISDN dialling information. When the handset goes on-hook, the switched call is cleared.

If call progress tones are not provided by ISDN A, they can be provided by VX Node A. To enable VX Node B tones, enable **E2E Service tones** for the Interwork 1, Line 1 Right.

Compressed tie lines to switched calls

This application is similar to the previous one and can also be used when bridging two networks where tie lines operate across one network and switched calls across the other. Similarly, it can provide tie lines with access to local switched calls on the same network. In this particular case, two voice lines share the one tie line using voice compression.



Note: ISDN A and ISDN B can be the same network. If this is the case, IPMT A and IPMT B can be the same ISDN interface

Configuration

VX Node A

Configure the VX as follows:

- 1 Install one Interwork unit.
- 2 If ISDN A and ISDN B are different networks, set **Outgoing TE** to **Exclusive**.
- 3 Create a Subrate Virtual Line with Interwork 1, Line 1 Right and Interwork 1, Line 2 Right as its members. Allocate 32 kbit/s to each line.
- 4 Configure the TimeFrame or Subrate Virtual Line to have:
 - **Auto answer** set to **On**.
 - **End-to-end** set to **On**.
- 5 Configure Interwork 1, Line 1 Left to have:
 - **Bearer Capability** set to **Speech A-law**.
- 6 Configure Interwork 1, Line 2 Left to have:
 - **Bearer Capability** set to **Speech A-law**.

7 Configure Interwork 1, Line 1 Right to have:

- **SPC** set to **On** (Australia only).
- **Member of Subrate Virtual Line** or **manually connected mode** set to **On** (UK only).
- **End-to-end signalling** set to **On**.

8 Configure Interwork 1, Line 2 Right to have:

- **SPC** set to **On** (Australia only).
- **Member of Subrate Virtual Line** or **manually connected mode** set to **On** (UK only).
- **End-to-end signalling** set to **On**.

VX Node B

Configure the VX as follows:

- 1** Create a Subrate Virtual Line with the two ALEM lines as its members. Allocate 32 kbit/s to each line.
- 2** Configure the Subrate or TimeFrame Virtual Line to have:
 - **Hotline number** set to **4811111**.
 - **Hotline** enabled.
 - **End-to-end** set to **On**.
 - **XSPC mode** set to **On**.
- 3** Configure both ALEM lines to have:
 - **End-to-end** set to **On**.

Operation

The configuration of the Subrate or TimeFrame Virtual Line in VX Node B causes a call to be established between it and the Virtual Line in VX Node A. This call also carries the IDAS end-to-end signalling.

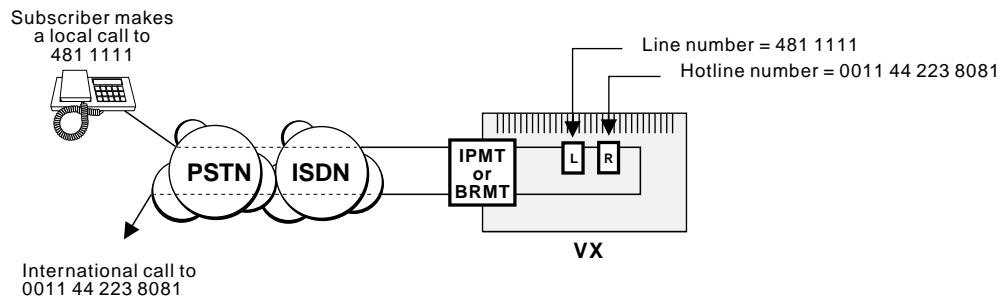
When one of the handsets connected to the ALEM in VX Node B goes off-hook, an IDAS setup message is sent to Interwork 1, Line 1 Right in VX Node A. This is passed to Interwork 1, Line 1 Left, which issues a setup message to ISDN A. Any signalling that is received from ISDN A is reflected back in IDAS messages to the ALEM. When digits are dialled into the ALEM, they are carried over IDAS to the Interwork unit in VX Node A, which then translates them into dialling information. When the handset goes on-hook, the switched call is cleared.

The two handsets can operate independently, while sharing the same B-channel through ISDN B.

If call progress tones are not provided by ISDN A, they can be provided by VX Node A. To enable VX Node B tones, enable **E2E Service tones** for the Interwork 1, Line 1 Right and Interwork 1, Line 2 Right.

Local call to an international number

This application allows you to make local calls which relay onto an international call to a predetermined destination. This may be useful to provide a toll-free international access to customers.



Configuration

- 1 Install one Interwork unit.
- 2 Configure Interwork 1, Line 1 Left to have:
 - **Line number** set to **4811111**
 - **Indial** mode set to **On**.
- 3 Configure Interwork 1, Line 1 Right to have:
 - **Hotline number** set to **0011442238081**
 - **Hotline** enabled.

Operation

An incoming call from a subscriber in the ISDN or PSTN to 4811111 will be directed to Interwork 1, Line 1 Left. This causes an outgoing call setup to be issued by Interwork 1, Line 1 right to 0011 44 223 8081. This call has the same ISDN information elements as the incoming call. All signalling, that is, Alerting, Connect, Disconnect and so on, received from the ISDN for the outgoing call is reflected back to the incoming call. When one party clears their call, the other is also cleared.

Multiple calls

The configuration described above supports only one call at a time. To support up to 30 calls, do not set the line number for Interwork 1, Line 1 Left. Instead, create a lineset with 4811111 for both the **Minimum** and **Maximum numbers**. Add the whole of the Interwork 1, Left to the lineset. Turn off Indial in all Interwork 1, Left lines and configure the Hotline number and Hotline mode for all Interwork 1, Right lines. To support 30 calls at a time, two IPMTs have to be fitted because each Interworking operation requires both an incoming and an outgoing call.

CallBack

CallBack is a useful feature that takes advantage of lower tariffs that may be available in one location. The feature allows a call to be set up from one end of a network, regardless of where the call originates. This ensures that only one end of the network pays for call charging. CallBack uses interworking units to set up the required signalling and call charging.

The feature can be used for compressed calls involving Subrate and TimeFrame Virtual Lines as well as non-compressed data calls. Non-compressed voice calls have limited support using CallBack. Non-compressed voice calls can be used if the application requires the appropriate voice port to be set to Auto Answer at the CallBack Originating end, with the connected device (e.g. PBX) providing the appropriate tones. An example of where CallBack could be used in this scenario is with Virtual eXtension (see page 5-86). CallBack can only be used for demand established calls.

When making a CallBack, the RM examines the incoming CLI and, if present, uses that to establish the call back to the initiating end. This allows multiple remote sites to call into a single CallBack site. If no CLI is present or the **CBack Hotline** is set, the **Hotline Number** in the CallBack Originating Right Hand End Interwork Unit is used to establish the CallBack.

The CLI Number Checking feature (see page 3-49) can be used to provide additional security on calls for networks where CLI is provided.

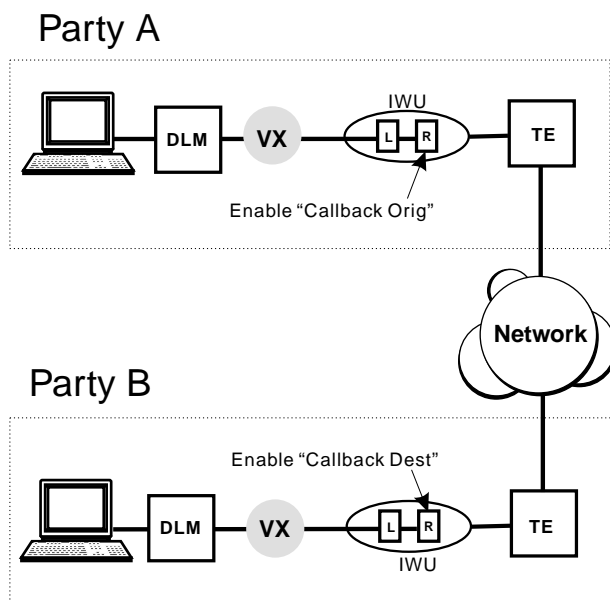
Note *CLI is not provided on all international calls. In this case security may be a concern.*

When using CallBack on a QBRM, the option Send Calling Party Number must be enabled in the configuration dialog in OmniVision. See page 4-27 for details on configuring the QBRM.

Note *For CallBack trunks carrying compressed voice that are initiated from the end nominated as the destination, the first voice call will take longer to establish than subsequent calls due to the time involved in making a call, clearing it and then calling back from the other end to establish the trunk.*

Non compressed data call

The following diagram shows a typical implementation for a non-compressed data call with no end-to-end signalling.



In the above scenario, Party A is the one selected to receive the billing for any calls occurring between the two parties. If Party A initiates a call, the call signalling setup occurs normally and either party can clear the call. If Party B initiates the call, the setup signal is detected by the Party A Interworking Unit and the call is setup in the opposite direction. Again either party can clear the call.

Configuration - Party A

To configure the CallBack feature for Party A:

- 1 Configure a lineset and direct it to the Interworking Unit (IWU) chosen for the CallBack feature.
- 2 In the chosen IWU line (right) Call Establishment Modes, enable the **CallBack Orig** checkbox. Enable and enter the **Hotline number** towards the B party IWU (right).
- 3 In the chosen IWU line (left) enable and enter the **Hotline number** towards the DLM port.
- 4 Select the Outgoing TE for the line.

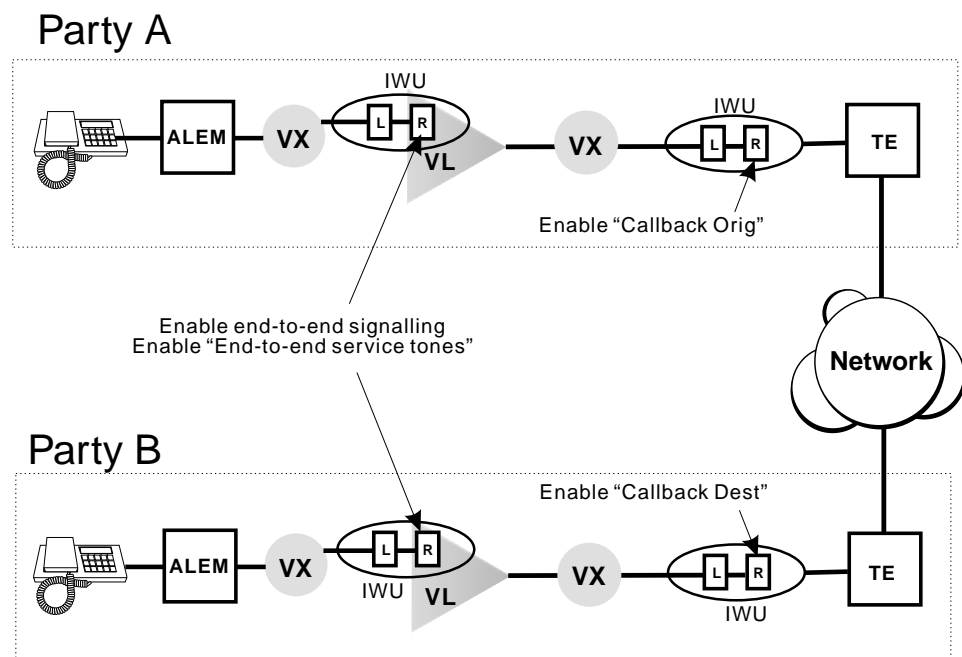
Configuration - Party B

To configure the CallBack feature for Party B:

- 1 Configure a lineset and direct it to the Interworking Unit (IWU) chosen for the CallBack feature.
- 2 In the chosen IWU line (right) Call Establishment Modes, enable the **CallBack Dest** checkbox. Enable and enter the **Hotline number** towards the A party IWU (right).
- 3 In the chosen IWU line (left) enable and enter the **Hotline number** towards the DLM port.
- 4 Select the Outgoing TE for the line.

Compressed call using Virtual Line

The following diagram shows a typical implementation of CallBack for a compressed call using a Virtual Line



In this scenario, compressed calls are implemented. Party A is the one selected to receive the billing for any calls occurring between the two parties. If Party A initiates a call, the call signalling setup occurs normally and either party can clear the call. If Party B initiates the call, the setup signal is detected by the Party A Interworking Unit and the call is setup in the opposite direction. Again either party can clear the call. Because the call has to pass through more than one VX, additional configuration is required.

Configuration - Party A

To configure the CallBack feature for Party A:

- 1 Configure a Subrate Virtual Line and direct it to an Interworking Unit (IWU).
- 2 In the chosen IWU line (right) Call Establishment Modes, enable the **CallBack Orig** checkbox. Enable and enter the **Hotline Number** towards the B party IWU (right).

Enable **CBack Hotline** if you prefer to use the Hotline Number for the CallBack, rather than the CLI of the **CallBack Dest**. This is generally used when the received CLI has been altered by the network, which can happen when CallBack is used across international boundaries.

- 3 In the chosen IWU line (left) enable and enter the **Hotline number** in the direction of the Virtual line.
- 4 In the Virtual Line IWU (right) Common Configuration, enable **End-to-end signalling** and **End-to-end service tones**.
- 5 Select the Outgoing TE for the line.

Configuration - Party B

To configure the CallBack feature for Party B:

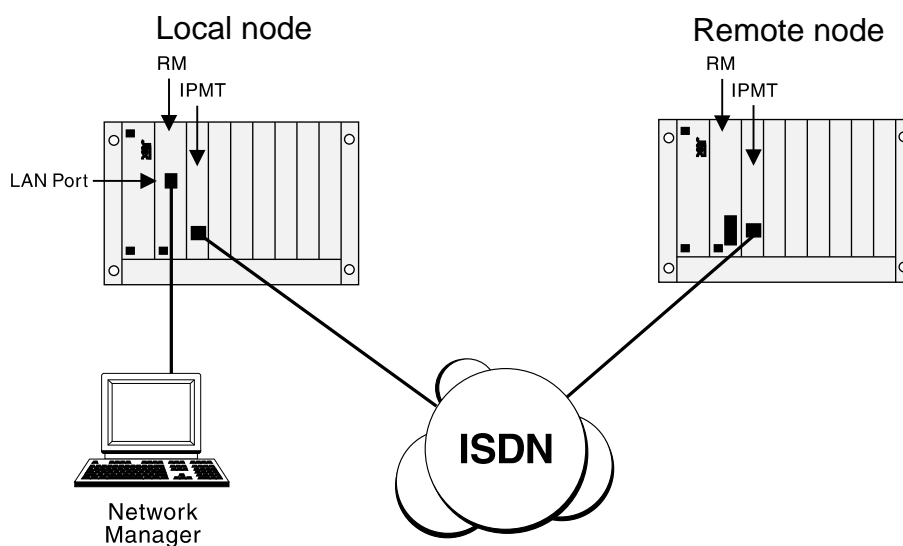
- 1 Configure a lineset and direct it to the Interworking Unit (IWU) chosen for the CallBack feature.
- 2 In the chosen IWU line (right) Call Establishment Modes, enable the **CallBack Dest** checkbox. Enable and enter the **Hotline number** towards the A party IWU (right).
- 3 In the chosen IWU line (left) enable and enter the **Hotline number** in the direction of the Virtual line.
- 4 In the Virtual Line IWU (right) Common Configuration, enable **End-to-end signalling** and **End-to-end service tones**.
- 5 Select the Outgoing TE for the line.

Inband signalling applications

This section details two ways of connecting and configuring the equipment to use inband signalling. For a fuller explanation of this feature see page 3-43 in the *System configuration* section.

Simple point-to-point connection

This example shows a simple point-to-point connection involving two nodes and an ISDN call.



Configuration details

The routing table for the local node is configured to use the backplane router port of the Resource Manager (RM) or Control Module (CM) as the primary route. This port has a Hotline number so that when a connection is required, a B-channel call will be made to the remote node.

Where D-channel calls are available, the secondary route uses one of the D-channel routing ports of the Local node. If the primary route fails, a D-channel call is made to the remote node.

The configuration details for each node are as follows:

Local node

SAM (CM/RM) inband:

- Set to 8 kbit/s
- Set Hotline number. For example, 3840050.

SAM (CM/RM) D-channel 1: (Australia only)

- Set Hotline number. For example 3840050.

CM/RM-SAM routing table

Addresses	Primary route	Secondary route
3840050	SAM (CM/RM) inband	SAM (CM/RM) D-channel 1

Remote node

SAM (CM/RM) inband:

- Set to 8 kbit/s.

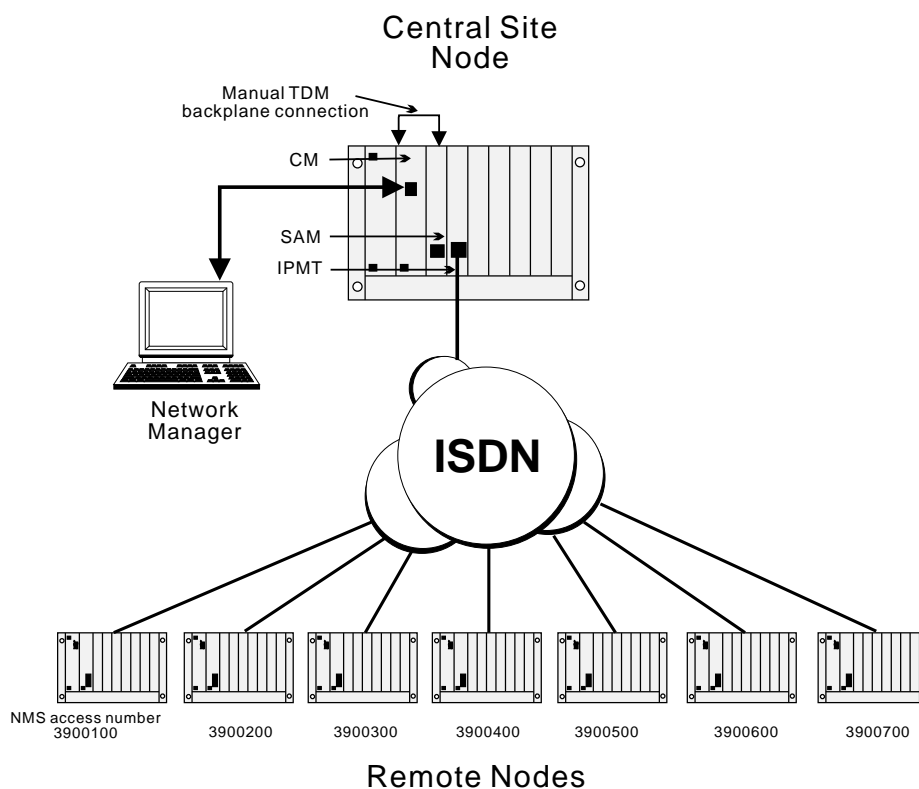
Manager access number

- Enter the manager access number. For example, 3840050.

Simple tree network

The following example shows a scenario where a SAM module is used at the central site to enable OmniVision to manage remote nodes at several branch offices. Each link is 8 kbit/s and can share the B-channel with compressed voice circuits or other data channels.

More branch offices can be added by using more SAMs to the central site chassis, or by adding SAMs to one or more of the branch offices. The latter method allows the remote nodes to be connected to other remote offices.



Configuration details

The network is configured as follows:

Central site node

SAM (CM) inband: Set to 8 kbit/s

SAM inb 1: Set to 8 kbit/s

SAM inb 2: Set to 8 kbit/s, Hotline = 3900100

SAM inb 3: Set to 8 kbit/s, Hotline = 3900200

SAM inb 4: Set to 8 kbit/s, Hotline = 3900300

SAM inb 5: Set to 8 kbit/s, Hotline = 3900400

SAM inb 6: Set to 8 kbit/s, Hotline = 3900500

SAM inb 7: Set to 8 kbit/s, Hotline = 3900600

SAM inb 8: Set to 8 kbit/s, Hotline = 3900700

CM/RM-SAM routing table:

Addresses	Primary route
3900100 — 3900700	SAM (CM/RM) inband

SAM routing table:

Addresses	Primary route
3900100	SAM inb 2
3900200	SAM inb 3
3900300	SAM inb 4
3900400	SAM inb 5
3900500	SAM inb 6
3900600	SAM inb 7
3900700	SAM inb 8

Manual connection: SAM (CM/RM) inband to SAM inb 1.

Remote nodes

All remote nodes are configured with their respective Manager access numbers.

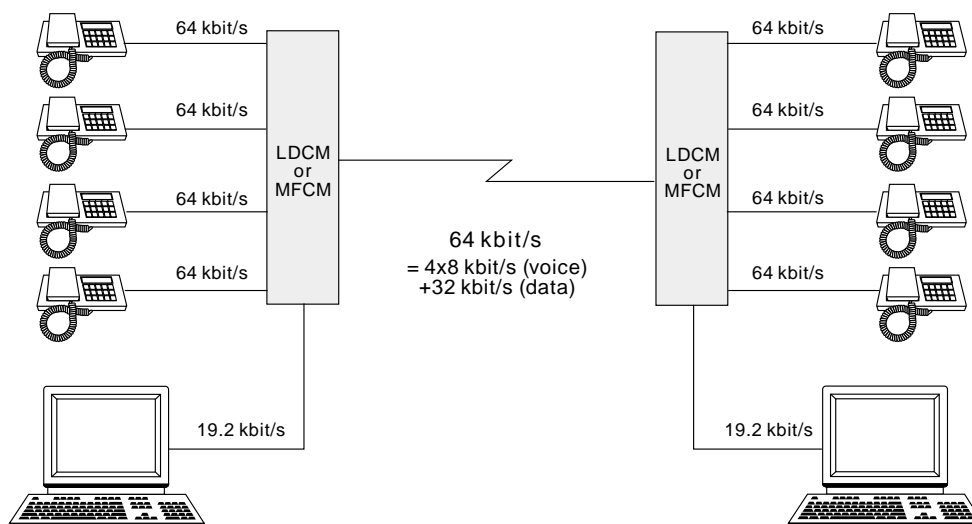
LDCM and MFCM applications

This section details several applications for both the LDCM and MFCM. For details of the operation and configuration of the LDCM see page 4-189 of the *Modules* section. For the same information on the MFCM see page 4-193 of the *Modules* section. All of the following applications can be performed with the CELP and MP-MLQ versions of the LDCM. Apart from the last application with the VCM, all other applications can be performed with the MFCM.

Tie line application

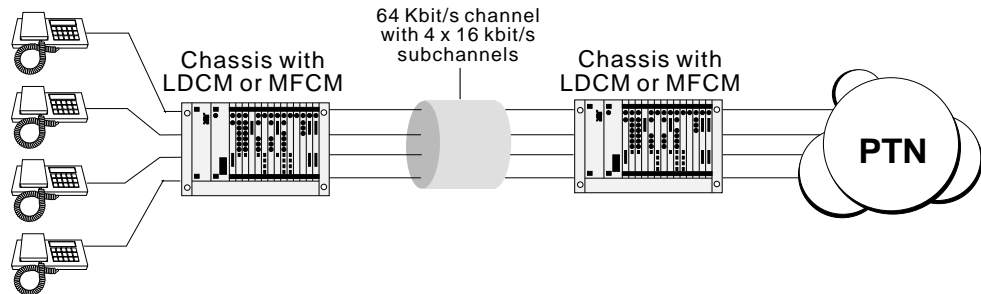
The LDCM and MFCM can be used where voice tie lines are needed between members of the this equipment series. The modules allows the voice signals to be compressed and switched onto a common channel, thus using fewer network access channels.

Where data leased lines are also needed between the sites, the voice and data sub-channels can be multiplexed together using the Substrate multiplexer. The following diagram depicts the concept:



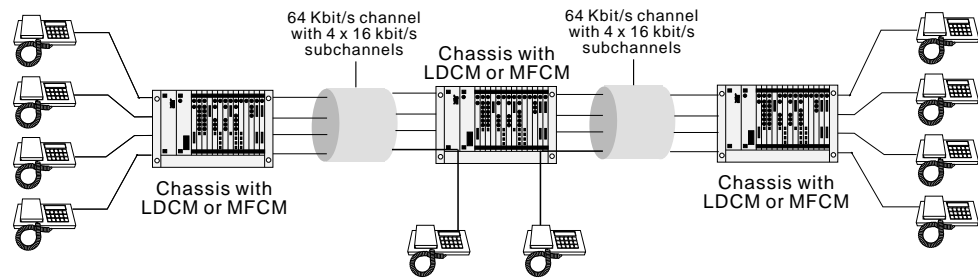
Private network with access to PTN

The following diagram shows how a private voice network can be constructed with access to the public network.



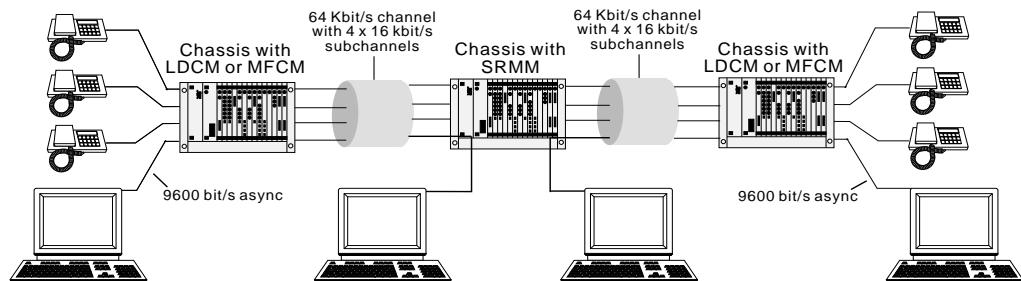
Transit node - voice only application

The following diagram shows how a private voice network can be constructed with a transit node drop and insert.



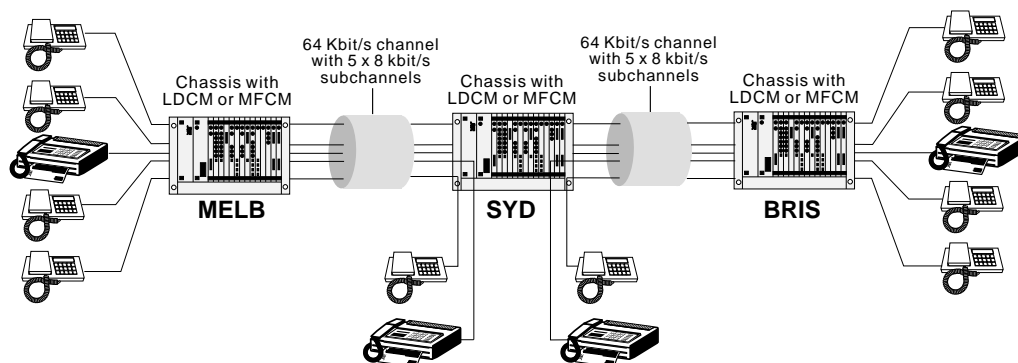
Transit node — voice / data application

The following diagram shows a compressed voice/data network with transit node drop and insert for data only. Note that the transit node does not require an LDCM or MFCM, but only an SRMM.



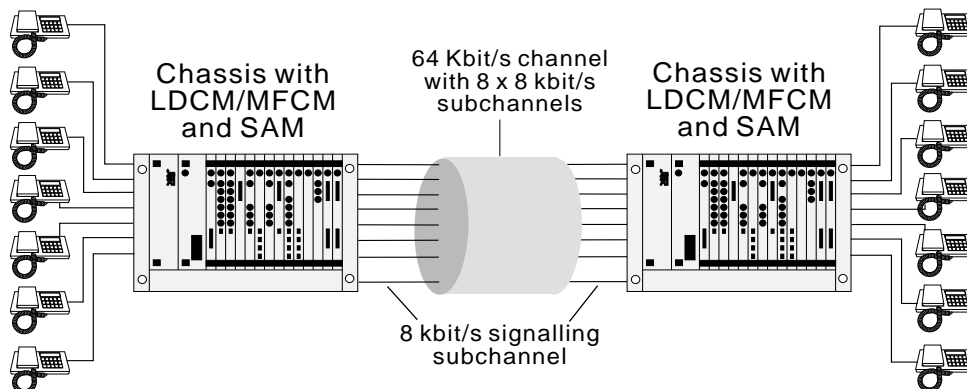
Transit node — voice /fax network application

The following diagram shows how a private voice/fax network can be constructed with a transit node drop and insert. This example shows a main office in Sydney and branch offices in Melbourne and Brisbane. The Sydney node acts as a transit node, so that any calls for Sydney from Melbourne or Brisbane are off-loaded (drop). If the call is for one of the other offices, that is Melbourne is calling Brisbane or vice versa, Sydney calls (insert) the respective destination.



Application with Signalling Access Module (SAM)

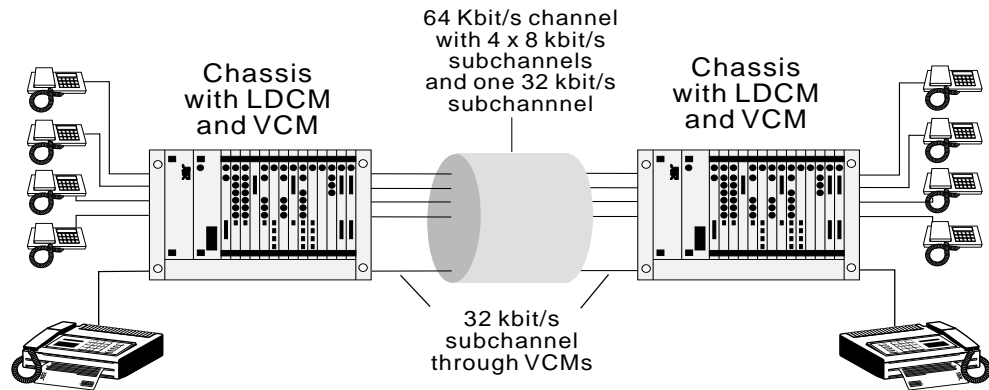
The following diagram shows a compressed voice network with inband (SAM) signalling:



Application with VCM

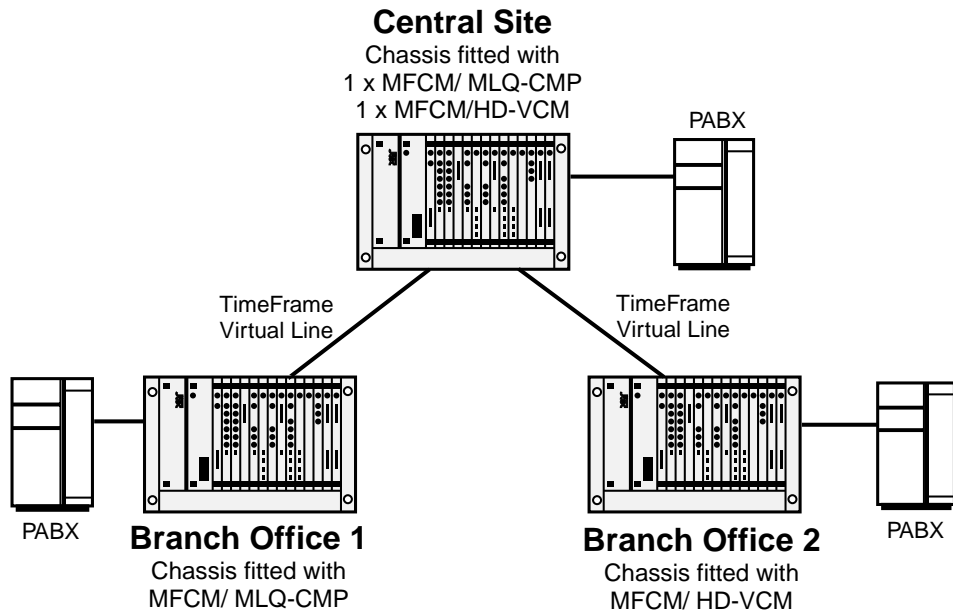
Note *The MFCM cannot be used with his application.*

The following diagram shows a compressed voice network with an LDCM for CELP compression of voice channels and a VCM for ADPCM compression of a full speed (that is, 9600 bit/s analog modulated) fax channel:



Mixed TimeFrame Network with HD-VCMs and MLQ-CMPs

You should give special consideration to networks that contain MFCMs with both HD-VCM and MLQ-CMP submodules. The main issue is the maintenance of compatibility rules for TimeFrame Virtual Lines. The following diagram shows a network that contains MFCMs equipped with both sorts of compression submodules.



In this network, a voice call can be established from Branch Office 1 to the Central Site across TimeFrame. The call cannot be made to Branch Office 2 because it does not have an MFCM fitted with MLQ-CMP submodules.

For similar reasons, Branch Office 2 can call the Central Site across TimeFrame, but cannot establish a call to Branch Office 1. In this case, Branch Office 1 does not have an MFCM with HD-VCMs.

The Central Site can call either Branch office across TimeFrame because it has both types of MFCM.

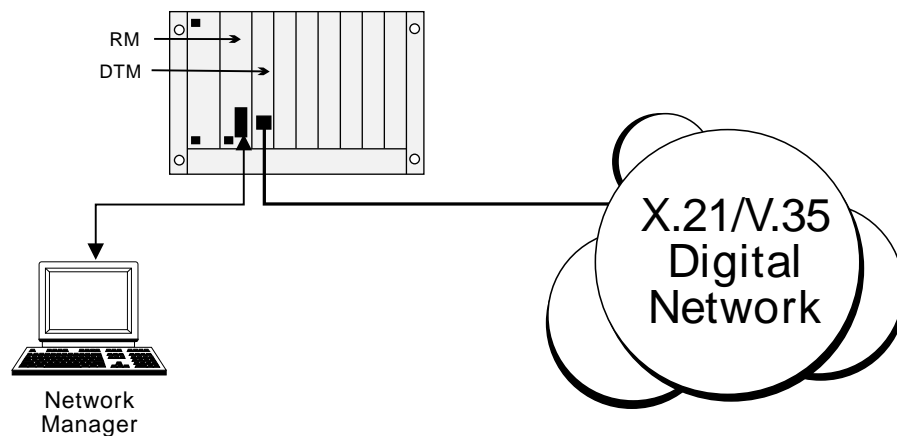
To set up this network to allow a site to call any other site, an MFCM fitted with HD-VCMs must be added to Branch Office 1. Alternatively, an MFCM fitted with MLQ-CMPs can be installed at Branch Office 2.

It is important to verify that the two end points of a call involving TimeFrame Virtual Lines have the same voice compression algorithms available. For more information refer to the *MFCM* section on page 4-193.

Digital Trunk Module (DTM) applications

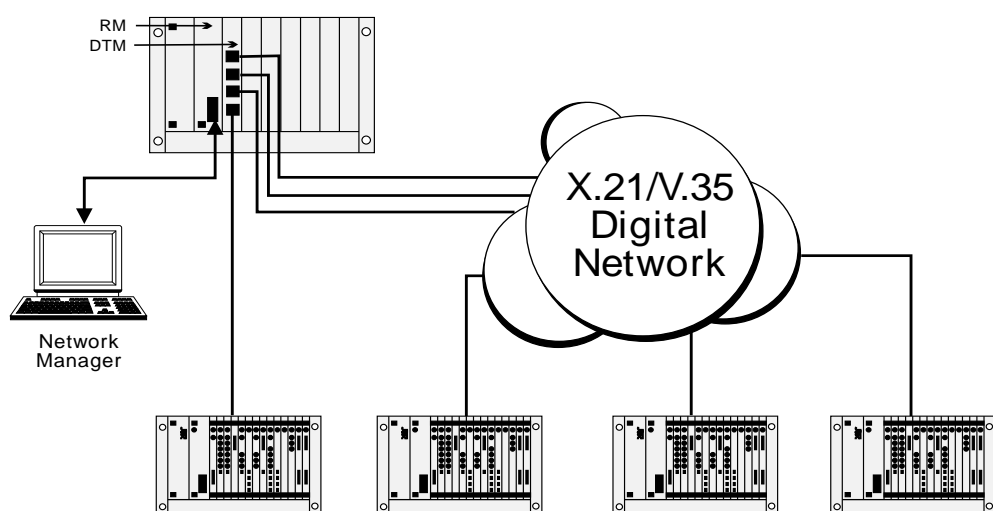
Overview

As with any other module, you configure the DTM using a Network Management PC connected to the CM or RM. The DTM provides a service similar to that provided by an IPM. As such, examples given in this section will highlight DTM-specific features only and how they are used. More comprehensive network configuration examples can be found in *Virtual eXchange* on page 5-30.



Remote Configure (Autosynchronise)

A Network Management PC locally connected to a chassis can use the D- or B-channels to establish remote sessions with other nodes. This feature is preserved with DTM connected nodes. The subB-channel cannot support Network Management calls. The diagram below depicts this scenario.

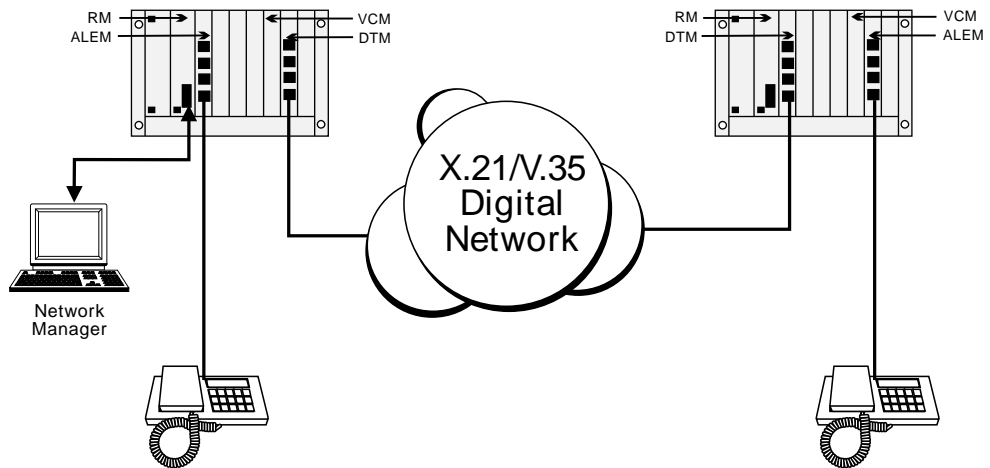


The local node should have the local DTM port configured for the corresponding network clock speed and so on. When an unconfigured chassis is powered up at a remote site, the DTM ports enter an Automatic Synchronisation mode, where clock and D-channel speeds are automatically determined (ensure the local DTM port is idle) and frame synchronisation is achieved. Once frame synchronisation is established, remote sessions can proceed as normal.

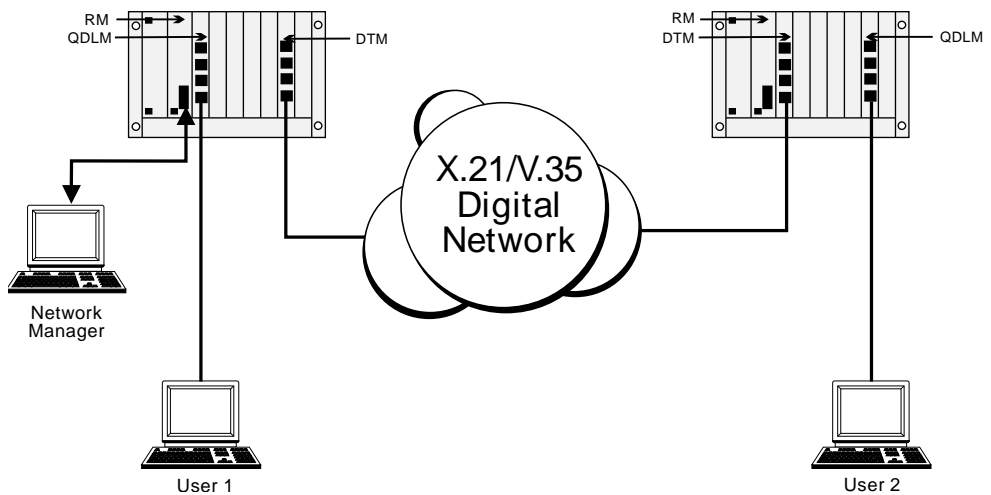
Notes

- *When using the D-channel of a DTM for remote Network Management sessions, a fast D-channel will improve performance.*
 - *If the DTM port that is providing the remote Network Management session path requires a configuration change that would cause a synchronisation loss, perform the remote change first. Synchronisation will be lost. You then configure the local DTM port for the change and restart the remote Network Management session.*
-

Using the subB-channel



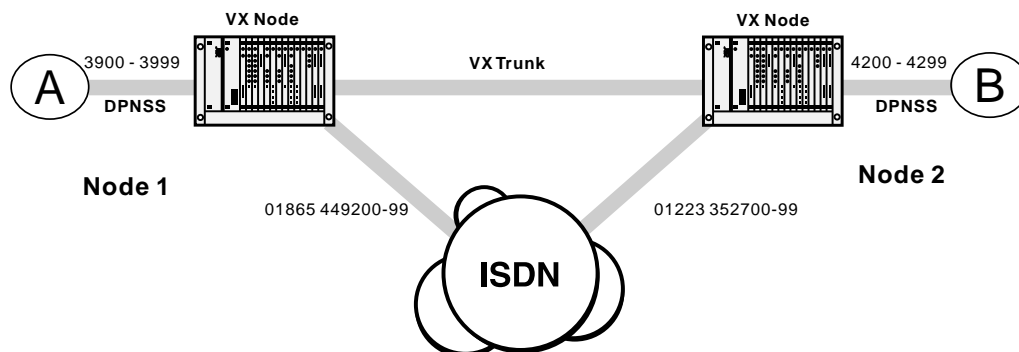
The subB-channel provides up to 56 kbit/s, that is, up to 7 bits of the full 8 bits per octet. This means that Subrate Lines, Subrate Virtual Lines and TimeFrame Virtual Lines can be used in the subB-channel, up to its configured speed. Every 8 kbit/s of subB speed corresponds to one bit (starting from Bit 7). Refer to *subB-channel usage* on page 4-85.



The subB-channel can also be used directly for Subrate speed digital connections, that is, without a Subrate multiplexing module. Ensure that the Digital Line Module has configured a V.110 Subrate. This Subrate shall be less than or equal to the subB speed.

DPNSS applications

This section details the configuration required for the following sample two node network.



PABX A can dial 2 types of numbers:

- 4200-4299 is used to reach extensions on PABX B.
- Anything else is used to reach numbers on the public network.

Similarly, PABX B can also dial 2 types of numbers:

- 3900-3999 is used to reach extensions on PABX A.
- Anything else is used to reach numbers on the public network.

Three alternative configurations are listed:

- A 64K uncompressed network
- 32K ADPCM voice compression on demand
- 32K ADPCM manually connected.

The basic configuration is the same for each; only the Bandwidth Configuration is different.

The following instructions assume you are starting with a defaulted configuration in each node and subsequently lists only the things you have to change from default.

In this scenario the chassis, in addition to a Resource Manager/Control Module, and Power Supply Unit, contains the following modules:

- IPMT-B190 in slot 1
- IPMN-ETSI in slot 3
- E1M-DPNSS in slot 5
- SAM in slot 10
- VCM in slot 15 (only required for the 32K examples).

General configuration

Node 1

NMS number	449298
Virtual eXchange	Min 449200 Max 449299 Area code 1865

Node 2

NMS number	352798
Virtual eXchange	Min 352700 Max 352799 Area code 1223

DPNSS module configuration

Node 1

Lineset	3900-3999	DPNSS 30-1
General settings	A,X, Terrestrial CPE Establishes Min Number length 4	

Node 2

Lineset	4200-4299	DPNSS 30-1
General settings	B,Y, Terrestrial CPE Establishes Min Number length 4	

Number Translation Configuration

Node 1

Number Range name	OneTwo	
Numbers	4200-4299	Apply to DPNSS called
Modify	Replace with	01223352700
Apply to	DPNSS Module Slot 5	All lines

Node 2

Number Range name	TwoOne	
Numbers	3900-3999	Apply to DPNSS called
Modify	Replace with	01865449200
Apply to	DPNSS Module Slot 5	All lines

Signalling configuration

Router tables

Node 1

CM/RM-SAM	Number Range	01223352700-01223352799
Route selection	Primary D-Channel 1	Secondary CM/RM Inband
SAM	Number Range	01223352700-01223352799
Route selection	Primary B-Channel 2	Secondary (none)

Node 2

CM/RM-SAM	Number Range	01865449200-01865449299
Route Selection	Primary D-Channel 1	Secondary CM/RM Inband
SAM	Number Range	01865449200-01865449299
Route Selection	Primary B-Channel 2	Secondary (none)

Signalling channels

Node 1

CM/RM-SAM D-Channel 1	Line Number	449295
	Hotline number	01223352795
	Other	Hotline mode Auto Answer on Acknowledge off
SAM B-Channel 2	Line Number	449296
	Hotline number	01223352796
	Other	Hotline mode Don't use for i/c Manager

Node 2

CM/RM-SAM D-Channel 1	Line Number	352795
	Hotline number	01865449295
	Other	Hotline mode Auto Answer on Acknowledge off
SAM B-Channel 2	Line Number	352796
	Hotline number	01865449296
	Other	Hotline mode Don't use for i/c Manager

Bandwidth Configuration

64K Uncompressed

Node 1

Lineset	Range	01223352700-01223352799
	Members	IPMN 3, 1-30 *
	Attributes	Non-Local

Node 2

Lineset	Range	01865449200-01865449299
	Members	IPMN 3, 30-1 *
	Attributes	Non-Local

*** Note** *End of link should have the channel order reversed to minimise channel collisions.*

32K Compressed on demand

Only four channels (two Virtual Lines) each way are shown in this example. To add capacity add more Virtual Lines.

Add one interwork unit (30 lines) to each node.

Node 1

Virtual Line 1 OneTwo1	Members	0,1,1 0,1,2 (32K ADPCM)
	Line Number	449281
	Hotline Number	01223352781
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 2 OneTwo2	Members	0,1,3 0,1,4 (32K ADPCM)
	Line Number	449282
	Hotline Number	01223352782
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 3 TwoOne1	Members	0,1,5 0,1,6 (32K ADPCM)
	Line Number	449283
	Hotline Number	01223352783
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 4 TwoOne2	Members	0,1,7 0,1,8 (32K ADPCM)
	Line Number	449284
	Hotline Number	01223352784
	Other	Hotline mode End To End Use CM/RM-SAM
Lineset 1	Range	01223352700-01223352700
	Members	0,0,1 0,0,2 0,0,3 0,0,4
	Attributes	Non-Local
Lineset 2	Range	01223352701-01223352799
	Members	IPMN 3, 1-30
	Attributes	Non-Local

Node 2

Virtual Line 1 OneTwo1	Members	0,1,1 0,1,2 (32K ADPCM)
	Line Number	352781
	Hotline Number	01865449281
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 2 OneTwo2	Members	0,1,3 0,1,4 (32K ADPCM)
	Line Number	352782
	Hotline Number	01865449282
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 3 TwoOne1	Members	0,1,5 0,1,6 (32K ADPCM)
	Line Number	352783
	Hotline Number	01865449283
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 4 TwoOne2	Members	0,1,7 0,1,8 (32K ADPCM)
	Line Number	352784
	Hotline Number	01865449284
	Other	Hotline mode End To End Use CM/RM-SAM
Lineset 1	Range	01865449200-01865449200
	Members	0,0,5 0,0,6 0,0,7 0,0,8
	Attributes	Non-Local
Lineset 2	Range	01865449201-01865449299
	Members	IPMN 3, 30-1 (Reversed)
	Attributes	Non-Local

32K compressed manual connections

Only four channels (two Virtual Lines) each way are shown. To add capacity add more Virtual Lines.

Add one interwork unit (30 lines) to each node.

Configuration tip *Use Multi-Line configuration and Interwork Common configurations to speed up the configuration process.*

Node 1

Virtual Line 1 OneTwo1	Members	0,1,1 0,1,2 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 2 OneTwo2	Members	0,1,3 0,1,4 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 3 TwoOne1	Members	0,1,5 0,1,6 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 4 TwoOne2	Members	0,1,7 0,1,8 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Interwork 1 RHS 1 (0,1,1)	Label	AB#1
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 2 (0,1,2)	Label	AB#2
	End to End Number	1223352700
	Other	SPC Mode (AUS)Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 3 (0,1,3)	Label	AB#3
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 4 (0,1,4)	Label	AB#4
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 5 (0,1,5)	Label	AB#5
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End

Interwork 1 RHS 6 (0,1,6)	Label	AB#6
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 7 (0,1,7)	Label	AB#7
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 8 (0,1,8)	Label	AB#8
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Lineset 1	Range	01223352700-01223352700
	Members	0,0,1 0,0,2 0,0,3 0,0,4
	Attributes	Non-Local
Lineset 2	Range	01223352701-01223352799
	Members	IPMN 3, 5-30 (missing 1-4!)
	Attributes	Non-Local
Manual Connection 1	Between OneTwo1 and B1	8,1,0 and 3,0,1
	Type	External, non-volatile, bi-directional
Manual Connection 2	Between OneTwo2 and B2	8,1,1 and 3,0,2
	Type	External, non-volatile, bi-directional
Manual Connection 3	Between TwoOne1 and B3	8,1,2 and 3,0,3
	Type	External, non-volatile, bi-directional
Manual Connection 4	Between TwoOne2 and B4	8,1,3 and 3,0,4
	Type	External, non-volatile, bi-directional

Node 2

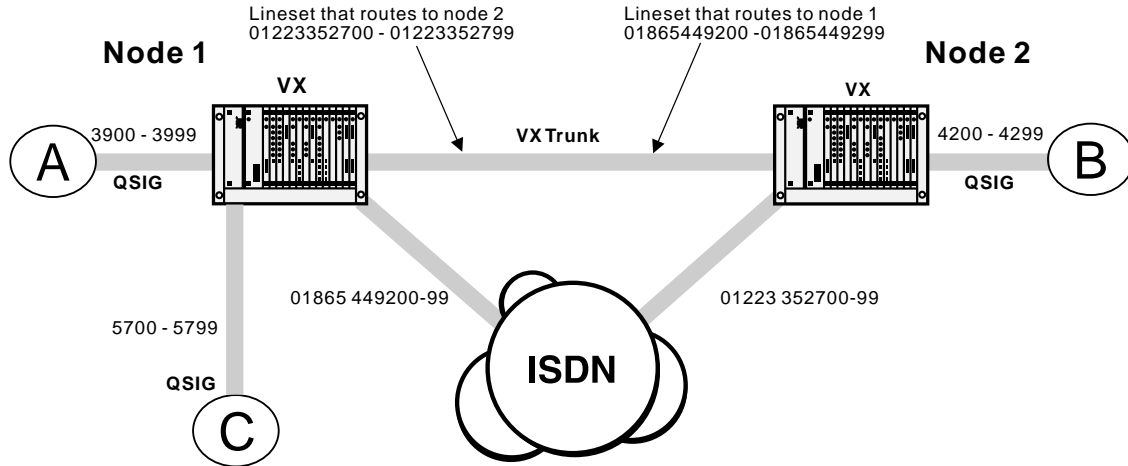
Virtual Line 1 OneTwo1	Members	0,1,1 0,1,2 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 2 OneTwo2	Members	0,1,3 0,1,4 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)

Virtual Line 3 TwoOne1	Members	0,1,5 0,1,6 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 4 TwoOne2	Members	0,1,7 0,1,8 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Interwork 1 RHS 1 (0,1,1)	Label	AB#1
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 2 (0,1,2)	Label	AB#2
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 3 (0,1,3)	Label	AB#3
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 4 (0,1,4)	Label	AB#4
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 5 (0,1,5)	Label	AB#5
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 6 (0,1,6)	Label	AB#6
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End

Interwork 1 RHS 7 (0,1,7)	Label	AB#7
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 8 (0,1,8)	Label	AB#8
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Lineset 1	Range	01865449200-01865449200
	Members	0,0,1 0,0,2 0,0,3 0,0,4
	Attributes	Non-Local
Lineset 2	Range	01865449201-01865449299
	Members	IPMN 3, 30-5 (missing 4-1!)
	Attributes	Non-Local
Manual Connection 1	BetweenOneTwo1 and B1	8,1,0 and 3,0,1
	Type	External, non-volatile, bi-directional
Manual Connection 2	BetweenOneTwo2 and B2	8,1,1 and 3,0,2
	Type	External, non-volatile, bi-directional
Manual Connection 3	BetweenTwoOne1 and B3	8,1,2 and 3,0,3
	Type	External, non-volatile, bi-directional
Manual Connection 4	BetweenTwoOne2 and B4	8,1,3 and 3,0,4
	Type	External, non-volatile, bi-directional

QSIG applications

This section details the configuration required for the following sample two node network.



PINX A can dial 3 types of numbers:

- 4200-4299 is used to reach extensions on PINX B.
- 5700-5799 is used to reach extensions on PINX C.
- Anything else is used to reach numbers on the public network.

Similarly, PINX B can also dial 3 types of numbers:

- 3900-3999 is used to reach extensions on PINX B.
- 5700-5799 is used to reach extensions on PINX C.
- Anything else is used to reach numbers on the public network.

General configuration

In this scenario the chassis, in addition to an RM or CM, and a Power Supply Module, contains the following modules:

- IPMN-B190 in slot 8
- IPMN-ETSI in slot 3
- E1M-QSIG in slots 5 and 6
- SAM in slot 10
- VCM in slot 15 (only required for the 32K examples).

The following instructions assume you are starting with a defaulted configuration in each node and subsequently lists only the things you have to change from default.

Node 1

Manager number	449298
Virtual eXchange	Min 449200 Max 449299 Area code 1865

Node 2

Manager number	352798
Virtual eXchange	Min 352700 Max 352799 Area code 1223

QSIG module configuration - connected locally (PINX A to PINX C)

Node 1

Lineset 1	Range	3900-3999
	Members	QSIG Slot 5 1-30*
	Attributes	Local
General settings	QSIG A/B configuration	A
	Exclusive/Preferred	Exclusive channel
	Min number length	4
Lineset 2	Range	5700-5799
	Members	QSIG Slot 6 30-1*
	Attributes	Local
General settings	QSIG A/B configuration	B
	Exclusive/Preferred	Exclusive channel
	Min number length	4

*Order of line set members corresponds to A/B configuration option chosen.

QSIG module configuration - connected across a network (PINX A to PINX B)

Three alternative configurations are listed:

- A 64K uncompressed network
- 32K ADPCM voice compression on demand
- 32K ADPCM manually connected.

The basic configuration is the same for each; only the Bandwidth Configuration is different.

The following instructions assume you are starting with a defaulted configuration in each node and subsequently lists only the things you have to change from default.

Node 1

Lineset	Range	3900-3999
	Members	QSIG 1-30*
	Attributes	Local
General settings	QSIG A/B configuration	A
	Exclusive/Preferred	Exclusive channel
	Min number length	4

Node 2

Lineset	Range	4200-4299
	Members	QSIG 30-1*
	Attributes	Local
General settings	QSIG A/B configuration	B
	Exclusive/Preferred	Exclusive channel
	Min number length	4

*Order of line set members corresponds to A/B configuration option chosen.

Number Translation Configuration

Node 1

Number Range name	OneTwo	
Numbers	4200-4299	Apply to QSIG called
Modify	Replace with	01223352700
Apply to	QSIG Module Slot 5	All lines

Node 2

Number Range name	TwoOne	
Numbers	3900-3999	Apply to QSIG called
Modify	Replace with	01865449200
Apply to	QSIG Module Slot 5	All lines

Signalling configuration

Router tables

Node 1

CM/RM-SAM	Number range	01223352700-01223352799
	Primary route	D-channel 1
	Secondary route	CM/RM inband
SAM	Number range	01223352700-01223352799
	Primary route	B-channel 2
	Secondary route	None

Node 2

CM/RM-SAM	Number range	01865449200-01865449299
	Primary route	D-channel 1
	Secondary route	CM/RM inband
SAM	Number range	01865449200-01865449299
	Primary route	B-channel 2
	Secondary route	None

Signalling channels

Node 1

CM/RM-SAM D-Channel 1	Line Number	449295
	Hotline number	01223352795
	Other	Hotline mode Auto Answer on Acknowledge off
SAM B-Channel 2	Line Number	449296
	Hotline number	01223352796
	Other	Hotline mode Don't use for i/c Manager calls

Node 2

CM/RM-SAM D-Channel 1	Line Number	352795
	Hotline number	01865449295
	Other	Hotline mode Auto Answer on Acknowledge off
SAM B-Channel 2	Line Number	352796
	Hotline number	01865449296
	Other	Hotline mode Don't use for i/c Manager calls

Bandwidth Configuration

64K Uncompressed

Node 1

Lineset	Range	01223352700-01223352799
	Members	IPMN 3, 1-30 *
	Attributes	Non-Local

Node 2

Lineset	Range	01865449200-01865449299
	Members	IPMN 3, 30-1 *
	Attributes	Non-Local

* **Note** End of link should have the channel order reversed to minimise channel collisions.

32K Compressed on demand

Only four channels (two Virtual Lines) each way are shown in this example. To add capacity add more Virtual Lines.

Add one interwork unit (30 lines) to each node.

Node 1

Virtual Line 1 OneTwo1	Members	0,1,1 0,1,2 (32K ADPCM)
	Line Number	449281
	Hotline Number	01223352781
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 2 OneTwo2	Members	0,1,3 0,1,4 (32K ADPCM)
	Line Number	449282
	Hotline Number	01223352782
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 3 TwoOne1	Members	0,1,5 0,1,6 (32K ADPCM)
	Line Number	449283
	Hotline Number	01223352783
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 4 TwoOne2	Members	0,1,7 0,1,8 (32K ADPCM)
	Line Number	449284
	Hotline Number	01223352784
	Other	Hotline mode End To End Use CM/RM-SAM
Lineset 1	Range	01223352700-01223352700
	Members	0,0,1 0,0,2 0,0,3 0,0,4
	Attributes	Non-Local
Lineset 2	Range	01223352701-01223352799
	Members	IPMN 3, 1-30
	Attributes	Non-Local

Node 2

Virtual Line 1 OneTwo1	Members	0,1,1 0,1,2 (32K ADPCM)
	Line Number	352781
	Hotline Number	01865449281
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 2 OneTwo2	Members	0,1,3 0,1,4 (32K ADPCM)
	Line Number	352782
	Hotline Number	01865449282
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 3 TwoOne1	Members	0,1,5 0,1,6 (32K ADPCM)
	Line Number	352783
	Hotline Number	01865449283
	Other	Hotline mode End To End Use CM/RM-SAM
Virtual Line 4 TwoOne2	Members	0,1,7 0,1,8 (32K ADPCM)
	Line Number	352784
	Hotline Number	01865449284
	Other	Hotline mode End To End Use CM/RM-SAM
Lineset 1	Range	01865449200-01865449200
	Members	0,0,5 0,0,6 0,0,7 0,0,8
	Attributes	Non-Local
Lineset 2	Range	01865449201-01865449299
	Members	IPMN 3, 30-1 (Reversed)
	Attributes	Non-Local

32K compressed manual connections

Only four channels (two Virtual Lines) each way are shown. To add capacity add more Virtual Lines.

Add one interwork unit (30 lines) to each node.

Configuration tip Use Multi-Line configuration and Interwork Common configurations to speed up the configuration process.

Node 1

Virtual Line 1 OneTwo1	Members	0,1,1 0,1,2 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 2 OneTwo2	Members	0,1,3 0,1,4 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 3 TwoOne1	Members	0,1,5 0,1,6 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 4 TwoOne2	Members	0,1,7 0,1,8 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Interwork 1 RHS 1 (0,1,1)	Label	AB#1
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 2 (0,1,2)	Label	AB#2
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 3 (0,1,3)	Label	AB#3
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End

Interwork 1 RHS 4 (0,1,4)	Label	AB#4
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 5 (0,1,5)	Label	AB#5
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 6 (0,1,6)	Label	AB#6
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 7 (0,1,7)	Label	AB#7
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 8 (0,1,8)	Label	AB#8
	End to End Number	1223352700
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Lineset 1	Range	01223352700-01223352700
	Members	0,0,1 0,0,2 0,0,3 0,0,4
	Attributes	Non-Local
Lineset 2	Range	01223352701-01223352799
	Members	IPMN 3, 5-30 (missing 1-4!)
	Attributes	Non-Local
Manual Connection 1	BetweenOneTwo1 and B1	8,1,0 and 3,0,1
	Type	External, non-volatile, bi-directional
Manual Connection 2	BetweenOneTwo2 and B2	8,1,1 and 3,0,2
	Type	External, non-volatile, bi-directional

Manual Connection 3	Between TwoOne1 and B3	8,1,2 and 3,0,3
	Type	External, non-volatile, bi-directional
Manual Connection 4	Between TwoOne2 and B4	8,1,3 and 3,0,4
	Type	External, non-volatile, bi-directional

Node 2

Virtual Line 1 OneTwo1	Members	0,1,1 0,1,2 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 2 OneTwo2	Members	0,1,3 0,1,4 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 3 TwoOne1	Members	0,1,5 0,1,6 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Virtual Line 4 TwoOne2	Members	0,1,7 0,1,8 (32K ADPCM)
	Other	SPC Mode (AUS) Manually Connected (UKS)
Interwork 1 RHS 1 (0,1,1)	Label	AB#1
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 2 (0,1,2)	Label	AB#2
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 3 (0,1,3)	Label	AB#3
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 4 (0,1,4)	Label	AB#4
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 5 (0,1,5)	Label	AB#5
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End

Interwork 1 RHS 6 (0,1,6)	Label	AB#6
	End to End Number	1865449200
	Other	SPC Mode (AUS Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 7 (0,1,7)	Label	AB#7
	End to End Number	1865449200
	Other	SPC Mode (AUS)Member of SRVL (UKS) Transparent End to End
Interwork 1 RHS 8 (0,1,8)	Label	AB#8
	End to End Number	1865449200
	Other	SPC Mode (AUS) Member of SRVL (UKS) Transparent End to End
Lineset 1	Range	01865449200-01865449200
	Members	0,0,1 0,0,2 0,0,3 0,0,4
	Attributes	Non-Local
Lineset 2	Range	01865449201-01865449299
	Members	IPMN 3, 30-5 (missing 4-1!)
	Attributes	Non-Local
Manual Connection 1	Between OneTwo1 and B1	8,1,0 and 3,0,1
	Type	External, non-volatile, bi-directional
Manual Connection 2	Between OneTwo2 and B2	8,1,1 and 3,0,2
	Type	External, non-volatile, bi-directional
Manual Connection 3	Between TwoOne1 and B3	8,1,2 and 3,0,3
	Type	External, non-volatile, bi-directional
Manual Connection 4	Between TwoOne2 and B4	8,1,3 and 3,0,4
	Type	External, non-volatile, bi-directional

Analog line module applications

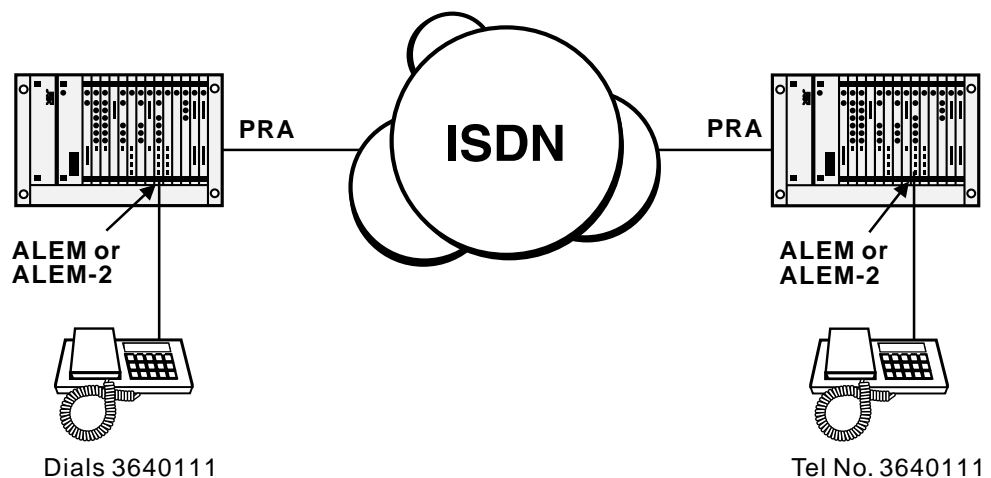
This section provides examples of using analog line modules in different scenarios. Each example details the type of module that can be used.

Notes *The ALEM and ALPM are not available in the UK.*

The ALIM-EXCH module on the COMBO can be used in the same applications as those described for the ALEM-2.

Indoor/outdoor handset extension using ALEM or ALEM-2

The figure below shows an application using the ALEM(-2) to provide extension line circuits for direct connection of handsets. Calls may be established between handsets through an ISDN. Either handset may initiate a call by generating a loop condition on the two-wire interface at the ALEM(-2). Destination handset address information is sent from the originating handset to the attached ALEM(-2) using decadic or DTMF dial signalling. The ALEM(-2) connected to the destination handset then generates a ring signal. When the destination handset answers the call, a bi-directional end-to-end voice frequency path is established.

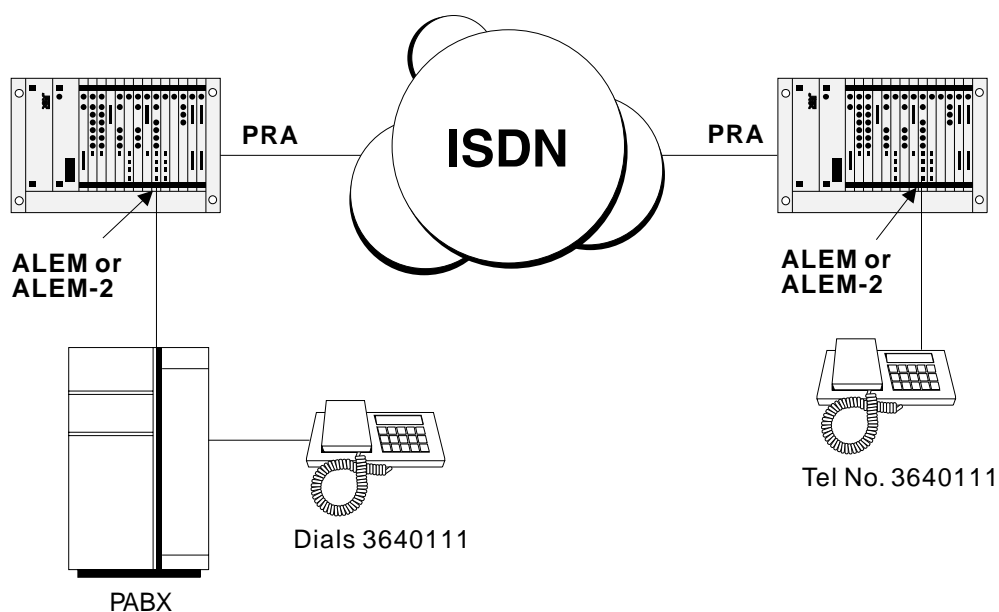


Remote Extension Circuit

(ALEM, ALEM-2, ALPM, ALPM-2)

The figure below shows an example application using the ALEM(-2) to provide a remote extension for a PABX through the ISDN. A call may be originated at the remote extension handset by generating a loop condition on the ALEM(-2) interface. Address information may then be forwarded to the ALEM(-2) by the originating handset using decadic or DTMF signalling. The call is then routed through the ISDN to an ALPM(-2) at the central PABX, which then passes the destination address information to the PABX. The PABX alerts the destination handset, and upon handset answer, generates a battery reversal signal to the ALPM (-2). This causes an end-to-end call path to be established.

A call to the remote extension may also be generated by the PABX equipment. In this case, the ring signal is generated by the PABX to the ALPM(-2). The ALPM(-2) Hotlines a call to the remote ALEM(-2), which then generates ring signal to the remote handset.

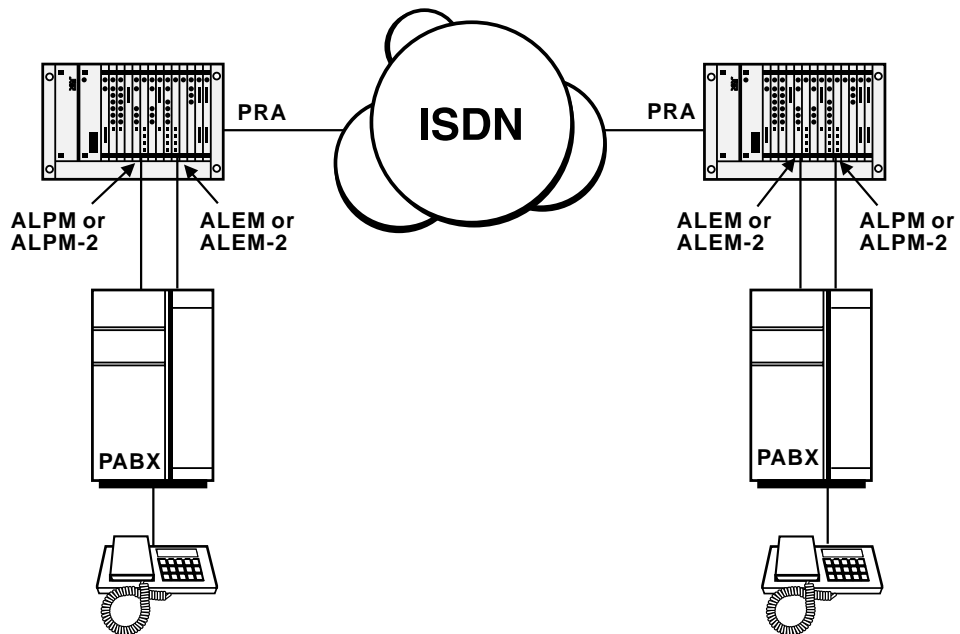


PABX-PABX Interworking

(ALEM, ALEM-2, ALPM, ALPM-2)

The figure below shows an example of PABX-PABX interworking through the ISDN. In order for bi-directional interworking, it is necessary for a combination of ALEM(-2) and ALPM(-2) to exist at each end of the network. Calls may be originated at either PABX by looping a line to an ALEM(-2), and then sending decadic or DTMF address information. The call is then routed through the ISDN to an ALPM(-2) at the destination end, which then dials address information into the destination PABX to cause the destination handset to be alerted. The destination PABX then indicates destination handset answer by generating battery reversal to the ALPM(-2), which in turn causes the ALEM(-2) to generate battery reversal to the originating PABX.

An active call may be terminated by either end. If the call is terminated at the destination end, then the destination PABX restores battery polarity to the ALPM(-2), which causes the ALEM(-2) to restore battery polarity to the originating PABX, thereby indicating destination party disconnect. If the call is terminated at the originating end, then the originating PABX generates a loop break condition to the ALEM(-2), which then causes the ALPM(-2) to generate loop break to the destination PABX.



PSTN Indial (Direct Dial In)

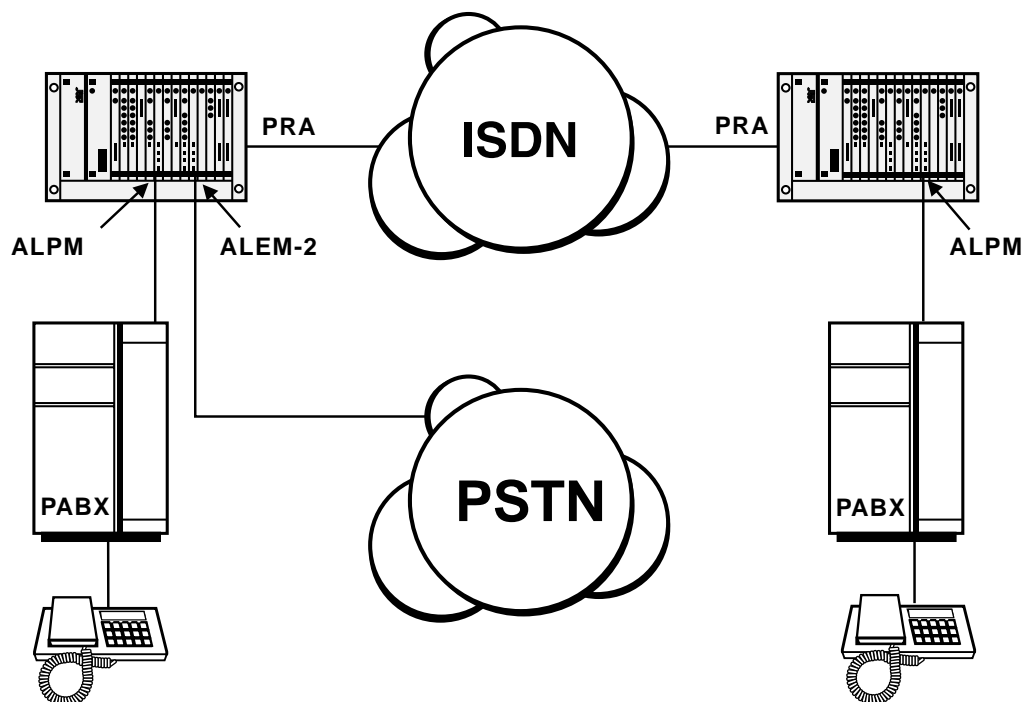
(ALEM-2)

The figure below shows an example of PSTN Indial operation. In this example, calls may only be incoming to the ALEM-2. Destination address information is passed from the PSTN to the ALEM-2 using decadic signalling (the ALEM-2 also supports DTMF signalling if it is supported by the PSTN). The call may then be routed through the node either directly or via the ISDN to an ALPM-2 connected to a destination PABX. Upon destination handset answer, an end-to-end call path is established.

A call is terminated from the PSTN when the PSTN breaks line loop to the ALEM-2.

Warning

The ALEM-2 and ALIM-EXCH are not permitted to be connected to PSTN lines in the UK or Australia.



Speaker Box Application

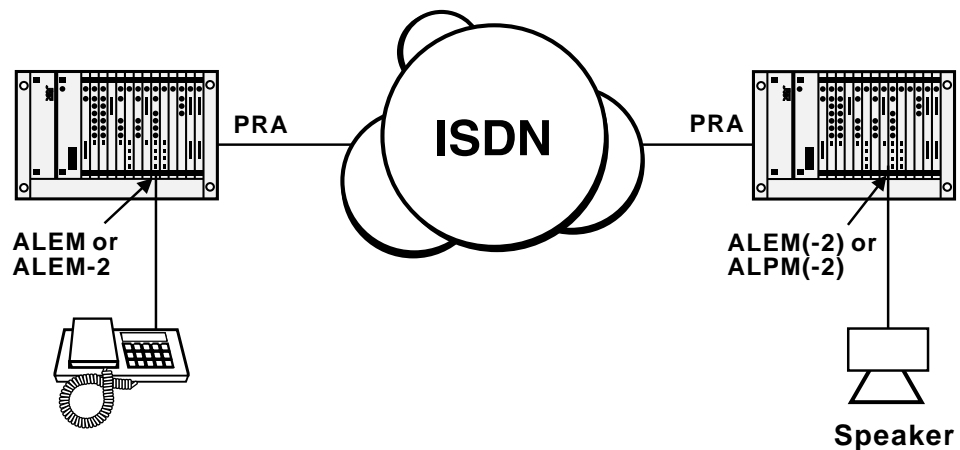
This application uses ALEM or ALPM modules to provide a speaker box, typically used in dealing rooms. The figure below shows how to connect the line modules to the system.

Using ALEM or ALEM-2

In this application, the destination ALEM(-2) is configured for auto-answer mode, which causes the ALEM(-2) to automatically answer incoming calls from the network side without generating ring signal to the speaker box. An end-to-end voice path is immediately established between call originating equipment and the speaker box.

Using ALPM or ALPM-2

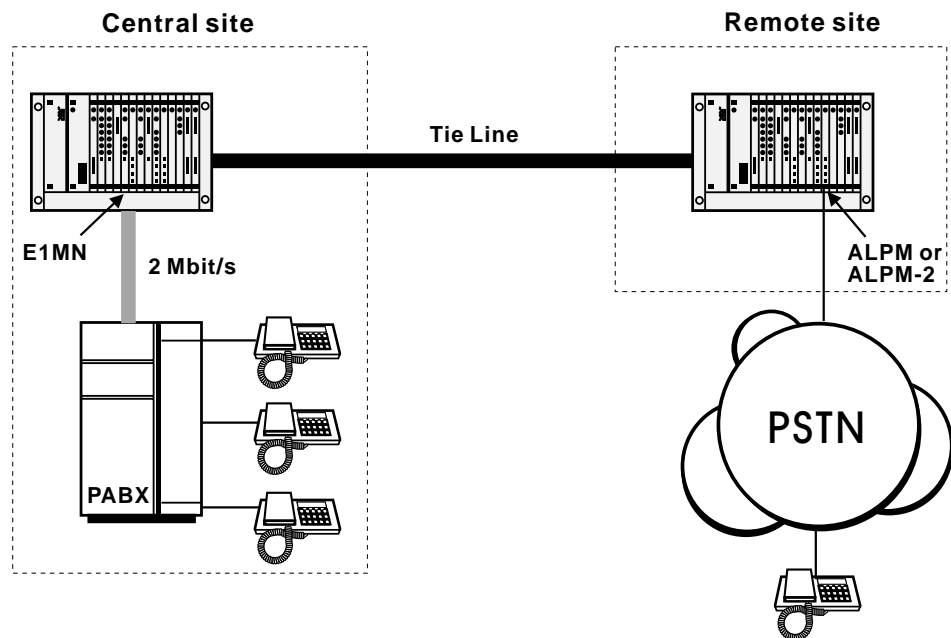
The destination ALPM(-2) is connected to a speaker box to provide a battery feed to its audio input line. The ALPM(-2) is configured for auto-answer mode, connect tone mode and dial type set to DTMF. The ALPM(-2) automatically answers incoming calls and loops the line. Two 'beeps' precede the voice path establishment.



Connection to PSTN Tie lines

(ALPM and ALPM-2)

The figure below shows the connection of an ALPM(-2) to a PSTN tie line.

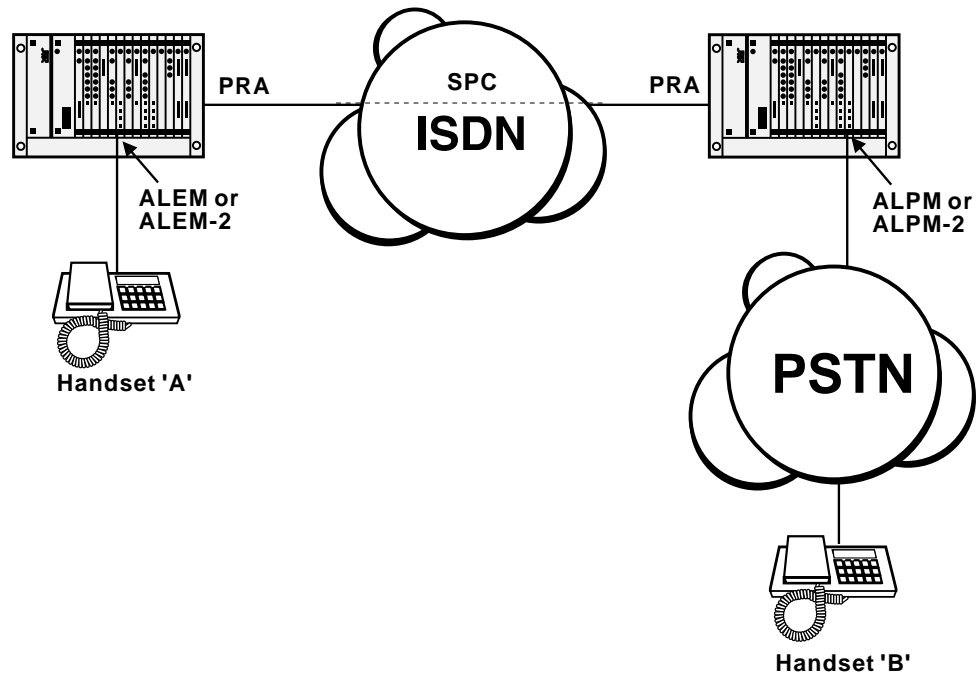


Calls originating in the PSTN will dial the Remote Site's PSTN number and be routed by the PSTN to the ALPM(-2). This will present a ring signal to the ALPM(-2) that will be carried across the ISDN to the Central Site's PABX causing a phone to ring. In the reverse direction, calls from the PABX are presented to the E1MN. This is then routed through ISDN to the Remote Site causing the ALPM(-2) to dial into the PSTN to generate ring to the remote handset.

End-to-end extension circuit

(ALEM, ALEM-2, ALPM, ALPM-2)

This example shows a handset having extension access to the PSTN via an ISDN SPC. Inband DTMF dialling is used with IDAS End-to-End mode to transfer two-wire signalling information across the ISDN D-channel or Inband on the B-channel.



D-channel signalling

When handset 'A' goes off hook, the ALEM(-2) sends an IDAS SETUP over the ISDN D-channel to the remote ALPM(-2). The ALPM(-2) will auto-answer and loop the PSTN line, thus providing a voice path for handset 'A' to hear PSTN tones/announcements and dial DTMF digits inband.

When handset 'B' causes the PSTN to ring into the ALPM(-2), the ALPM(-2) sends an IDAS SETUP over the ISDN D-channel to the remote ALEM(-2). The ALEM(-2) rings into handset 'A'.

Disconnections can either be made by handset 'A' or the PSTN (if it provides a call clearing mechanism using signalling such as battery reversal or battery disconnection).

Inband B-channel signalling

When handset 'A' goes off-hook, the ALEM(-2) sends an IDAS SETUP over the ISDN B-channel to the remote ALPM(-2). The ALPM(-2) will auto-answer and loop the PSTN line, thus providing a voice path for handset 'A' to hear PSTN tones/announcements and dial DTMF digits inband.

When handset 'B' causes the PSTN to ring into the ALPM(-2), the ALPM(-2) sends an IDAS SETUP over the ISDN B-channel to the remote ALEM(-2). The ALEM(-2) rings into handset 'A'.

Disconnections can either be made by handset 'A' or the PSTN (if it provides a call clearing mechanism using signalling such as battery reversal or battery disconnection).

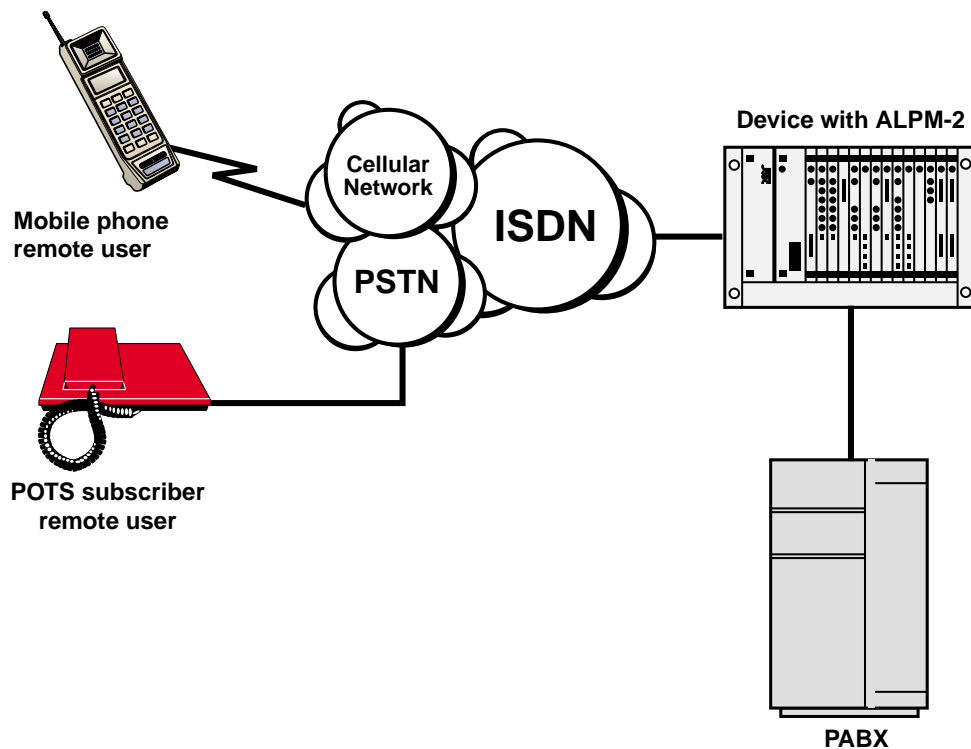
Virtual eXtension application

This application uses the ALPM-2 to provide remote POTS subscribers or mobile phone users access to a central PABX as if they were an extension directly connected to the PABX. During a call, the users have access to all the facilities offered to in-house analog extensions of the PABX, such as call hold, call transfer, call back when free and so on.

The Virtual eXtension feature is designed for one-to-one operation, that is, a single PABX extension port is attached to one ALPM-2 port and is associated with one POTS line or mobile phone.

The ALPM-2 port will always 'Hotline' to the same number in response to a ring applied from the PABX extension port. Equally, the POTS or mobile phone will always call the same line number of the ALPM-2 to gain access to the PABX.

The following diagram shows a typical setup for a Virtual eXtension application:



Operation

Calls to the PABX

The remote user dials the number of the ALPM-2 line. The ALPM-2 automatically connects the call and call charging commences. If the Access Password mode is enabled, a sequence of four short tones is heard. The tones prompt the user to enter the DTMF password. If the entered sequence of numbers and characters matches the preconfigured password on the ALPM-2 line, two tones are heard as confirmation of the correct password. The user then hears the PABX dial tone and can use the service as normal. The ALPM-2 will disconnect the call if the password is incorrect.

If the Access Password is not enabled the user hears the PABX dial tone immediately.

Calls from the PABX

When the ALPM-2 detects a ring from the PABX extension port, it initiates a call to a preconfigured 'Hotline' number. This Hotline number matches the number of the remote user.

The destination phone rings and, when answered, connects the call back to the ALPM-2 which in turn applies a loop to the PABX extension port, opening the line for a normal telephone conversation.

Configuration

The configuration of the Virtual eXtension involves the setting of up to four parameters in the ALPM-2. These are:

- Timed Break period
- Inband DTMF digit sequence as a Timed Break Request
- Inband DTMF digit sequence as a Disconnect Request
- Inband DTMF digit sequence as an Access Password.

These parameters can be set in the ALPM-2 Line configuration dialog.

Timed Break period

A user-configured timed loop break can be applied to the analog line in response to a user-to-user timed break (hookflash) request received from the far end of the connection. This is an alternative to the break duration set in the user-to-user message.

To configure:

- 1 In the ALPM-2 line configuration dialog box, choose **PABX**. This opens the Virtual eXtension Feature Control dialog box.
- 2 In the **Use break time of n x (10ms)** field enter the required break time in multiples of 10ms. This should ideally be set to suit the PABX.

Inband DTMF digit sequence as a Timed Break Request

During a call, the remote user can apply the timed break configured above. This is applied when the ALPM-2 detects an inband sequence of DTMF digits that match the user-configured string. This can only be configured if you have firstly configured the Timed Break period.

To configure the string:

- 1 In the ALPM-2 line configuration dialog box, choose **PABX**. This opens the Virtual eXtension Feature Control dialog box.
- 2 In the **DTMF digits to interpret as a Timed Break request** field, enter up to four digits from the range: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, #. These correspond to the keys on the remote user's telephone key pad.

During the call the remote user can dial the digits from the telephone key pad to initiate the configured timed break. To ensure correct validation of the digit sequence, the interval between dialled digits should not exceed three seconds.

Inband DTMF digit sequence as a Disconnect Request

At any time during a call, the remote user can disconnect the call by dialling a preconfigured series of digits. When the ALPM-2 detects an inband sequence of DTMF digits that match the user-configured string for the disconnect, the loop applied to the analog line is removed and the ALPM-2 clears the call towards the remote user.

To configure the string:

- 1** In the ALPM-2 line configuration dialog box, choose **PABX**. This opens the Virtual eXtension Feature Control dialog box.
- 2** In the **DTMF digits to interpret as a Disconnect request** field, enter up to four digits from the range: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, #. These correspond to the keys on the remote user's telephone key pad.

During the call the remote user can dial the digits from the telephone key pad to initiate the disconnection. To ensure correct validation of the digit sequence, the interval between dialled digits should not exceed three seconds.

Confirmation of the receipt of the valid digits is given by two tones immediately before the call is disconnected.

Inband DTMF digit sequence as an Access Password.

It is possible that anyone knowing the number of the ALPM-2 port could dial in to the PABX and use the facility to make unauthorised calls. This can be prevented by configuring an Access Password.

To configure a password:

- 1** In the ALPM-2 line configuration dialog box enter up to eight digits in the **Port Name** field. These must be from the range: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, #. These correspond to the keys on the remote user's telephone key pad.

When enabled, the password facility effectively provides a 'firewall' to the PABX. If a remote user dials in to the ALPM-2, the password feature is indicated by a four tone sequence. The user is then required to enter the DTMF digits that match the configured password in the Port Name field.

The remote user must dial the sequence with no more than 12 seconds between digits. An incorrect password results in the disconnection of the call. A correct password is confirmed by a two tone sequence followed by the PABX dialling tone.

Tips for using the Virtual eXtension feature

These tips provide advice on problems that you may encounter when using the Virtual eXtension feature.

Identical timed break and disconnect codes

It is possible that two connected users of the Virtual eXtension feature may have identical timed break or disconnect codes. The ALPM-2 cannot determine from which party the inband DTMF digits originate, so it is possible for a user on the PABX side of the call to accidentally trigger a timed break or a disconnect by dialling a specific series of digits.

You should try to make certain that all users apply different sets of codes for each feature. This will ensure that only the remote user can initiate a timed break or a disconnect.

Single user per ALPM-2 port

You should ensure that only one user is allocated the ALPM-2 port number as the ALPM-2 can only 'Hotline' to one destination.

Recorded Voice Announcements (RVA)

PABX users calling out to a Virtual eXtension will not receive RVAs indicating that the caller is out of range or the mobile phone is turned off. They will only hear a ring tone until the PABX cancels the call by timing out, or when they terminate the call.

Busy called party

PABX users calling out to a Virtual eXtension that is busy will hear a ring tone until they terminate the call or the PABX cancels it. Alternatively, they may hear a quick burst of ring tone followed by a 'called extension disconnect' tone. The format of the latter tone is dependent on the PABX characteristics.

Inter-digit Timeout

There is a two second inter-digit timeout interval when entering digits for the Disconnect and Hookflash, and a six second inter-digit timeout for the password digits. If these timers expire the part of the string already entered is discarded, and you will have to re-enter the whole string.

Entering the Timed Break Request

If the key for last digit of the Timed Break Request code is held down across the Timed Break, the PABX, after the Timed Break period, detects the held-down digit as the first digit to be dialled. For example, holding down the '9' of the Timed Break entry '##89' causes the PABX to detect and dial '9' immediately after the Timed Break. Avoid holding keys down when entering codes.

Mobile phone voice messaging systems

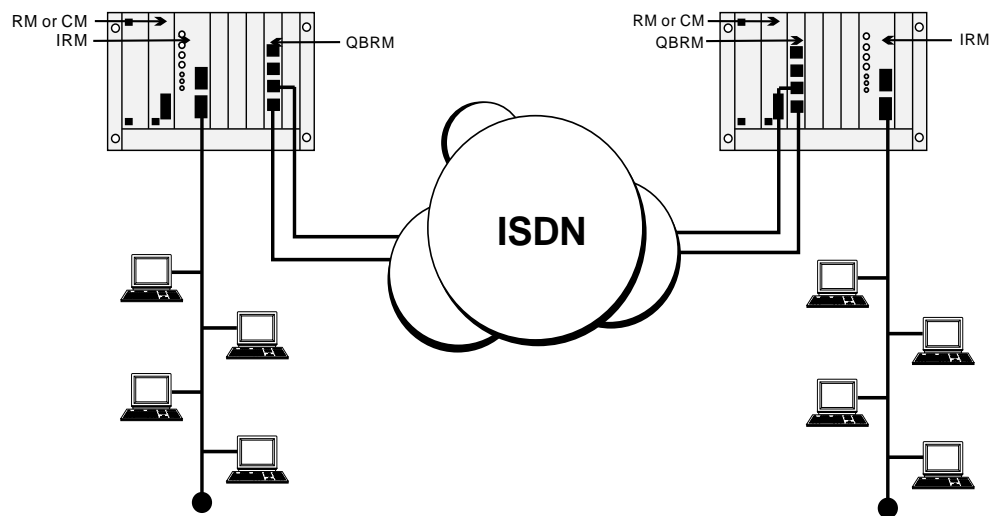
If a Virtual eXtension is a mobile phone that has a voice messaging system, any recorded voice messages will contain NU tone for a period of 30 to 90 seconds after the caller has left the message and has hung up. As the ALPM-2 cannot detect network tones and DTMF tones at the same time you should not use the **Release on Busy Tone** option in OmniVision.

IRM applications

This section describes several applications for the Integrated Router Module (IRM). For full details on the line module configuration aspects refer to page 4-215. To configure the Router/Bridge refer to the *IRM Router Configuration Manual*.

Multilinking IRM to IRM

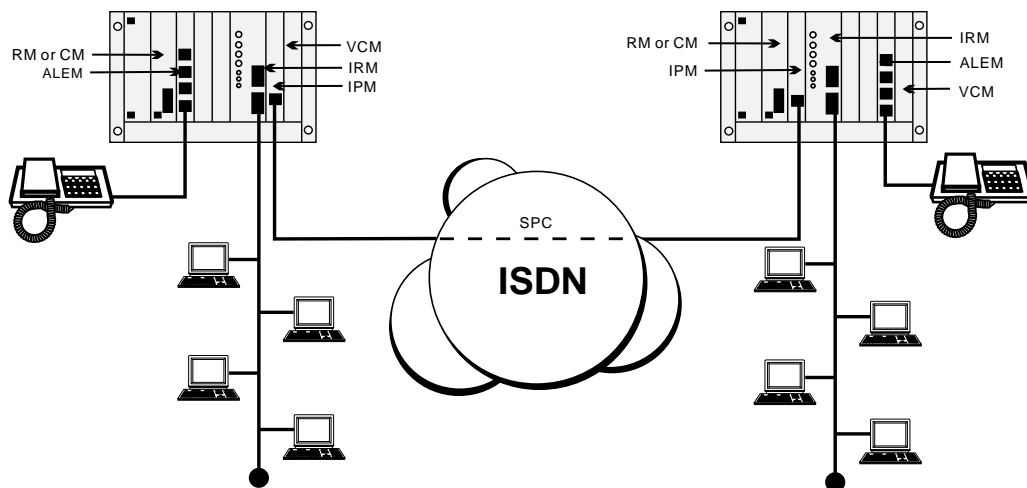
This application allows a WAN link greater than 64 kbit/s to be created without using an BCAM-ISO. Multilinking is a facility provided by the IRM that links one logical WAN port to multiple physical ports (see *IRM Router Configuration Manual* for more details).



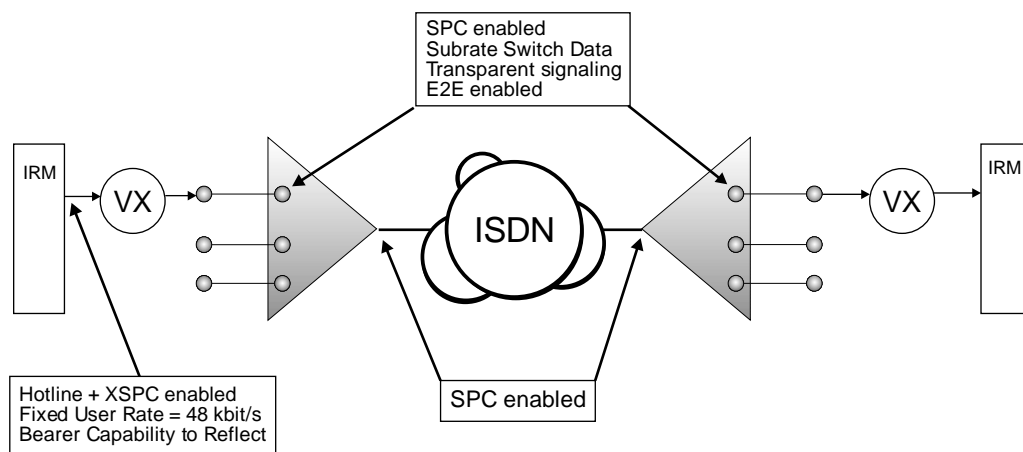
In this application the QBRM TEs are providing the ISDN connectivity between the two nodes. The call LEDs on the IRM front panels indicate that all WAN ports are active through the two Basic Rate links. The WAN throughput of 256 kbit/s is made up of all four WAN ports running at 64 kbit/s.

Subrate primary and bandwidth on demand

This application shows an IRM running a primary WAN at a Subrate speed, for example, 48 kbit/s. The other multi-linked WAN ports, configured to Bandwidth on Demand (BoD) raise and clear additional 64 kbit/s calls in response to detected traffic loads. If the primary link fails to maintain a data path, a BoD port raises a call until the primary link is restored.



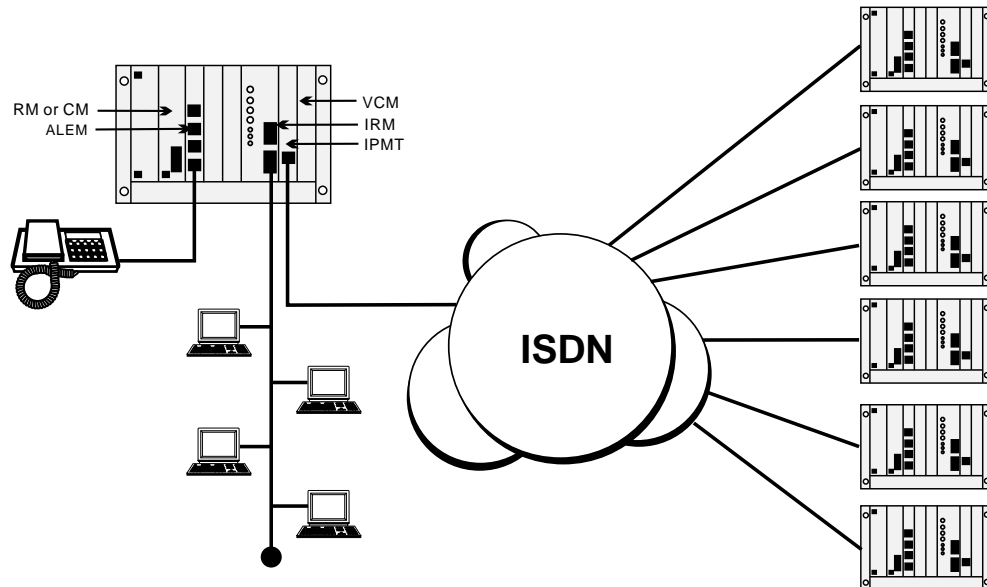
The IPMT provides the ISDN switched and SPC connectivity to the remote site. In this setup one primary WAN port and two other BoD WAN ports are activated to handle traffic load. A Subrate Virtual Line is configured to include both a 48 kbit/s IRM WAN port and a compressed voice circuit on the single 64 kbit/s permanent link. This concept is shown in the diagram below.



Linking multiple remote devices to a central site

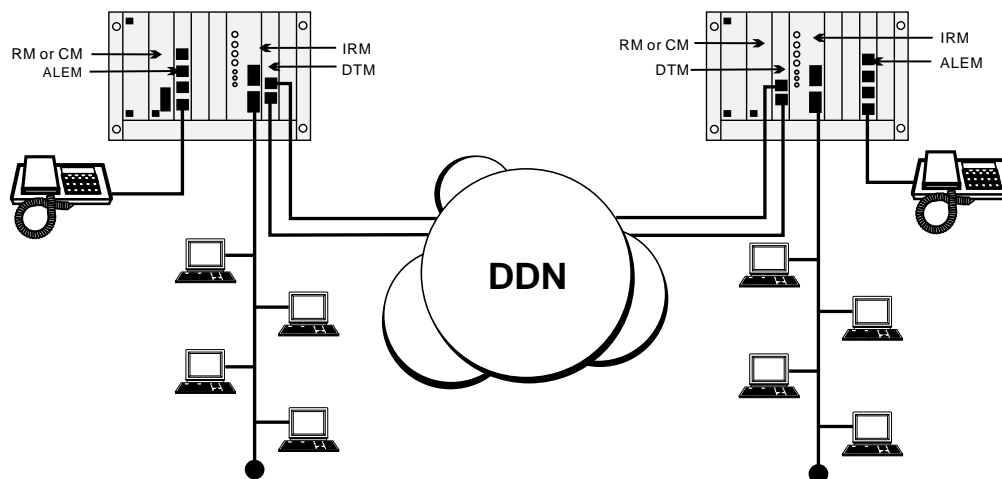
A central site is configured as a hunt group, and expects to receive calls from any one of a number of remote site devices. In this scenario, a given remote site always calls the same central site. Calls are initiated by significant data (for example, Unicast Frames generated by an IRM) and cleared after a set time due to inactivity.

When all central site WAN ports are busy, any incoming calls will receive a busy signal until a port becomes available.



Constant delay (with redundancy) — committed rate

This application shows an IRM multi-linking two dynamic, constant delay WAN connections to provide redundancy over the network. Each DTM trunk is clocking at 1Mbit/s and carries compressed voice as well as the IRM traffic. Each IRM WAN port has a maximum data rate capacity of 512 kbit/s.



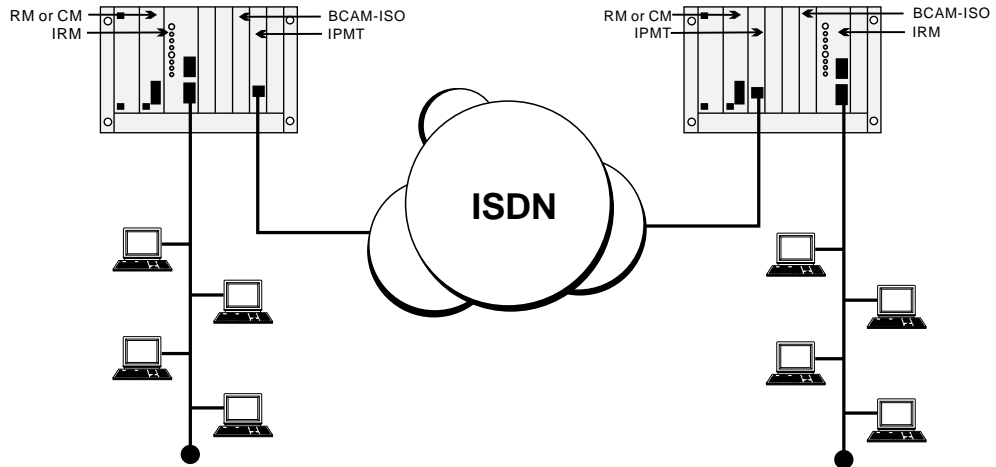
The DTM is configured to reserve the first two lines of each trunk for access only by the IRM WAN ports. The remainder of the trunk capacity is available as a resource for the other modules in the device. This means that the IRM has a 'committed' data rate of 128 kbit/s on each trunk. Dynamic call negotiation beyond 128 kbit/s is dependent on channel availability. The dynamic multi-linked WANs should be configured identically to evenly distribute the dynamic bandwidth.

In this scenario, the two multi-linked, manual call mode WAN ports are active. The XSPC maintains the initial call, and dynamic bandwidth management controls the speed of WAN ports, based on the detected variation in the traffic flow.

IRM with BCAM-ISO

Connecting LANs directly

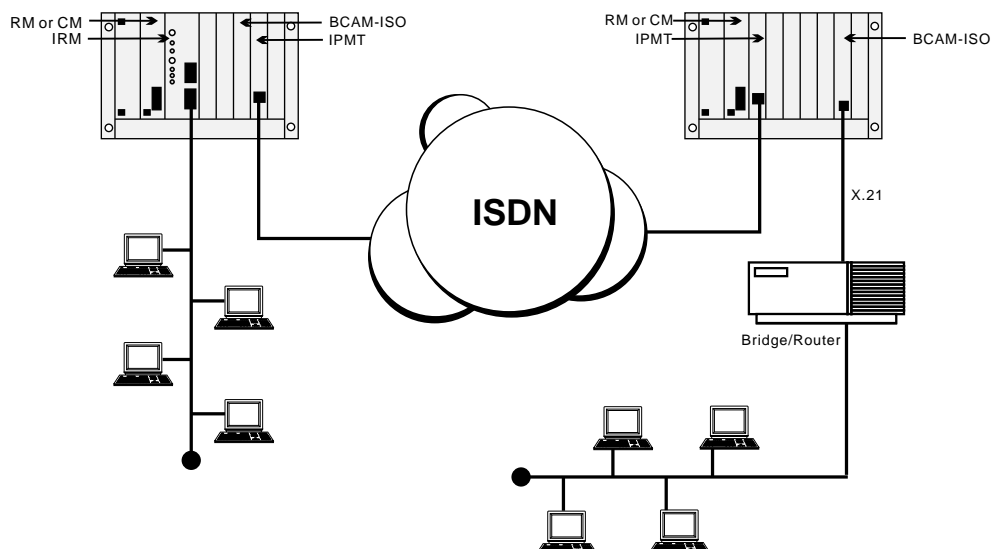
In this application a WAN link greater than 64 kbit/s is created by using the IRM in conjunction with an BCAM-ISO. Note that only a single Group A WAN port is needed to obtain a WAN speed of up to 1024 kbit/s.



The IPMTs provide ISDN connectivity between the two nodes. The call LEDs on the IRM front panels indicate that only one Group A WAN port is operating. The WAN throughput of 1024 kbit/s is achieved by configuring the BCAM-ISO to run in Mode 1 and to connect to the IRMs on the chassis backplanes. Alternatively, Mode 3 can be used at a speed of 960 kbit/s. In this mode the BCAM-ISO determines the quality of the data path and can re-establish the ISDN calls if the quality is poor.

Connecting LANs via a third-party Bridge /Router

This example shows how an IRM WAN link greater than 64 kbit/s uses the BCAM-ISO to connect to a third-party Bridge/Router using X.21. Here, only one Group A WAN port is needed to achieve a WAN speed up to 1024 kbit/s. The remote BCAM-ISO connects the data path to an X.21 front panel port.

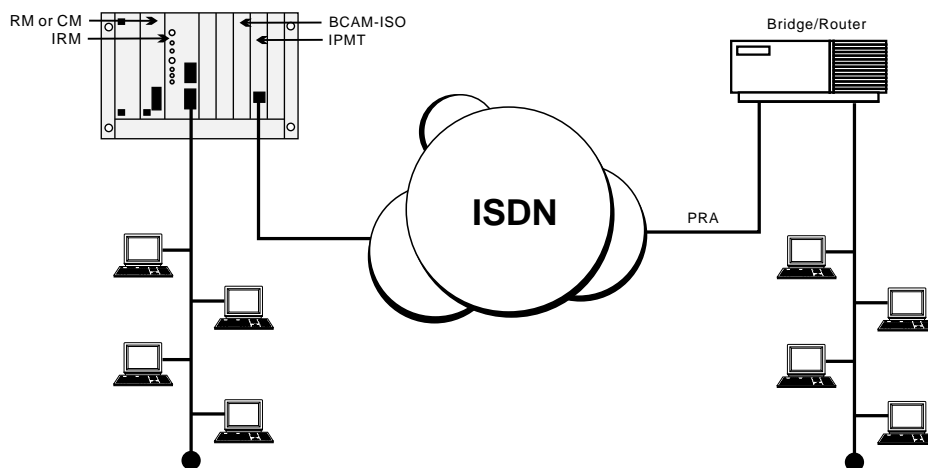


The IPMTs provide ISDN connectivity between the nodes. The call LEDs on the IRM front panels indicate that only one Group A WAN port is operating. The WAN throughput of 1024 kbit/s is achieved by configuring the BCAM-ISO at the local site to operate in Mode 1 and to connect to the IRM on the chassis backplane. At the remote site, the BCAM-ISO is configured to operate in Mode 1 and to connect to the X.21 front panel port.

Alternatively, for both BCAM-ISOs, Mode 3 can be used at a speed of 960 kbit/s so that the BCAM-ISO recognises whether the data path is of poor quality and can re-establish the ISDN call.

Connecting LANs through a single node and a third-party Bridge/Router

This application shows how an IRM WAN link greater than 64 kbit/s uses the BCAM-ISO to connect to a third-party Bridge/Router with an ISO standard channel aggregation capability and an ISDN interface. Here, a single Group A WAN port is needed to achieve a WAN speed of up to 1024 kbit/s.

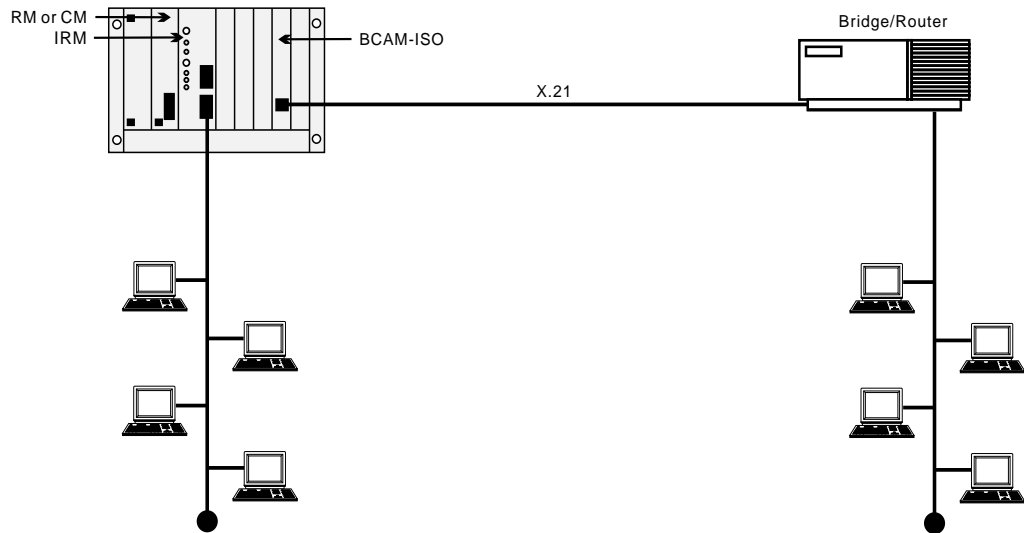


The IPMT provides the ISDN connectivity to the remote Bridge/Router. The call LEDs on the IRM front panel indicate that only one Group A WAN port is active. The WAN throughput of 1024 kbit/s is achieved by configuring the BCAM-ISO to operate in Mode 1 and connecting to the IRM on the chassis backplane.

Alternatively, Mode 3 can be used at a speed of 960 kbit/s so that the BCAM-ISO recognises if the data path is of poor quality and can re-establish the ISDN calls.

IRM to transparent mode BCAM-ISO

This application shows how an IRM WAN link, greater than 64kbit/s, uses the X.21 BCAM-ISO in transparent mode to connect to a third-party Bridge/Router.

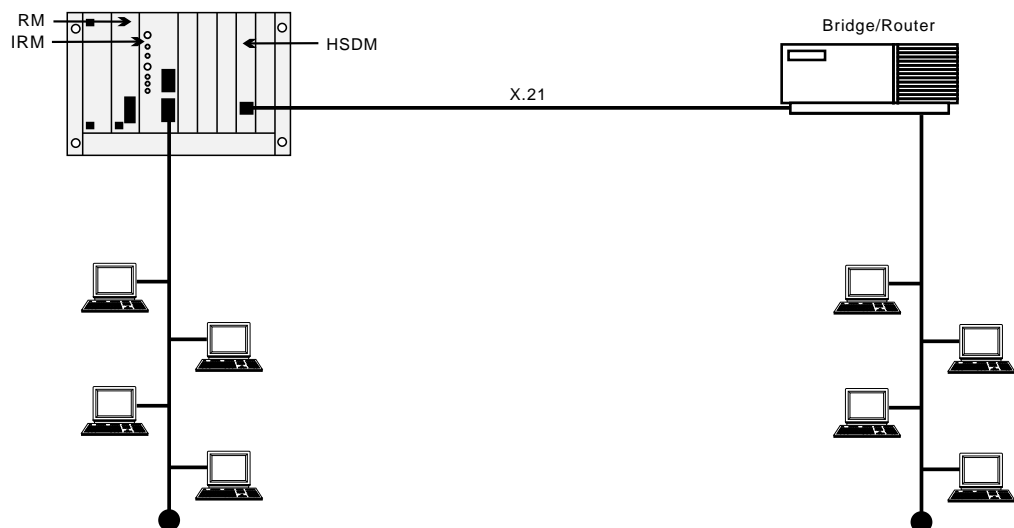


Only a single Group A WAN port is needed to achieve a WAN speed up to 1.024 Mbit/s. The BCAM-ISO interconnects the data path locally to an X.21 front panel port.

When connecting to an IRM port, **Manual Connect (Backplane)** must be selected in the BCAM-ISO configuration.

IRM with a High Speed Data Module (HSDM)

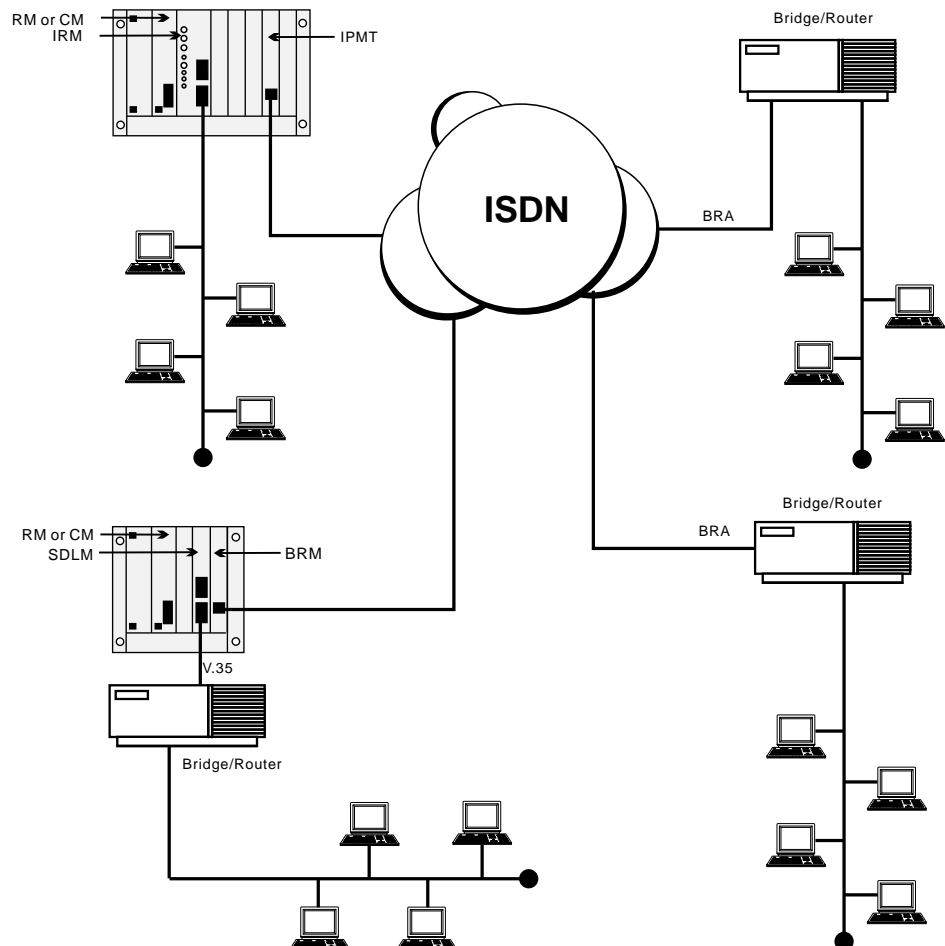
This application shows how an IRM WAN link, greater than 64kbit/s, uses an HSDM to connect to a third-party Bridge/Router.



Only a single Group A WAN port is needed to achieve a WAN speed up to 1984 kbit/s. The HSDM interconnects the data path locally to an X.21 front panel port.

IRM with a Digital Line module (DLM)

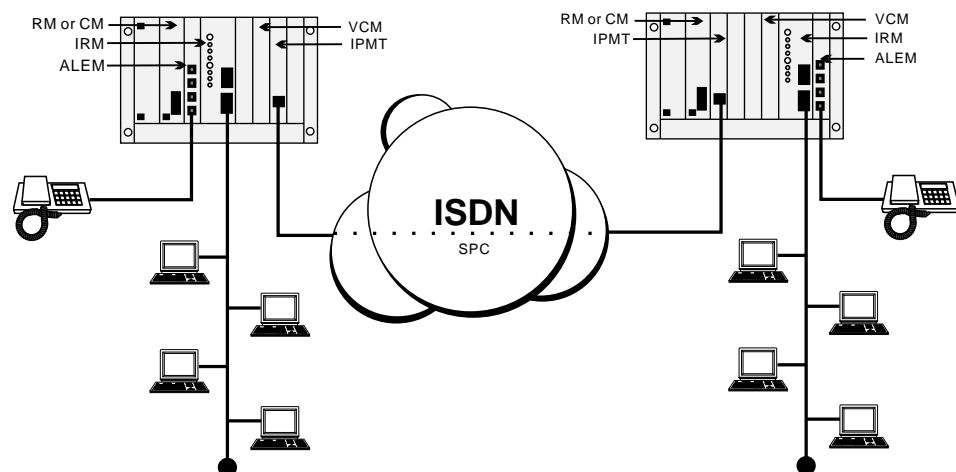
This application shows how a central site can use the IRM WAN links at 64 kbit/s and less using a DLM to connect to a V.35 third-party Bridge/Router through ISDN. Only one WAN port is connected to the remote V.35 SDLM. The other two are connected to the ISDN third-party Bridge/Routers.



The IPMT provides the ISDN connectivity to the remote sites. The call LEDs on the IRM front panel indicate that three WAN ports are active. The central IRM routes traffic to all remote networks and provides a common central node.

Subrate IRM and compressed voice

This application uses Subrate multiplexing and voice compression together with IRMs running at Subrate speeds (for example, 32kbit/s). Only one IRM WAN port is used with a compressed voice circuit. The principles shown in the previous applications that use non-aggregated case are still valid.



The IPMT provides the ISDN SPC (or XSPC) connectivity to the remote site. The call LEDs on the IRM front panels indicate that one WAN port is active. A Subrate Virtual Line is configured to include both a 32 kbit/s IRM WAN port and a compressed voice circuit on the one 64 kbit/s permanent link.

Dial Gateway application

The Dial Gateway is a solution that provides for remote dial-up TCP/IP applications such as Internet and Intranet. The Dial Gateway satisfies a variety of remote access needs including telecommuting, remote office, and Internet/Intranet access.

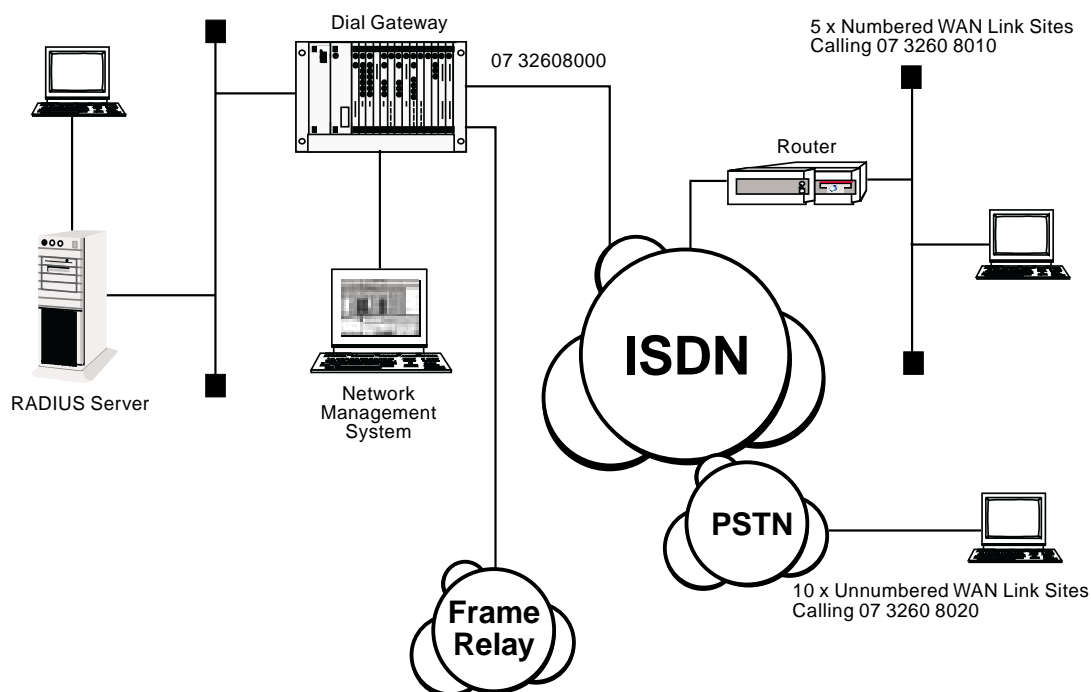
The system provides the power of remote access to a wide range of equipment including:

- analog PSTN modems
- GSM mobiles
- ISDN PC cards or terminal adaptors
- remote LANs via routers.

The Dial Gateway supports 30 remote users who are accessing the system using PSTN modems or ISDN digital connections. It can be expanded to provide support for up to 120 remote users including up to 90 analog PSTN users. The amount of expansion depends on slot availability and the configuration of the system.

The Dial Gateway includes an ISDN interface for connecting to the carrier's ISDN (IPMT), a 32 port TCP/IP router supporting PPP and Frame Relay WAN protocols (FSM), 30 digital modems (DMM), plus a Network Management System such as OmniVision.

This diagram below shows a typical Dial Gateway application.



Configuring the Dial Gateway

Note *The following information assumes that you are familiar with the modules that make up the Dial Gateway package (IPMT, FSM, DMM and respective PSM).*

The following steps are typical of those necessary to connect and configure a Dial Gateway,

- 1 Connect the Dial Gateway to the power source. Refer to the *Installation Manual* for details.
- 2 Connect the following interfaces:
 - The ISDN Primary Rate Module (IPMT) to the ISDN Primary rate NT1 provided by your carrier.
 - The 10BaseT port of the Frame Switch Module (FSM) to the Ethernet hub.
 - You can optionally connect the high speed Frame Relay port on the front panel of the FSM to the Frame Relay NTU provided by your carrier. This port can provide a high speed (up to 2Mbit/s) connection to the Internet backbone provided by your carrier. Alternatively it can be connected to an upstream ISP's router.
 - See the respective chapters in the *Installation Manual* for further details.
- 3 Install the network management system. See the *OmniVision User Manual* for details.
- 4 Configure the IPMT:
 - Enter the Group Number and Area code given by your ISDN provider. See IPMT configuration on page 4-3 for further details.
- 5 Configure the FSM Ethernet port:
 - Enter an IP Address. See FSM configuration on page 4-227 for further details.
- 6 Configure the FSM WAN ports:
 - Enter an IP Address for each WAN port. See FSM *IP addressing and Routing* on page 4-229 for details.
 - Set up Authentication if required. See *Login and Authentication* on page 4-233 for details.

Note *We recommend that you keep similarly configured ports contiguous. For example, you may set WAN ports 1 to 5 as unnumbered ports using CHAP authentication, ports 6 to 10 as numbered ports using text login authentication, and ports 11 to 30 as unnumbered ports using PAP authentication. Keeping similarly configured ports in groups aids call routing and troubleshooting.*

- 7 Configure RADIUS. See *RADIUS* on page 4-232 for details.

8 Configure Call Routing.

The simplest method for configuring call routing is to create Linesets which direct calls to the appropriate FSM ports. Linesets provide the ability to have multiple WAN ports exist as a rotary group for a single number. We recommend that you group similarly configured ports in a single lineset. For example, Unnumbered ports running CHAP would all exist in a single lineset while Numbered ports running CHAP would be members of a different lineset. See *Linesets* on page 3-17 for details.

Note *If there are no DMM modem resources free, calls are not answered. In this instance, the calls are rejected with cause 'User Busy'. This avoids a flag-fall charge.*

After these basic steps, the Dial Gateway is ready for use.

Example configuration

The following information summarises the configuration and provides the various entries to configure the typical network shown on page 5-99.

ISDN service

The ISDN Primary Rate Interface supplied by the ISDN provider has the group number 07 3260 8000 and includes a 100 number range.

Group Number 07 3260 8000

Number Range 07 3260 8000 - 07 3260 8099

The IPMT should be configured with a Group Number of 3260 8000 and the National Significant Area Code is 7.

WAN links

The example shows a mixture of Numbered and Unnumbered WAN link sites. We recommend that you group similarly configured ports in a single lineset. For example, the Unnumbered WAN link sites would call a different number and be routed by a single lineset while the Numbered WAN link sites would each need to call an individual number (so that remote and local WAN IP addresses correspond). Linesets should still be used to configure the Numbered WAN links because it provides a single configuration point for all.

In this example, a number within the PRA number range has been chosen for each of the Numbered WAN links to call, while a single number has been selected for the Unnumbered WAN links.

Numbered WAN Links

Port # 1 = 3260 8011

Port # 2 = 3260 8012

Port # 3 = 3260 8013

Port # 4 = 3260 8014

Port # 5 = 3260 8015

Unnumbered WAN Links:

Ports # 6 to 15 = 3260 8020

Configure Linesets**Numbered WAN Links Linesets**

Port	Parameter	Setting
Port 1	Minimum number	32608011
	Maximum number	32608011
	Member of lineset	fsm line 1
	Attributes for destination type	Local
Port 2	Minimum number	32608012
	Maximum number	32608012
	Member lines	fsm line 2
	Attributes for destination type	Local
Port 3	Minimum number	32608013
	Maximum number	32608013
	Member lines	fsm line 3
	Attributes for destination type	Local
Port 4	Minimum number	32608014
	Maximum number	32608014
	Member lines	fsm line 4
	Attributes for destination type	Local
Port 5	Minimum number	32608015
	Maximum number	32608015
	Member lines	fsm line 5
	Attributes for destination type	Local

Unnumbered WAN Links Lineset

Ports	Parameter	Setting
Ports 6 to 15	Minimum number	32608020
	Maximum number	32608020
	Member lines	fsm line 6 fsm line 7 fsm line 8 fsm line 9 fsm line 10 fsm line 11 fsm line 12 fsm line 13 fsm line 14 fsm line 15
	Attributes for destination types	Local



Channel Bank application

Overview

Channel Banks are devices that provide connectivity between analog telephone interfaces, and G.703/G.704 E1 and T1 interfaces. They provide internal mapping of an analog interface to a channel of either an E1 or T1 digital interface. This may be with or without their various signalling capabilities. The E1 and T1 interfaces operate in Channel Associated Signalling (CAS) or Robbed Bit Signalling (RBS) modes, respectively.

This section only deals with Channel Banks that provide point-to-point connections to other Channel Banks over a typical core network. The Channel Bank (CB) analog interfaces described here support three modes of connection across a variety of 2-wire and 4-wire interfaces. The connection modes are:

- Automatic Ring Down (ARD)
- Manual Ring Down (MRD)
- Tone Only (TO)

Automatic Ring Down (ARD)

ARD is a basic ring-out/loop-in 2-wire battery interface. The circuit operates in a similar way to a Hotline call. The far end Channel Bank provides ring tone to the near end. Communication is via telephone handset or microphone-headset.

Manual Ring Down (MRD)

MRD is either a 2-wire or a 4-wire 600 Ω dry interface for speech transfer and signalling. The speech path is open for two-way communication at all times, except when signalling occurs. The term 'GEN-GEN interface' is often used for this type of connection. The alerting device, at the near end, is activated locally during the application of the calling signal to the far end. Communication is often through a personal loudspeaker. However, multiple personal loudspeakers, public loudspeakers, a telephone handset or a microphone-headset may also be used.

Tone Only (TO)

TO is either a 2-wire or a 4-wire 600 Ω dry interface for speech transfer. No signalling is provided. The speech path is open for two-way communication at all times. TO is effectively MRD without the signalling. Communication is usually through a public loudspeaker, although a single personal loudspeaker or multiple personal loudspeakers may also be used.

Types of Channel Banks

This section deals with three known types of Channel Bank:

- N.E.T. ADNX (T1 interface)
- Telco Systems Channel Bank (T1 interface)
- Craycom Channel Bank (E1 interface).

There are three point-to-point combinations for each of the ARD, MRD and TO circuits:

- Channel Bank (E1) to Channel Bank (E1)
- Channel Bank (T1) to Channel Bank (T1)
- Channel Bank (E1) to Channel Bank (T1).

The signalling provided by the analog interfaces is converted to AB bit signalling for transport via the T1/E1 digital network.

Note *The core network must provide signalling mapping between T1 and E1 interfaces.*

ARD Channel Bank Signalling

The signalling states for ARD are IDLE, SEIZE, ALERTING, ANSWER, CLEAR FORWARD, CLEAR BACKWARD. Only two signals are used to represent the state:

- On-hook Signal
- Off-hook Signal.

ADNX and Telco

The ADNX (T1) and Telco Systems (T1) Channel Banks use the following RBS AB bit states for ARD connections:

- On-hook Signal: AB = 11
- Off-hook Signal: AB = 00

Only the input A bit is evaluated for signal change. This signalling is called PLAR (Private Line Automatic Ring-down).

Craycom

The Craycom (E1) Channel Bank uses the following CAS AB bit states for ARD connections:

- On-hook Signal: AB = 11
- Off-hook Signal: AB = 01

Only the input A bit is evaluated for signal change. This signalling is called SSDC5A.

MRD Channel Bank Signalling

The signalling states for MRD are RING OFF and RING ON. The two signals used to represent the state are the ring-off signal and the ring-on signal.

ADNX and Telco

The ADNX (T1) and Telco Systems (T1) Channel Banks use the following RBS AB bit states for MRD connections:

- On-hook Signal: AB = 00
- Off-hook Signal: AB = 11

Only the input A bit is evaluated for signal change. This signalling is called E&M.

Craycom

The Craycom (E1) Channel Bank uses the following CAS AB bit states for MRD connections:

- On-hook Signal: AB = 11
- Off-hook Signal: AB = 01

Only the input A bit is evaluated for signal change. This signalling is called SSDC5A.

Channel Bank Emulation

ARD support

This equipment supports ARD via an on-demand Hotline connection between an ALEM Line and a T1M Line or between an ALEM Line and an E1M Line. Hotline operation allows ring tone to be supplied to the near end caller when the device is acting as the far end.

MRD support

This equipment supports MRD via a UDAS protocol, XSPC Hotline between an EMM Line and a T1M Line, or between an EMM Line and an E1M Line. An external adapter converts between the 4-wire E&M interface of the EMM and 2-wire 'GEN-GEN interfaces'. Use of the UDAS protocol allows accurate transfer of the user-controlled ring cadence between the two ends.

TO support

This equipment supports TO via a permanent manual connection between an EMM Line and either a T1M Line or an E1M Line. An external hybrid is used to convert the 4-wire speech interface provided by the EMM to a 2-wire interface. The E and M leads of the EMM are not connected. Manual connections provide permanent, two-way voice paths and disregard any spurious signals that occur on the signalling input lead of the EMM ports.

Note *Both Channel Banks and our equipment use a variety of signalling to provide ARD and MRD connections over T1 and E1 interfaces. Mapping of the signalling, to provide compatibility, must be performed in the core network.*

For more detailed information on E1 and T1 signalling associated with Channel Banks, refer to *Appendix O*.

Configuring the equipment for a Channel Bank application

This section details the configuration for each module used in a particular Channel Bank scheme.

Module capabilities

The T1M supports any combination of ARD, MRD and TO circuits, on a channel-by-channel basis.

The E1M supports a combination of ARD and TO circuits, on a channel-by-channel basis, from a given module. It also supports a combination of MRD and TO circuits, on a channel-by-channel basis, from a given module. However, it does not support both ARD and MRD circuits on the same module.

Call establishment method

This equipment emulates a Channel Bank by using general switching functions and specific signalling features:

- ARD connections are established between ALEM and T1M/E1M Line pairs as on-demand Hotline connections.
- MRD connections are established between EMM and T1M/E1M Line pairs as XSPC Hotline connections.

Note *Line pairs in this context are defined by configuring two Lines to Hotline to each other.*

This requires the allocation and configuration of a numbering plan in each device. A numbering plan of three digits or more will be adequate for this purpose. However, a 7-digit number will minimise confusion and simplify configuration. For example, a plan with the number range 5550100 through 5551599 could be used. Using this plan, Line numbers are allocated according to module card slot number and port or channel number.

Note *Line numbers are not permitted to have zero as the first digit. Only Linesets can accommodate numbers commencing with zero.*

Module placement

For ease of configuration and to maintain symmetry, we recommend that you follow these general rules:

- Install the CM or RM in slot 15
- Install ALEMs and EMMs in slots 14 and down
- Install T1Ms and E1Ms in slots 1 and up.

This module arrangement ensures the efficient use of the backplane TDM channel allocation. It also provides a convenient system for allocating numbers. For example, the Line number of the third port of an ALEM in slot 12 becomes 5551203 and the Line number of the 17th channel of a T1M in slot 5 becomes 5550517.

Speech encoding

Encoding and decoding of analog speech to digital format is performed in codecs on the ALEM and EMM modules. Both modules support μ -law or A-law encoding on per-port basis. The coding used by a given port is selected during the Line configuration.

Note *The core network must provide μ -law/A-law conversion where necessary.*

Supervisory Tones

Supervisory tones, that is, Ring Tone, Fast Busy Tone and so on, are generated on the Control Module or Resource Manager. The tones are generated by a Tone ROM on the CM and via software on the RM. The tones generated vary according to country and market requirements, providing different cadences with either μ -law or A-law encoding.

Refer to the *Installation Manual* for information on each country variant for the CM Tone ROM.

Note *It is important that the CM is fitted with the correct Tone ROM for the target listener. Ensure the correct Tone ROM is fitted whenever changing a CM. The RM is capable of generating all tones without the need to change hardware.*

Configuring the system parameters

The following are the minimum requirements necessary to ensure functional operation of a stand-alone device for Channel Bank operation.

Configure the CM or RM Configuration parameters

- 1 Enter a **Manager Access Number**. (e.g. 5551598)
- 2 Enter a **Test Line Number**. (e.g. 5551599)

- 3 Configure the Signalling Access (CM-SAM/RM-SAM), **Inband Signalling Router Table** and Enable **Default use D channel**.

Configure the VX Configuration parameters:

- 1 Enter a **Minimum Number** and **Maximum Number**. (e.g. Min = 5550100, Max = 5551599). Do not enter an Area Code.

Confirm Country Code settings

- 1 Select **Version Display** from the **Device** menu.
- 2 Confirm that the correct Country Codes are set for your application.

Automatic Ring Down (ARD) parameters

The following are the minimum requirements necessary to ensure functional operation of a stand-alone device for ARD Channel Bank operation.

T1M (ARD)

Configure the **T1M Card Control** parameters:

- 1 Set **Framing** and **Zero Suppression** as appropriate for the connected equipment.

Configure the **Line Configuration** parameters for each T1M Line (channel) operating as an ARD circuit. The following parameters are the exceptions to the module's default configuration:

- 1 Enter a **Line Number**.
- 2 In **Bearer Capability**
 - Set **Octet 3 Transfer** to **Speech**
 - Enable **Octet 5** and set to **μ-law**.
- 3 In **Call Establishment Modes**:
 - Enter a **Hotline Number** (the same as the Line Number of the paired ALEM port).
 - Enable **Hotline**
 - Disable **DDI (Indial)**
- 4 Set **Protocol** to **PLAR**.

E1M (ARD)

Note *ARD operation is an entire module setting, and not a Line-by-Line setting. Individual E1M channels cannot be configured for MRD operation when the module is configured for ARD operation. However, individual channels can be configured for TO operation when the module is configured for ARD operation.*

Configure the **E1M Card Control** parameters:

- 1 Enable **CAS enable**.
- 2 Set **CRC enable** as appropriate for the connected equipment.
- 3 Set Protocol to **ARD**.

Configure the **Line Configuration** parameters, for each E1M Line (channel) operating as an ARD circuit. The following parameters are the exceptions to the module's default configuration:

- 1 Enter a **Line Number**.
- 2 In **Bearer Capability**:
 - Set **Octet 3 Transfer** to **Speech**
 - Enable **Octet 5** and set to **A-law**
- 3 In **Call Establishment Modes**:
 - Enter a **Hotline Number** (make the same as the Line Number of the paired ALEM port).
 - Enable **Hotline**
 - Disable **DDI (Indial)**.

ALEM (ARD)

Configure the **Line Configuration** parameters, for each ALEM Line (port) operating as an ARD circuit. The following parameters are the exceptions to the module's default configuration.

Note *Prior to configuration, set the required physical port impedance for each port by setting the appropriate links on the ALEM circuit board. Refer to the **Installation Manual** for instructions.*

1 Enter a **Line Number**.

2 In **Bearer Capability**:

- Set **Octet 3 Transfer to Speech**
- Enable **Octet 5** and set to **μ-law** if paired with a T1M or to **A-law** if paired with an E1M.

3 In **Low Layer Compatibility**:

- Set **Octet 3 Transfer to Speech**
- Enable **Octet 5** and set to **μ-law** if paired with a T1M or to **A-law** if paired with an E1M

4 In **Call Establishment Modes**:

- Enter a **Hotline Number** (the same as the Line Number of the paired T1M/E1M channel).
- Enable **Hotline**.

5 In **Codec, Relative Levels**:

- Set **Digital** → **Analog** and **Digital** ← **Analog**, to values appropriate for the installation.

6 In **Signals**:

- Set **Signalling Options** to **Ring Out / Loop In**.

Manual Ring Down (MRD) Parameters

The following are the minimum requirements necessary to ensure functional operation of a stand-alone device for MRD Channel Bank operation.

T1M (MRD)

Configure the T1M **Card Control** parameters:

- 1 Set **Framing** and **Zero Suppression** as appropriate for the connected equipment.

Configure the **Line Configuration** parameters, for each T1M Line (channel) operating as an MRD circuit. The following parameters are the exceptions to the module's default configuration:

- 1 Enter a **Line Number**.
- 2 In **Bearer Capability** confirm these settings:
 - Set **Octet 3 Transfer** to **Unrest. Digital**
 - Disable **Octet 5**.
- 3 In **Call Establishment Modes**:
 - Enter a **Hotline Number** (the same as the Line Number of the paired EMM port).
 - Enable **Hotline**
 - Enable **XSPC**
 - Disable **DDI (Indial)**
 - Enable **End-to-End**.
- 4 Set **Protocol** to **E&M Immediate Start**.

E1M (MRD)

Note *MRD operation is an entire module setting for the E1M and not Line-by-Line. Individual E1M channels cannot be configured for ARD operation when the module is configured for MRD operation. However, individual channels can be configured for TO operation when the module is configured for MRD operation.*

Configure the **E1M Card Control** parameters.

- 1 Enable **CAS enable**.
- 2 Enable **OPS mode**.
- 3 Set the **CRC enable** as appropriate for the connected equipment.
- 4 Set Protocol to **MRD**.

Configure the **Line Configuration** parameters, for each E1M Line (channel) operating as an MRD circuit. The following parameters are the exceptions to the module's default configuration.

- 1 Enter a **Line Number**.
- 2 In **Bearer Capability**:
 - Set **Octet 3 Transfer** to **Unrest. Digital**.
 - Disable **Octet 5**.
- 3 In **Call Establishment Modes**:
 - Enter a **Hotline Number** (the same as the Line Number of the paired EMM port).
 - Enable **Hotline**
 - Enable **XSPC**
 - Disable **DDI (Indial)**
 - Enable **End-to-End**.

EMM (MRD)

Configure the **Line Configuration** parameters, for each EMM Line (port) operating as an MRD circuit. The following parameters are the exceptions to the module's default configuration.

1 Enter a **Line Number**.

2 In **Bearer Capability**:

- Set **Octet 3 Transfer to Unrest. Digital**
- Disable **Octet 5**.

3 In **Low Layer Compatibility**:

- Set **Octet 3 Transfer to Speech**
- Enable **Octet 5** and set to **μ-law** if paired with a T1M or to **A-law** if paired with an E1M.

4 In **Call Establishment Modes**:

- Enter a **Hotline Number** (the same as the Line Number of the paired T1M/E1M channel).
- Enable **Hotline**
- Disable **DDI (Indial)**
- Disable **Back Busy**
- Enable **End-to-End**
- Enabled **Unack**
- Disable **Auto Answer**.

5 In **Codec, Relative Levels**:

- Set **Digital** → **Analog** and **Digital** ← **Analog** to values appropriate for the installation.

Tone Only (TO) Parameters

The following are the minimum requirements necessary to ensure functional operation of a stand-alone device for Channel Bank operation.

T1M (TO)

Configure the **T1M Card Control** parameters:

- 1 Set **Framing** and **Zero Suppression** as appropriate for the connected equipment.

Configure the **Line Configuration** parameters, for each T1M Line (channel) operating as a TO circuit. The following parameters are the exceptions to the module's default configuration.

- 1 Set **Protocol** to **None**.
- 2 In **Call Establishment Modes**:
 - Set **Subrate virtual member or manually connected** to **Yes**.

E1M (TO)

Configure the **E1M Card Control** parameters for either ARD or MRD module operation.

Configure the **Line Configuration** parameters, for each E1M Line (channel) operating as a TO circuit. The following parameters are the exceptions to module's default configuration.

- 1 In **Call Establishment Modes**:
 - Set **Subrate virtual member or manually connected** to **Yes**.

EMM (TO)

Configure the **Line Configuration** parameters, for each EMM Line (port) operating as an TO circuit. The following parameters are the exceptions to module's default configuration.

- 1 Enable **End-to-End**.
- 2 Set **Subrate virtual Line member or manually connected** to **Yes**.
- 3 In **Bearer Capability**:
 - Set **Octet 3 Transfer** to **Speech**
 - Enable **Octet 5** and set to **μ-law** if paired with a T1M or to **A-law** if paired with an E1M.

4 In Low Layer Compatibility:

- Set **Octet 3 Transfer to Speech**
- Enable **Octet 5** and set to **μ-law** if paired with a T1M or to **A-law** if paired with an E1M

5 In Codec, Relative Levels:

- Set **Digital** → **Analog** and **Digital** ← **Analog** to values appropriate for the installation.

Establishing ARD Connections

The connections between ARD Line pairs are temporary. They are established in response to SEIZE from the analog equipment attached to the ALEM or from the CB equipment attached to the T1M/E1M. When both the analog equipment and the CB equipment return to idle, the connection between the ARD Line pairs is cleared down.

This type of connection is known as an 'on-demand' call. Our equipment provides supervisory tones, such as Ring Tone to Lines (channel/ports) establishing connections as on-demand calls.

Establishing MRD Connections

The connections between MRD Line pairs are established as permanent connections. Once established, signalling transfers between the EMM port and the T1M/E1M channel and a two-way speech path is open.

To establish the connection, enable the **XSPC** checkbox control for the T1M or E1M Line as shown in the configuration details above.

A connection between the Line pairs will automatically establish within 20 seconds. This can be monitored using OmniVision port status display on the EMM Port and T1M/E1M Channel. In both cases, the colour changes from blue to yellow on successful establishment of the connection.

The XSPC connection between the Line pairs remains established, irrespective of T1M or E1M link conditions. The connection will also automatically re-establish after CM initialisation or restoration of power following an outage.

Establishing TO Connections

The connections between TO Line pairs are established as permanent connections. On establishment, a two-way speech path is open.

The TO connection is a 'Manual Connection'. To establish a Manual Connection between TO Line pairs, follow the instructions provided below:

- 1 Establish an on-line session with the device.
- 2 Select **Connection Manager Control** from the **Other** menu.
- 3 Select **Bidirectional**.
- 4 Enable **Non-Volatile**.
- 5 Select **External**.
- 6 Scroll through the entries in the **This Connection** table and select an EMM port entry.
- 7 Select the **From** button (or double-click on the EMM port entry). The Line information, from the selected entry appears in Card, Sub and Line fields beside the From button.
- 8 Scroll through the entries in the **This Connection** table again and select a T1M or E1M channel entry.
- 9 Select the **To** button (or double-click on the T1M/E1M channel entry). The Line information from selected entry appears in the Card, Sub and Line fields beside the To button.
- 10 If you used the double-click method, go to step 12.
- 11 Select the **Connect** button.
- 12 A new entry appears in the **Active Connections** table.

Note *The **From** and **To** Line information fields become blank.*

- 13 Verify that the Line information parts of the new listing match the required EMM and T1M/E1M pairs.
- 14 Repeat steps 4 - 12 for all required TO connections.
- 15 Press the **OK** button when you have finished installing connections. This closes the dialog.
- 16 Verify that the appropriate EMM ports and the T1M/E1M channel indicators have changed colour from blue to green.

Appendices

Appendix A

Data rates for Subrate multiplexing

User data rate (kbit/s)	Sync	Async	Subchannel rate with V.110	Subchannel rate without V.110
64	Yes	No	—	64 kbit/s
56	Yes	No	64 kbit/s	56 kbit/s
48	Yes	No	—	64 kbit/s
24	Yes	No	—	24 kbit/s
19.2	Yes	No	32 kbit/s	—
16	Yes	No	—	16 kbit/s
9.6	Yes	Yes	16 kbit/s	—
8	Yes	No	—	8 kbit/s
4.8	Yes	Yes	8 kbit/s	—
2.4	Yes	Yes	8 kbit/s	—
1.2	Yes	Yes	8 kbit/s	—
0.6	Yes	Yes	8 kbit/s	—
0.3	No	Yes	8 kbit/s	—

Appendix B

ITU-T (CCITT) recommendations

- E.164** The numbering plan for the ISDN era
- G.703** Physical and electrical characteristics of hierarchical digital interfaces
- G.704** Functional characteristics of interfaces associated with network nodes
- I.030** ISDN et al
- Q.422** R2 signalling system
- Q.921** ISDN L2 signalling
- Q.931** ISDN L3 signalling
- Q.933** Digital Subscriber Signalling System No.1 (DSS1) – Signalling Specification for Frame Mode Basic Call Control
- V.24** List of definitions for interchange circuits between DTE and DCE
- V.35** Data transmission at 48 kbit/s using 60-108 kHz group band circuits
- V.110** Support of DTEs with V-series type interfaces by an ISDN
- X.21** Interface between DTE and DCE for synchronous operation on public data networks
- X.25** Interface between DTE and DCE for terminal operating in the packet mode and connected to public data networks by dedicated circuit
- X.200** OSI 7 layers

American National Standards

- ANSI T1.617a** Integrated Services Digital Network (ISDN) – Signaling Specification for Frame Relay Bearer Service for Digital Subscriber Signaling System Number 1 (DSS1) (Protocol encapsulation and PICS).

Appendix C

AT commands

An ADLM V.24 and a QDLM V.24 line can simulate an asynchronous V.24 modem by operating in asynchronous mode. This is achieved by enabling the AT commands option in the call establishment modes dialog box in OmniVision.

When simulating the operation of a modem, the DLMs have two modes of operation, *command* mode and *online* mode.

Command mode connects the Data Terminal Equipment (DTE) to the command processor which interprets and executes configuration and call control commands.

Online mode connects the DTE to a remote DTE via the ISDN B-channel.

Conventions

- The ATTENTION code (AT), which signals that one or more commands follow, must begin each command line
- Letters can be entered in upper, lower or mixed case
- Numbers are decimal values unless otherwise specified
- Spaces may be used to separate commands.

Format

Commands must be in the following format:

```
AT [command(s)]
```

For example, to answer an incoming call using the ANSWER Command (A):

Type ATA and press <enter>

Note *If an invalid command, or a valid command with an invalid parameter is entered, an ERROR response is issued and command processing is aborted. However, all valid commands preceding the error are completed.*

Commands

The most frequently used commands are shown in the following table and described in detail on the following pages. Use these commands following the ATTENTION (AT) code.

Command	Parameter	Default
A (Answer)	N/A	N/A
D (Dial)	20 digits	Hotline No.
E (Echo)	0-1	1
H (Hang Up)	N/A	N/A
I (Inquiry)	N/A	N/A
O (Online)	N/A	N/A
Q (Quiet)	0-1	0
S (Set Register)	0-255	Register 2: 43 Register 12: 50 All others: 0
Z (Reset)	N/A	N/A
&D (DTR Control)	0-2	0
&H (Help)	N/A	N/A

A (ANSWER)

Causes the DLM V.24 to immediately answer an incoming call and enter online mode. It will not wait for the ring signal before answering.

To use the ANSWER command:

Type A in the command string.

Note *If the ANSWER command is used, it must be the last character in the command line (all of the characters following it are ignored).*

A/ (REPEAT PREVIOUS)

Repeats the previous command. QDLM only.

To use the REPEAT PREVIOUS command:

Type A/ without the AT attention code preceding it.

&C (CARRIER DETECT CONTROL)

Determines how the QDLM controls the CD output to the attached DTE.

To use the CARRIER DETECT CONTROL command:

Type &Cn or &C? in the command string.

When n = 0, the CD signal is permanently asserted.

When n = 1, the CD follows connection status, that is V.110 frame status (default).

&C? reads the value stored in the &C register.

D (DIAL)

Causes the DLM V.24 to originate a call.

To use the DIAL command:

Type `Dn` or `Dn;` in the command string where

`n` is a telephone number (may be 0-9, *, # and A, B, C and D) and the semicolon (;) instructs the ADLM V.24 to remain in command mode after dialling.

Note *If ; is used, it must be the last character in the command line (all the characters following it will be ignored).*

The DIAL command is aborted if the space character is received from the DTE.

&D (DTR CONTROL)

Determines how the DLM V.24 responds to the presence or absence of the DTR signal from the DTE.

To use the DTR OPTIONS command:

Type `&Dn` or `&D?` in the command string.

When `n = 0`, the DTR signal is ignored (default).

When `n = 1`, if DTR goes off, the DLM V.24 terminates the call and returns to the command state.

When `n = 2`, if DTR goes off, the QDLM terminates the call and returns to the command state (QDLM only).

`&D?` reads the value stored in the &D register (QDLM only).

E (ECHO)

Controls the echo to the DTE of characters received by the DLM V.24 from the DTE when in command mode.

To use the ECHO command:

Type `En` in the command string.

When `n = 0`, local echo is disabled.

When `n = 1`, local echo is enabled - default.

&F (LOAD DEFAULT)

Causes the QDLM to assume the default parameter values.

The LOAD DEFAULT command is equivalent to the command string `ATE1V1QO&DO`.

To use the LOAD DEFAULT command:

Type `&F` in the command string.

H (HANG UP)

Causes the immediate termination of a call.

To use the HANGUP command:

Type `H` or `H0` in the command string.

&H (HELP)

Displays the list of commands that are supported by the DLM V.24.

To use the HELP command:

Type `&H`.

I (INQUIRY)

Use the INQUIRY command to request the DLM V.24 product code and revision level.

To use the INQUIRY command:

Type `I` in the command string.

In verbose mode the ADLM V.24 will respond with:

```
ADLM V.24 Vxx.yy
```

In verbose mode the QDLM V.24 will respond with:

```
QDLM V.24 Vxx.yy
```

In terse mode the DLM V.24 will respond with:

```
Vxx.yy
```

where `xx.yy` is the DLM V.24 firmware version number.

O (ONLINE)

Causes the DCE to enter online mode from command mode.

To use the ONLINE command:

Type `O` in the command string.

Note *To re-enter command mode from online mode, use the ESCAPE code.*

Q (QUIET)

Determines whether result codes are sent to the DTE by the DLM V.24 when commands are executed.

To use the QUIET command:

Type `Qn` in the command string.

When $n = 0$, messages and result codes are sent (default).

When $n = 1$, messages and result codes are not sent.

When $n = 2$, connect response is suppressed on incoming calls (QDLM only).

S (SET REGISTER)

Used for configuration register writing and reading.

To use the SET REGISTER command:

Type `Sr=n` or `Sr?`

Where n = the value to write to register r .

`Sr?` is used to read the value of register r .

(r must be in the range 0 to 38 and is represented by one or two decimal digits.

n must be in the range 0 to 255 and is represented by one, two or three decimal digits.)

If either value is out of range, the ERROR response is issued by the DCE.

Note *The DLM V.24 does not interpret the contents of the registers. This command is only included to allow operation with commonly used IBM™ PC communications packages.*

The result of the register read command is always a three digit decimal number followed by an end of line sequence (enter in terse mode and enter, line feed in verbose mode).

S0 (AUTOANSWER CONTROL)

Disables the auto answer facility. QDLM only.

To use the AUTOANSWER CONTROL command:

Type S0 in the command string.

When n = 0, autoanswer is disabled.

When n = anything else, autoanswer is enabled (default).

Note *Auto Answer only works when the Auto Answer option is enabled in the Manager.*

S2 (ESCAPE CHARACTER)

Disables the escape character facility. QDLM only.

To use the ESCAPE CHARACTER command:

Type S2 in the command string.

When n = 0, the escape sequence detection is disabled.

When n = anything else, the +++ escape sequence detection is enabled (default).

S12 (ESCAPE CODE GUARD TIME)

Sets the time delay required immediately before and after entering the escape code. The default delay time is 50 (1 second). QDLM only.

To use the ESCAPE CODE GUARD TIME command:

Type S12 in the command string.

V (VERBOSE)

Determines whether the DLM V.24 sends the result codes to the DTE as words (verbose mode) or as digits (terse mode). Unless the results are being returned to a program that requires digital responses, words are recommended.

To use the VERBOSE command:

Type Vn in the command string.

When n = 0, the result codes are sent as digits.

When n = 1, the result codes are sent as words (default).

X (EXTENDED RESULT CODES)

Sends extended CONNECT messages. QDLM only.

To use the EXTENDED RESULT CODES command:

Type `xn` or `x?` in the command string.

When `n = 0`, sends CONNECT message when a connection is established (default).
When `n = any other value`, sends CONNECT XXXX message reflecting bit rate when a connection is established.

Z (RESET)

Causes the DLM V.24 to assume the default parameter values.

The RESET command is equivalent to the command string `AT E1 Q0 V1 &C0 &D0`.

To invoke the RESET command:

Type `z` in the command string.

Escape code

Used to return to online command mode from data mode.

To enter the ESCAPE code:

- 1 Wait one second.
- 2 Type `+++` within one second.
- 3 Wait one second.

Result codes

After processing a command the ADLM V.24 responds with one or more of the result codes shown in the following table.

Digit Code	Result Code	Meaning
0	OK	The command(s) has been successfully executed (except for the dial command).
1	CONNECT	A dial command has been completed successfully and the ADLM V.24 is connected to the ISDN, unless the semi-colon modifier has been used with the dial command.
2	RING	An incoming call is detected. If the DCE is configured for auto-answer the call will be answered automatically and the ADLM V.24 will be connected to the ISDN via the data rate adaptor.
3	NO CARRIER	The ISDN connection has been broken by the other party clearing the call or there is an unacceptable level of transmission errors (an unacceptable level is defined by V.110).
4	ERROR	An invalid command is entered by the ADLM V.24.
6	NO DIAL TONE	The DCE is unable to complete a dial command due to an error between it and the local exchange.
7	BUSY	The DCE is unable to complete a dial command because the called party is busy.
8	NO ANSWER	The DCE is unable to complete a dial command because the call is not answered.
15	(Aborted!)	A dial command has been interrupted by the ADLM V.24.

In verbose mode, the full result code will be generated.

In terse mode, only the digit code will be generated.

See VERBOSE command for further information.

Appendix D

Operating analog line interfaces

Seven analog line interface modules are available:

- ALEM
- ALEM-2
- ALPM
- ALPM-2
- COMBO
- EMM
- EMM-2.

Note *The ALEM and ALPM are only available in Australia.*

In the following paragraphs ALEM(-2), ALPM(-2) and EMM (-2) signify that either type of card can be used.

All seven modules support semipermanent connections as well as manual connections and standard switch call connections. The COMBO module supports a variety of analog interfaces. The options available for configurations are **Auto Answer** and **End-to-end Signalling** on all six modules plus **Indial** on the ALPM(-2) and the EMM(-2). Care must be taken when choosing these options as they greatly alter the operating mode of the module.

	Auto Answer	Indial	End-to-End
ALEM	Yes	No	Yes
ALEM-2	Yes	No	Yes
ALPM	Yes	Yes	Yes
ALPM-2	Yes	Yes	Yes
COMBO ALIM-EXCH	Yes	No	Yes
EMM	Yes	Yes	Yes
EMM-2	Yes	Yes	Yes

Auto Answer

When **Auto Answer** is set on the ALPM(-2), as soon as the **Setup** comes in from the network, a **Connect** is sent back and the line looped. Thus in a switched call mode, call charging commences immediately.

Dialling requests received by analog modules from the CM/RM can be dialled out to user equipment either from the **Dialling** state or from the **Active** state.

A call is in the **Dialling** state when it has received an incoming call from the ISDN and has responded with **Call Proceeding**. A call is in the **Active** state when a **Connect** has been sent back to the Network.

Like the ALPM(-2), the EMM(-2) responds to a **Setup** from the network immediately with a **Connect** message and sends a **Seize** onto the EMM(-2) E lead. Thus when the call is finally answered by the PABX no additional message is sent out.

The EMM(-2) can also dial from either the **Dialling** or the **Active** states but only Decadic dialling is possible. When **Auto Answer** is enabled, an ALEM(-2) responds to a **Setup** with a **Connect**, however, no ring signal is sent out.

India

India is available on the ALPM, ALPM-2 and EMM(-2). This option allows you to select how many digits of the **Called** party number will be dialled into the terminal equipment. The first few digits can be ignored if **Number Of Digits To Ignore** is configured. For example, if the called number is a seven digit number and the number of digits to ignore is set to three, the last four digits will be dialled.

End-to-End Signalling

End-to-end Signalling Enabled is available for all six modules. It is extremely important that this option is set correctly as it greatly affects the actions of the modules. It is an option which affects both manually connected calls and semipermanent connections.

In the case of semipermanent connections with **End-to-end Signalling** switched off, the module will change from the **Idle** state to the **Active** state as soon as the **Setup** comes in for the semipermanent connection.

For example, the **Setup** comes in on an ALPM or ALPM-2 and the line is looped; the ALPM(-2) goes into the **Active** state and no signalling takes place. That is, battery reversals do not make the call disconnect and the only way out of the **Active** state is for a **Disconnect** message to come from Call Control. This involves either manual intervention by an operator in the exchange or an **Interface Restart**. Deactivating the semipermanents will not cause this to happen (if you switch **End-to-end Signalling 'On'** later, no change occurs).

For normal switched operation **End-to-end Signalling** must be switched off. If **End-to-end Signalling** is switched on and no semipermanent or manual connect call has been set up, no calls can originate from that particular line. For instance, looping the line on an ALEM(-2), or ringing in to an ALPM(-2), will not send a **Setup** out to the network. Thus no calls can be set up from the interface side.

For ALEM(-2)s, ALPM(-2)s, COMBOs and EMM(-2)s to talk as a switched call over a semipermanent, **End-to-end Signalling** must be switched on. Firstly, this enables the CM/RM to set up a D-channel or Inband B-channel path for signalling to be passed in **User Information** messages. Additionally, it tells the line entity to send its signalling information in User Information packets as described by the proprietary IDAS protocol.

The IDAS (Interpreted D-channel Associated Signalling) protocol uses messages which represent call control and dialling information, as interpreted by the modules from the attached equipment. About six **User Information** messages are required per completed call (before dialling) with an additional data packet per number digit.

Using IDAS, an ALEM(-2), ALPM(-2), COMBO or EMM(-2) can be at either end of the semipermanent connection or manual connection and sensible signalling will happen at each end. For instance, if an ALEM(-2) is looped and an ALPM(-2) is at the far side, the ALPM(-2) will then loop.

Dialling is also possible. However, if DTMF dialling is used then **Indial** must be switched off on the ALPM(-2) since a B-channel is available. The tone will go along the B-channel to the customer equipment at the far end (e.g. PABX). Additionally, the tone will be interpreted by the ALEM(-2) and sent through as a user data request. The ALPM(-2) will then dial that out if the **Indial** is switched on, thus creating two DTMF tones.

If Decadic dialling is required the ALEM(-2) can be dialled into. It will send a user packet through to the other side and the ALPM(-2) side can either dial DTMF or Decadic depending on what has been selected using NMS. For this to occur, **Indial** must be switched on.

All of this holds true for the EMM(-2) except that DTMF dialling is not available on earlier versions of the EMM(-2). Please check with your supplier to confirm whether your EMM(-2) supports DTMF.

Appendix E

UDAS and IDAS signalling with analog modules

SPCs are the ISDN replacement of tie lines or leased lines, and are significantly cheaper in many cases.

An application requiring an E&M Module (EMM) uses E&M signalling for inter-PABX transfer of calls.

Most PABXs have E&M cards which use one pair of wires for voice transmission, one pair for voice reception, an input signalling wire (E) and an output signalling wire (M).

For these PABXs to signal over leased lines, it is necessary to convert the signalling leads into analog waveforms and transmit them on a dedicated signalling channel.

To duplicate this with an EMM sending signalling information across an SPC, D-channel signalling was adopted because it provides a number of advantages:

- SPCs increase in cost with distance whereas the cost of a D-channel packet is constant.
- The number of D-channel packets (and hence the cost) is directly proportional to the number of telephone calls made on that line.
- A number of signalling calls can be made without taking up B-channel bandwidth.

The two types of proprietary signalling are Uninterpreted D-channel Associated Signalling (UDAS) and Interpreted D-channel Associated Signalling (IDAS). Despite their names they can also be used with Inband B-channel signalling. They are each described in more detail below.

Uninterpreted end-to-end signalling

UDAS is a protocol that is transparent to PABXs. That is, whatever signalling scheme the PABX uses (assuming a single E and a single M signalling lead), the lead status, including signal pulse length and state is transmitted.

Acknowledgment packet data can be selected to ensure that all data is received and that signal leads reliably indicate the status of those at the other end. This uses twelve packets per call overall.

Although not guaranteed, testing has indicated that reliability is not compromised by the omission of acknowledgment packets, so the unacknowledged UDAS signalling mode option is provided. This uses four packets per call.

Interpreted end-to-end signalling

IDAS provides signalling between different modules as well as inter-module functionality for modules other than the EMM. For example, it provides the ability to connect an ALEM, ALEM-2 or COMBO across an SPC to another ALEM, ALEM-2, ALPM, ALPM-2, COMBO, EMM, EMM-2 or an E1M.

This interpretive signalling system can decide what information to send and what action to take on receipt of signals (for example, ring the handset, dial a digit or disconnect the call). This requires six packets per call plus one packet per dialled digit.

Appendix F

Note *Appendix F applies to Australia only.*

ALPM Indial operation

The ALPM is provided with Indial capabilities which are invoked using the Call Establishment Modes dialog for the relevant line of the module.

This facility enables you to select how many digits of the Called Party Number will be dialled in to the terminal equipment (typically a PABX with Indial capabilities).

The operator can select the number of Called Party Number digits to ignore. Normally, the ISDN exchange presents seven Called Party Number digits. If the last four digits are required by the PABX, then the first three should be ignored.

Both Pulse (decadic) and DTMF (tone) dialling from the ALPM is available.

If required, multiple ALPM lines can be associated using the Line Set Associations dialog.

Configuration

The ALPM is initially configured for normal operating mode.

To invoke the correct dialling type, select either DTMF or Pulse in the Dial Type dialogue.

Select the Call Establishment Modes dialog and set the number of digits to be ignored using the numeric keypad in the Ignore First field.

To invoke the Indial feature of the ALPM, the Indial Enable option must be selected. When using the ALPM in semipermanent connection (SPC) mode, Auto Answer must be enabled.

If the ALPM is used in normal switched mode, Auto Answer should be disabled. When Auto Answer is enabled and the ALPM is in Indial mode, all calls will be charged by the carrier regardless of the condition (Idle or Busy) of the terminal equipment because the call will be accepted irrespective of whether it is accepted by the terminal equipment.

Setting the Auto Answer option to disabled when the ALPM is in Indial mode prevents unwarranted charging of the call. Busy and ring signals from the terminating equipment are detected, and an indication of idle and busy states are provided to the network.

Configuration of the ALPM to detect Ring and Busy signals from the terminating equipment is performed using the Service Tones dialog. The signal on/off durations must be configured correctly so that appropriate indications are made to the network.

Detection of ring and busy signals are made with a +/- 30% tolerance of that specified in the Service Tones dialog.

The standard Telstra signal specifications used in Australia appear in the table below.

	Make	Break	Make	Break
Ring	400 ms	200 ms	400 ms	2000 ms
Busy	375 ms	375 ms	375 ms	375 ms

Note *We round up the Busy Signal duration to 400 ms.*

Some PABXs may require the ALPM to be tuned using the Service Tones dialog. The operation of this should be verified during the system's commissioning.

Appendix G

X.21 interfaces

ITU-T (CCITT) recommendation X.21 defines a synchronous DCE-DTE interface with procedures for circuit switched and leased line operation. The Synchronous Digital Line Module X.21 (SDLM X.21) provides the DCE side of this interface and supports both circuit types via the ISDN.

This appendix describes:

- The X.21 physical interface
- The X.21 call establishment and clearing procedures
- How the SDLM X.21 translates these procedures to the ISDN D-channel protocol
- How the user data is transferred on the B-channel
- How to configure the SDLM X.21 for a number of typical situations.

X.21 physical interface

The X.21 recommendation relies on other recommendations and standards to define the physical nature of the X.21 interface. These are ISO4903, and ITU-T (CCITT) recommendations V.11 and X.24.

The ISO4903 standard defines the 15-pin D-type connector which is fitted to the front panel of the SDLM X.21. The SDLM X.21, which is a DCE, has female contacts, and the DTE has male contacts.

The V.11 recommendation defines the electrical characteristics of the signals which appear in the 15-way connector. It specifies the conditions for the **ON**, **OFF**, **0**, and **1** states of the X.21 signals in terms of the differential voltage between the two signal-carrying conductors (V_A and V_B). If V_A is low and V_B is high, then the signal is **1** (for data circuits) or **OFF** (for control circuits). If V_A is high and V_B is low, then the signal is **0** (for data circuits) or **ON** (for control circuits).

The differential output voltage of most X.21 devices (including the SDLM X.21) is 5V. This enables them to overcome a certain amount of interference and line loss, and yet still be correctly received since the receiver needs only a 0.3V input voltage difference.

The V.11 recommendation provides for data rates up to several million bits per second, however, the SDLM X.21 supports user data rates of 600, 1200, 2400, 4800, 8000, 9600, 16000, 19200, 24000, 32000, 48000, 56000, and 64000 bits per second. At these rates, circuits which conform to the V.11 recommendation are capable of operating on twisted pair cables up to 1200m in length, provided that the clock and data are both travelling in the same direction. This is not true for most X.21 interfaces, therefore the cables are limited to 100 metres for the SDLM X.21.

V.11 is very similar to the RS-422 standard, thus making the X.21 electrically compatible with the RS-449 interface. It is possible to use the SDLM X.21 for the connection of RS-449 equipment at the physical level by using an adaptor cable which connects the **S**, **R**, and **T** signals of the X.21 end to the clock, receive data, and transmit data pins at the RS-449 end.

The X.24 recommendation specifies the names and meanings of the various X.21 signals. These are: **S** (signal timing element or clock), **T** (transmit), **R** (receive), **C** (control), **I** (indication) and **B** (byte timing).

The control signals **S**, **C**, **I** and **B** can have the values **ON** or **OFF**. The data signals **R** and **T** can have the values **0** or **1**.

Transitions of the **T**, **R**, **C** and **I** signals occur at the **ON** to **OFF** transitions of the **S** signal and are sampled at the **OFF** to **ON** transitions of the **S** signal.

On the SDLM X.21, the **B** signal is provided only when the user data rate is 64 kbit/s. In addition, the **B** signal on the SDLM X.21 is **OFF** for the whole period of the eighth bit, rather than only half of the bit period, as specified by X.24.

X.21 circuit switched operation

ITU-T (CCITT) recommendation X.21 defines a set of call establishment and clearing procedures which enable a DTE to make outgoing calls and accept incoming calls in much the same way as a telephone handset.

The DTE controls the SDLM X.21 using various combinations of signals on the **T** and **C** leads. Similarly, the SDLM X.21 indicates its status to the DTE using the **R** and **I** leads.

The **C** (control) and **I** (indication) signals can be in one of two states, either **ON** or **OFF**.

The **T** (transmit) and **R** (receive) signals can have several states, but the most important states are continuous 0s, continuous 1s and a state in which IA5 (international alphabet number 5) characters are being transferred.

The 0 and 1 states must persist for at least 16 bit periods in order for the SDLM X.21 to recognise them.

The IA5 character set is similar to the ASCII character set. When IA5 characters are being transferred, they are always 7 bits with even parity, and are always preceded by at least a pair of SYN (synchronisation) characters.

Ready state

When the DTE and the DCE are in the **READY** state, the **C** and **I** signals will be **OFF**, and the **T** and **R** signals will be continuous 1s (that is, C=OFF, T=1, I=OFF, R=1).

Outgoing calls

When the DTE wants to make an outgoing call it tells the SDLM X.21 by setting **C=ON** and **T=0**, thereby entering the **Call Request** state. The SDLM X.21 then sends a **Setup** packet to the ISDN exchange which usually responds with a **Setup Acknowledge** packet.

The SDLM X.21 then enters the **Proceed To Select** state by sending, on the **R** signal, a pair of **SYN** characters followed by continuous + (plus) characters. When the DTE sees the + characters it sends a telephone number on the **T** signal, preceded by a pair of **SYN** characters and followed by a single + character.

If the DTE does not send a telephone number, the call will be cleared by the SDLM X.21. The telephone number is forwarded to the ISDN exchange by the SDLM X.21 using several **Info** packets. At this point the DTE will be in the **DTE Waiting** state (**C=ON, T=1**) and the SDLM X.21 will be in the **Connection In Progress** state (**I=OFF, R=1**).

When the exchange sends a **Connect** packet, the SDLM X.21 enters the **Ready For Data** state by setting **I=ON**. At this point the calling DTE is connected to the called DTE and they can each send data using the **T** and **R** signals. The SDLM X.21 does not interpret any of the user data, and the DTEs are free to use any protocol.

If the ISDN fails to complete a call connection, the SDLM X.21 will initiate the X.21 call clearing procedure after first sending a clearing code to the DTE. The clearing codes are translated from the ISDN cause codes in the **Disconnect** or **Release** packet.

Incoming calls

The ISDN exchange notifies the SDLM X.21 of an incoming call by sending a **Setup** packet. It responds with a **Setup Acknowledge** packet. The DTE is notified by the SDLM X.21 entering the **Incoming Call** state, during which it sends continuous **BEL** characters (preceded by a pair of **SYN** characters).

The DTE answers the call by entering the **Call Accepted** state (**C=ON, T=1**). When the SDLM X.21 sees this state, it stops sending **BEL** characters and enters the **Ready For Data** state (**I=ON, R=1**) and the DTEs are connected.

If the DTE fails to answer the call within 0.5 seconds, the SDLM X.21 will clear the call.

Call clearing

When the DTE wishes to clear a call, it notifies the SDLM X.21 by entering the **Dte Clear Request** state (**C=OFF, T=0**). The SDLM X.21 sends a **Disconnect** or **Release** packet to the ISDN exchange which replies with a **Release Complete** packet. The SDLM X.21 then enters the **Dce Clear Confirm** state (**I=OFF, R=0**) and then both the DTE and the SDLM X.21 go to the **Ready** state (**C=OFF, T=1, I=OFF, R=1**).

If the SDLM X.21 receives a **Disconnect** or **Release** packet from the ISDN exchange it enters the **DCE Clear Indication** state (**I=OFF, R=0**). The DTE replies with **DTE Clear Confirmation** (**C=OFF, T=0**) and then both go to the **Ready** state (**C=OFF, T=1, I=OFF, R=1**).

When the SDLM X.21 clears a call, either in response to a DTE clear request or due to abnormal call failure, it first sends an X.21 call progress signal to the DTE. The call progress signal consists of a pair of IA5 numeric characters preceded by a pair of **SYN** characters and followed by a + character.

The set of call progress signals issued by the SDLM X.21 is:

20 = No Connection
21 = Number Busy
22 = Selection Signals Procedure Error
41 = Access Barred
42 = Changed Number
43 = Not Obtainable
44 = Out Of Order
48 = Invalid Facility Request
49 = Network Fault In Local Loop
52 = Incompatible Class Of User Service
61 = Network Congestion

X.21 leased line operation

When the DTE is not using the X.21 call establishment and clearing procedures, it may use leased line operation. In this mode, the **C** signal indicates that the DTE is ready for data from the SDLM X.21; the **I** signal indicates that the SDLM X.21 is connected to the remote SDLM X.21 and is ready for data from the DTE.

When a DTE is operated in leased line mode and D-channel calls are available, the SDLM X.21 should be configured for semipermanent circuit operation.

When a DTE is operated in leased line mode and no D-channel call facility exists, the Subrate Virtual Line or manual connection options should be enabled.

The SDLM X.21 provides two ways of simulating an X.21 leased line: **C** signal call control, and manual call connection. In either of these two cases, the SDLM X.21 must be configured for Hotline operation.

With **C** signal call control, the DTE can control the connection with the **C** signal. At the calling end, a transition of **C** from **OFF** to **ON** will cause an outgoing call to be made. When the call has been connected, the SDLM X.21 sets **I=ON**. When **C** goes from **ON** to **OFF**, the call will be cleared and **I** will go **OFF**.

At the called end, a transition of **I** from **OFF** to **ON** indicates the presence of an incoming call. This should be answered by the DTE setting **C** from **OFF** to **ON**. Alternatively, the SDLM X.21 can be configured to automatically answer the call and ignore the **C** signal.

With manual call connection, both ends of the call should be configured for Hotline, auto-answer operation. A call can then be initiated from either end using the **Manual Connect** facility.

Note that in all cases (both X.21 circuit switched and leased line) the **Manual Terminate** facility can be used to clear a call.

B-channel data transfer

In order to transfer user data on the B-channel, the SDLM X.21 must perform data rate adaptation so that the user rate data can be carried on the 64 kbit/s B-channel. The technique that is used conforms to the ERMA 102 standard and also to the ITU-T (CCITT) X.30 and V.110 recommendations.

At data rates of 48 kbit/s and less, user data is encapsulated in a frame structure. The frames carry the user data, provide for end-to-end data synchronisation and for interchange signal transfer (although this is not used by the SDLM X.21).

When the user data rate is 56 or 64 kbit/s, there is no framing, however, at 56 kbit/s, the eighth bit of each B-channel octet is not used and is set to 1.

The data rate adaptation technique used by the SDLM X.21 is the same as that used by the SDLM V.24 and the SDLM V.35. This means that any combination of these line modules can be present at either end of a connection, provided that the user data rate and ISDN parameters are the same at both ends.

When a B-channel connection is made, the SDLM X.21 must establish and maintain frame synchronisation with the other end of the connection. Initially, it allows 10 seconds for frame synchronisation. If frame synchronisation cannot be achieved or is lost for 3 seconds or more, the call will be cleared.

If frame synchronisation is lost on a semipermanent circuit, a Hotline backup can be automatically set up by using the **Semipermanent Backup** option in the **Call Establishment Modes** configuration menu. When the backup facility is enabled, and frame synchronisation is lost for 3 seconds, the SDLM X.21 at one end will attempt to make one (and only one) Hotline call to the other end (or to an alternate site). When the backup call is no longer required, it is terminated using the **Manual Terminate** facility and the semipermanent circuit is activated in the usual manner.

SDLM X.21 Call Establishment Modes

This section describes the use of the options available in **Call Establishment Modes** for the SDLM X.21 when using the NMS configuration application.

Hotline

This must be switched on and a telephone number entered whenever X.21 Call Control or semipermanent circuit are not being used.

SPC (where available)

This is switched on when a semipermanent circuit is being used and a semipermanent label has been entered. When the SDLM X.21 is operating in this mode, it will set **I=ON** while the circuit is active and will ignore transitions of the **C** signal.

SPC backup (where available)

This is used to automatically backup a faulty semipermanent circuit operating at 48 kbit/s or less. The SDLM X.21 monitors the B-channel signal and initiates a Hotline call if frame synchronisation is lost for more than 3 seconds. For this to operate correctly, it should only be enabled at one end (to avoid a call collision) and **Hotline** must also be switched on. The other end should be set to **Auto Answer**.

Auto Answer

This is used at the called end of a connection when the DTE is not using X.21 circuit switched call establishment and clearing procedures, and cannot answer a call by setting its **C** signal to **ON**. In this case, an incoming call will automatically be answered and the **I** signal will be switched **ON**. In addition, the **C** signal from the DTE will be ignored and the DTE cannot clear the call by setting its **C** signal to **OFF**.

X.21 call control

This should only be switched on if the attached DTE is using X.21 circuit switched call establishment and clearing procedures. In this case, **Hotline**, **Auto Answer** and **C Signal Call Control** must be switched off.

If X.21 Call Control is switched off, an incoming call must be answered by the DTE setting its **C** signal from **OFF** to **ON** in response to the **I** signal going from **OFF** to **ON**, unless **Auto Answer** is switched on.

C signal call control

This is used at the calling end of a connection when the DTE is not using X.21 circuit switched call establishment and clearing procedures. An outgoing call will be made when the DTE sets its **C** signal to **ON**. The call will be cleared when the DTE sets the **C** signal to **OFF**.

This call establishment procedure cannot be implemented by the plugging and unplugging of the DB15 DTE connector into and out of the SDLM X.21 port. This is because contact bounce can occur under these circumstances. Contact bounce can force the SDLM into an indeterminate state, with the result that calls may possibly not be initiated or terminated.

The C signal should only be raised under the soft control of a DTE which is permanently attached to the SDLM X.21 port. If soft control is not available, a bounce-free switch can be inserted in-line between the DTE and the SDLM to provide the C Signal Control functions.

Example 1

DTE using X.21 Circuit Switched Procedure

Hotline	Off.
AutoAnswer	Off.
X.21 call control	On.
C call control	Off.
Semipermanent (where available)	Off.
Semipermanent Backup (where available)	Off.
Member of Substrate Virtual line or manually connected	Off.

Example 2

Simulation of X.21 Leased Line using a semipermanent circuit (where available)

Hotline	Off.
Semipermanent	On.
Semipermanent Backup	Off.
Auto answer	Off.
X.21 Call Control	Off.
C Call Control	Off.

Example 3**Simulation of X.21 Leased Line using a manual connect and Auto answer**

Hotline	On.
Auto Answer	On.
X.21 Call Control	Off.
C Call Control	Off.
Member Of Subrate Virtual Line or manually connected	Off.
Semipermanent (where available)	Off.
Semipermanent Backup (where available)	Off.

Example 4**Simulation of X.21 Leased Line using C signal calling and answering**

Hotline	On.
Auto Answer	Off.
X.21 Call Control	Off.
C Signal Call Control	On.
Member Of Subrate Virtual Line or manually connected	Off.
Semipermanent (where available)	Off.
Semipermanent Backup (where available)	Off.

Appendix H

Dual/multiple node access

This equipment includes support for Dual/Multiple Node Access. This enables connection to more than one PRA or BRA, with each network access link potentially leading to a separate ISDN exchange. This increases the call handling capacity of this equipment and the overall reliability of the customer system.

The maximum limitations for the IPMTs and BRMTs are described in *Appendix J*.

The Dual/Multiple Node Access feature includes a simple load sharing algorithm for outgoing calls. A setup will always go out via the TE module which has the most available B-channels. Least cost routing is not implemented.

TE selection for outgoing calls also takes into account module health and IPMT Layer 2 status, avoiding a TE module (BRMT or IPMT) which is out of service or an IPMT on which the link to the exchange has failed or is not established.

TE Base Number and Line numbers

This feature enables the user to specify the lowest number in the range for a given PRA or BRA (usually a seven or eight digit number ending in a 0) in the IPMT or BRMT configuration dialog. The line numbers can then be configured as offsets from this base number (usually just one or two digits).

For example, on a PRA with a number range from 3640000 to 3640099, the base number is set to 3640000 and the line with the number 55 is called by dialling 3640055. Alternatively, zero (0) can be entered as the base number (or the field left blank) and full phone numbers can be entered for all lines.

Note that the Test Number and the OmniVision Manager Number, configured in the Resource Manager dialog, are treated in the same way as any other line phone number. If the base number is used, they must be entered as offsets from the base number. If the base number is not used, the full number should be entered in each of these fields.

The significance of this feature can be seen in utilising Dual Node Access. In this example, the chassis is fitted with two ISDN IPMTs. Each of these modules has a unique number range and, hence, a unique base number. If the two number ranges are 3640000-3640099 and 3900000-3900099, the base numbers on the IPMTs could be set to 3640000 and 3900000, respectively. The line with the number 55 could then be dialled using either of the numbers 3640055 or 3900055. If the line were to make an outgoing call, the calling party number included in the outgoing setup would be 3640055 if the first PRA were used and 3900055 if the call went out on the second PRA.

This feature does not affect linesets. Full phone numbers must still be entered for a lineset, and if access through n nodes is required, then the set must be entered n times with the appropriate number ranges specified.

Appendix I

Database limitations

Resource Manager

The following table shows the database limitations for a Resource Manager Version 1.99 or under and Version 2.00 and above:

Data	RM Ver. 1.99 and below	RM Ver. 2.00 and above
Max lines	658	1000 ¹
Max linesets	64	100
Max members per lineset	1000	1000
Max lineset members	1000	1000
Max individual phone numbers	100	100
Max CLI/Number Checking numbers	833 or (416 ranges)	1666 or (833 ranges)
Max CLI/Number Checking lists	254	254
Max VX Calls	75	120
Max Virtual Lines	64	64
Max Alarm logs	50	100
Max Call logs	70	400

¹ Max lines is increased to 1500 for RM Ver. 4.00 and above.

Control Module

The following table shows the database limitations for a Control Module:

Data	CM
Max lines	587
Max linesets	64
Max members per lineset	220
Max lineset members	1000
Max individual phone numbers	100
Max CLI/Number Checking numbers	1,666 or (833 ranges)
Max CLI/Number Checking lists	254
Max VX Calls	75
Max Virtual Lines	64
Max Alarm logs	50
Max Call logs	155

Appendix J

System limits

When selecting the type and mix of modules that can be used, the following factors must be considered:

- the backplane bandwidth requirements of the module
- the power supply or heat dissipation requirements of the module
- the database requirements of the module, in particular, the number of lines
- the type of chassis.

Backplane bandwidth limits

The fifteen and six slot chassis have eight 2Mb TDM buses, while the four slot chassis has four TDM buses. Each TDM bus comprises 32 x 64k channels. The allocation of these channels depends on which modules are fitted to the chassis.

The following table shows the number of backplane TDM channels required by each module:

Module type	TDM channels
CM	0, 1 or 2
RM ¹	2-18
IPMT, IPMN, E1M, T1M	32
DTM	0-128
BRMN, BRMT	2
ALEM, ALEM-2, ALPM, ALPM-2, EMM, QDLM	4
SDLM V.24, SDLM V.35, SDLM X.21	2
HSDM	32
SRMM, BCAM, BCAM-ISO, LDCM, MFCM	none
SAM, QBRM	8
COMBO ²	2 -10
DMM-8, DMM-56K-8	1
DMM-15, DMM-56K-16	2
FSM-15	15
FSM-30	30
IRM	64

- 1 The RM can use backplane slots on a temporary basis to complete Wideband connections.
- 2 The COMBO uses up to two TDM slots for analog ports and up to eight for digital ports.

TDM channel allocation

The CM normally requires no backplane channels. The VX and CM-SAM require one channel.

The RM provides its own channels for connecting to other modules. For this reason it does not occupy any TDM slots.

The maximum number of 2 Mbit/s modules (IPMT, IPMN, E1M) in a 15 slot VX is limited by the total backplane bandwidth of eight 2 Mbit/s buses.

The RM and CM's Tone Generator and SAM signalling channel, and all low speed modules (defined below) only have physical access to TDM bus 0. In physical slots 1 to 7, the low speed modules have reserved access to TDM bus 0. In slots 8 to 15, buses 0 and 2 are crossed over, so that these modules have reserved access to TDM slots on bus 2.

Note *In a J1700 fitted with four 2Mbit/s modules, buses 0, 2, 4 and 6 are all allocated. In this event, the CM tone generator and SAM signalling channel are reallocated to TDM bus 1, timeslot 0.*

The low speed modules that use these reserved buses are:

- ADLM-V.24
- SDLM-V.24
- SDLM-V.35
- SDLM-X.21
- QDLM
- ALEM and ALEM-2
- ALPM and ALPM-2
- EMM and EMM-2.

If a low speed module is fitted in each half of the 15 slot chassis then they effectively use up 4 Mbit/s worth (64 timeslots) of the available backplane bandwidth even though they may only be using a few channels. For this reason, it is more beneficial to pack these low speed modules into one half of the chassis so that one of the reserved buses can be used by a 2 Mbit/s module.

If no low speed modules are fitted to one half of a chassis then the reserved 2 Mbit/s bus for that half becomes available for use by a 2 Mbit/s module.

If no low speed modules are used in the chassis, up to eight 2 mbit/s modules can be fitted.

TDM Bus access priority

The system is designed with a module hierarchy such that a module designated with a higher priority can take over a bus used by a lower priority module. If this occurs, a lower priority module will lose its TDM bus access and will not function. The highest access priority is given to the following low speed modules:

- ADLM, SDLM, QDLM, ALEM, ALEM-2, ALPM, ALPM-2, EMM and EMM2.

Next highest priority is given to the following Network/2Mb modules:

- IPMN (all models), IPMT (all models), E1M (all models), T1M.

The lowest priority is given to the following modules:

- BRMN, BRMT, QBRM, DBRM, BCAM, COMBO, DMM, DTM, FSM, IRM, HSDM.

Power supply/heat dissipation limits

The 5V supply in the 15 slot chassis is rated according to the PSM being used. Total current and related heat dissipation is dependent on the number and types of modules fitted to the chassis. Refer to the *Installation Manual* for current consumption figures for the modules.

Database limits

The number of line records required by each module is shown below:

Module type	Line records	Comments
SRMM	64	
IPMT, IPMN, E1M, T1M	32	
DTM	0 -128	
BCAM	8	
SAM	10	
ALEM, ALEM-2, ALPM, ALPM-2, EMM, QDLM	4	
BRMN, BRMT	4	
SDLM V.24, SDLM V.35, SDLM X.21	2	
HSDM	28 to 60	28 for Subrate switches, 32 for digital ports
QBRM	16	
BCAM-ISO	32	
LDCM, MFCM with MLQ-CMP	14 to 20	
MFCM with HD-VCM	91	
CM	5 to 587	
RM	78	See Note 1 below
RM (Version 4.0)	58	See Note 2 below
COMBO	18	
COMBO (Version 5.0)	30 to 38	2 for companders, 28 for Subrate switches, 8 for digital ports
DMM-8, DMM-56K-8	9	
DMM-15	17	
DMM-56K-16	18	
FSM-15	17	
FSM-30	32	
IRM	64	

Notes

1 The stated number of line records is composed of the following:

- 10 for UMPE, UMPS, tone, E2E, SAM, SSH and so on
- 16 for additional SAM B- or D-channels
- 12 for Packet Line interfaces
- 32 for default 15 VX calls
- 8 for TimeFrame Virtual Lines.

The actual number of line records required by the RM depends on whether the VX, ISDN Interworking and Virtual Lines are used. The line record usage over and above the stated number is as follows:

- *the VX uses 32 line records for each group of 15 calls that it is configured to handle. For example, if VX is configured to handle 75 calls then it will use 160 line records;*
 - *ISDN Interworking uses 62 line records for each group of 30 left/right interworking pairs. There can be up to four interworking groups using a total of 248 line records;*
 - *A maximum of 64 line records are allocated for all Virtual Lines. Each TimeFrame VL uses up to eight line records whilst all other VLs use one line record.*
- 2** *The stated number of line records is identical to the number stated in Note 1, with the exception of:*
- *There are 13 Packet Port Interfaces, as opposed to 12 Packet Line interfaces*
 - *The 8 line records for TimeFrame Virtual Lines are not automatically allocated.*

The actual number of line records required by the RM depends on whether the VX, ISDN Interworking and Virtual Lines are used. The line record usage over and above the stated number is as follows:

- *the VX uses 32 line records for each group of 15 calls that it is configured to handle. For example, if VX is configured to handle 75 calls then it will use 160 line records;*
 - *ISDN Interworking uses 62 line records for each group of 30 left/right interworking pairs. There can be up to four interworking groups using a total of 248 line records;*
 - *A maximum of 64 line records are allocated for all Virtual Lines excluding TimeFrame Virtual Lines;*
 - *A maximum of 23 line records are allocated for all TimeFrame Virtual Lines and Category 1 Packet Ports;*
 - *A maximum of 6 line records are allocated for the Category 2 Packet Ports;*
 - *A maximum of 160 line records are allocated for TimeFrame Resources.*
-

Submodule limits

In addition to the line record limit detailed above, there are further limits defined for software 'submodules'. The number of software submodules supported by each of the system controlling modules are as follows:

- Control Module supports a maximum of 48 submodules
- Resource Manager V1.xx supports up to a maximum of 48 submodules
- Resource Manager V2.xx supports up to a maximum of 64 submodules
- Resource Manager V4.xx supports up to a maximum of 106 submodules.

The minimum and maximum submodule requirements for each module or feature are detailed in the table below:

Module/Feature	Minimum submodules	Maximum submodules
CM Tone Generator and SAM	1	1
RM Tone Generator and SAM	1	1
Virtual Lines	0	2
TimeFrame Virtual Lines	0	8
CM VX (15 calls per submodule, 75 calls max)	0	5
RM VX (15 calls per submodule, 120 calls max)	0	8
Interwork Units (2 submodules per IWU, 4 IWUs max)	0	8
RM Packet Port	0	13
BRM (NT and TE)	1	1
DBRM (NT and TE)	2	2
QBRM-U (NT and TE)	4	4
DBRM-U (NT and TE)	2	2
QBRM (NT and TE)	4	4
T1M	1	1
E1M (all models)	1	1
HSDM	2	2
IPM (NT and TE)	1	1
DTM (1 submodule per interface)	1	4
Low speed modules (DLMs and Analog modules)	1	1
IRM	4	4
SAM	1	1
BCAM	2	2
VCM: Subrate Switches (32 per submodule) Companders (32 per submodule)	0 1	1 1

Module/Feature	Minimum submodules	Maximum submodules
COMBO	6	6
LDCM	2	2
MFCM (with MLQ-CMP)	2	2
MFCM (with HD-VCM)	3	3
FSM	1	1
DMM (All models)	2	2

Note *The number of submodules used in each chassis must not exceed the number supported by the installed CM or RM. If the number is exceeded, the device will not operate correctly.*

Absolute maximums

Based on the above limitations, the following table summarises the absolute maximum number of each type of module that can be used in each type of chassis.

Module type	Maximum number in		
	4 slot	6 slot	15 slot
IPMT, IPMN, E1M, E1T1	2	4	8
DTM-1 ¹	3	4	12
DTM-4 ¹	1	1	8
BRMT, BRMN, ALEM, ALEM-2, ALPM, ALPM-2, EMM, SDLM V.24, SDLM X.21, QDLM, BCAM, BCAM-ISO, SAM, QBRM	limited by available card slots	limited by available card slots	limited by available card slots
SDLM V.35	limited by available card slots	limited by available card slots	12
HSDM	3	3	7
VCM-32	limited by available card slots	limited by available card slots	6
VCM-16	limited by available card slots	limited by available card slots	7
VCM-8	limited by available card slots	limited by available card slots	8
VCM-4	limited by available card slots	limited by available card slots	8
SRMM	limited by available card slots	limited by available card slots	9
LDCM, MFCM	3	4	5
COMBO	3	0	0
FSM	3	5	8
DMM-8, DMM-56K-8	2	4	4
DMM-15, DMM-56K-16	2	4	4

¹ The number of lines and timeslots used by the DTM are determined by both the number of submodules fitted and the configuration of the submodules. Each DTM can have between 1 and 4 submodules, each with 1 to 32 timeslots.

Notes

- Where modules are grouped on the same row of the table, the limits apply to any mix of those particular modules. For example in a 15 slot chassis you can fit either eight IPMTs, or four IPMTs plus four E1Ms, but you cannot fit eight of each.
- For a random mix of modules, this table does not apply. In this case the above information must be used in order to determine whether a particular module mix is possible. For example, it is not possible to have eight 2 Mbit/s modules and six QDLMs because you have exceeded the backplane bandwidth limitation. Furthermore, you cannot have seven 2 Mbit/s cards with QDLMs spread over both halves of the chassis.

Appendix K

LDCM and MFCM Fax compatibility

The low Bit rate Voice Codec/Fax Relay Processor used in the Voice/Fax submodule is an AudioCodes AC4801F-C. This processor has been tested for compatibility with the following fax equipment:

AT&T 9015 PF	Okidata Okifax 2200
AT&T PPF200	Olivetti OFX-3100
Brother Intellifax 2500	Panasonic KXF-500
Brother Intellifax 950M	Panasonic KXF-3000
Canon B70	Panasonic PX-5
Canon Fax 750	Pitney Bowes 8050
Canon Fax TF-301	Ricoh 3500L
Canon L777	Ricoh Fax 240
Cardinal FaxTalk	Ricoh RF-05
Compaq 9600 PRMCIA	Samsung FX-1502
Dex 85	Sanyo SFX 11
Eiger 14.4 PRMCIA	Scout 14.4 PRMCIA
Fujitsu Dex 740	Sharp FO-445
Gammalink Fax Modem	Sharp FO-5400
Hayes Optima 14.4	Sharp UX-104
Hewlett Packard OfficeJet	Supra 28.8 MacIntosh
Konica 7310	Supra FAX Modem V.32bis (FaxTalk)
Minolta 3300	T30 and NSF/NSC
Mita LDC-570	Telecom AM-11
Monroe MX-4020	Telecom NP-80
Motorola CELlect 14.4	Toshiba TF-501
Motorola Power 14.4 (FaxTalk)	U.S Robotics Sportster
Motorola Power 14.4 (FaxWorks)	U.S Robotics Sportster 14.4 Internal
Motorola Power 14.4 (BitFax)	U.S Robotics Sportster 28.8 (MacIntosh)
Motorola Power 14.4 (QuickLink)	U.S Robotics WorldPort PRMCIA
Motorola Power 14.4 (WinFax)	Viva 14.4 Fax Modem
Motorola UDS 14.4	Xerox 3004
Muratec F-56	Xerox 7033
Muratec M-820	Zoom Telephonics 14.4 Internal
NEC Nefax 480	Zoom Telephonics V.32 bis

Appendix L

RM Bridge/Router Standards Conformance

The following table lists the Routing and Bridging standards to which the RM conforms:

Protocol	Standard Description	Document
IP Router	User Datagram Protocol (UDP)	RFC768
IP Router	Internet Protocol (IP)	RFC791
IP Router	Internet Control Message Protocol (ICMP)	RFC792
IP Router	Address Resolution protocol (ARP)	RFC826
IP Router	Transmission of IP Datagrams over Ethernet Networks	RFC894
IP Router	Reverse Address Resolution Protocol (RARP)	RFC903
IP Router	Internet Standard Subnetting Procedure	RFC950
IP Router	Internet Gateway Requirements	RFC1009
IP Router	Using ARP to implement transparent subnet gateways	RFC1027
IP Router	Transmission of IP Datagrams over IEEE 802 Networks	RFC1042
IP Router	Routing Information Protocol (RIPv1)	RFC1058
IP Router	MIB-II for Network management of TCP/IP-based Networks	RFC1213
TCP	Transmission Control Protocol	RFC793
TCP	MIB-II for Network management of TCP/IP-based Networks	RFC1213
STP	Media Access Control (MAC) Layer Bridges	IEEE802.1d
STP	Definitions of Managed Objects for Bridges	RFC1493
FR	Inverse Address Resolution Protocol (InARP)	RFC1293
FR	Management Information Base for Frame Relay DTEs	RFC1315
FR	Multiprotocol Interconnect over Frame Relay	RFC1490
SNMP	Simple Network Management Protocol (SNMP)	RFC1157
Bootp	Bootstrap Protocol	RFC951
Bootp	Bootp Vendor Information Extensions	RFC1497
Bootp	DHCP Options and Bootp Vendor Extensions	RFC1533
Bootp	Clarifications and Extensions for the Bootstrap Protocol	RFC1542
TFTP	The TFTP Protocol (Revision 2)	RFC783
TELNET	TELNET Protocol Specification	RFC854
Ethernet	Definitions of Managed Objects for Ethernet-like Interface Types	RFC1284

Appendix M

Resource Manager Telnet commands

The RM has an on-board Telnet Server that allows either a remote Telnet Client or the OmniVision Virtual Terminal to access configuration and status information.

To connect to the RM Telnet Server using a Telnet client on a PC or Workstation, a Management IP Address must be configured. For more information, see *Configuring the Management IP Address* on page 3-6.

This appendix summarises the Telnet commands for the Resource Manager.

Logon and Password

A default logon of username 'admin' and no password exists for the first time connection. To get help on the list of commands supported by the RM Telnet Server enter ? at the `rm:telnet>` prompt and press <enter>.

Note *A username and password is not required when accessing the RM Telnet Server via the OmniVision Virtual Terminal.*

Exit

To exit the Telnet Server, type `quit`, `exit` or `logout` at the `rm:telnet>` prompt.

Telnet commands

The following commands can be entered at the `rm:telnet>` prompt:

- | | |
|--|--|
| • <code>hostname [host]</code> | identify the RM by name |
| • <code>monitor</code> | divert Jexec monitor to Telnet |
| • <code>router</code> | start router configure and operate mode |
| • <code>tftp</code> | run TFTP Client application |
| • <code>configure</code> | general configure and operate mode |
| • <code>system</code> | show system status of RM |
| • <code>run</code> | show/set/clear the image file to run |
| • <code>download</code> | download software to a module in the chassis |
| • <code>reboot</code> | clear active sessions and reboot |
| • <code>ls</code> | list files in the Flash File System |
| • <code>dir</code> | as for <code>ls</code> |
| • <code>cat [filename]</code> | type a file onto the Telnet screen |
| • <code>type [filename]</code> | as for <code>cat</code> |
| • <code>mv [currentfilename] [newfilename]</code> | rename current file to new filename |
| • <code>ren [currentfilename] [newfilename]</code> | as for <code>mv</code> |
| • <code>del [filename]</code> | delete a file from flash |
| • <code>rm [filename]</code> | as for <code>del</code> |
| • <code>user</code> | show users, add/remove users |

- `echo` on or off
- `timeout` enable or disable a Telnet session timeout
- `logout` quit from the Telnet server (alias `exit`)
- `?` display main help

Monitor commands

Type `monitor` at the `rm:telnet>` prompt to start the monitor application.

- `monitor` start monitor application
- `?` list monitor commands
- `exit` exit monitor application

Note *The monitor commands are intended for the use of support personnel only.*

Router commands

- `router` start the router application
- `?` list the router commands
- `exit` exit the router application

The following commands are supported in the router application and these must be entered at the `rm:router>` prompt.

- `show stack ip route` show routing table
- `show stack ip cache` show route cache
- `show stack ip protstats` show protocol statistics
- `show stack ip portstats` show port statistics
- `show stack ip arptable` show ARP table
- `show stack ip rarptable` show RARP table

To exit the router application, type `quit` or `exit` at the `rm:router>` prompt.

TFTP commands

To access the TFTP client enter `tftp` at the `rm:telnet>` prompt. A `?` can be used to display the list of commands available within the RM TFTP Client. These commands include:

- `connect` connect to a TFTP server (e.g. `connect 192.195.112.15`)
- `put` send a file to the TFTP server (e.g. `put user.ini`)
- `get` receive a file from the TFTP server (e.g. `get rm.bin`)
- `status` display the state of connection to a Telnet Server
- `quit` quit the TFTP Client application
- `?` display TFTP help

Configure commands

- **hostname** [name] change hostname (maximum 10 characters)
- **show** show configuration or status parameter
 - **fr-int** [name] show Packet Line or Trunk interface config
 - **fr-route** [entry] show Frame Relay Route configuration
 - **compression** show data compression statistics
 - **lan** show LAN port settings for Bridge/Router
 - **wan** [port] show WAN port 1 to 8 settings for Bridge/Router
 - **ip** show IP Router global settings
 - **bridge** show Bridge global settings
 - **defaultroute** show IP Default Route
 - **staticroute** [entry] show IP Static Route(s)
 - **info** [pktport/trunk/psw] show pktport, trunk or PSW statistics
 - **info** [vc pktport] show packet port virtual circuit statistics
 - **call** [trunk] show TimeFrame Virtual Line statistics
 - **channel** [pktport/trunk] show channel statistics
 - **alarm** [trunk] show Alarms on TimeFrame VL
 - **?** display show help
- **reset** reset statistics parameter
 - **info** [pktport/trunk/psw] reset pktport, trunk or PSW statistics
 - **call** [trunk] reset TimeFrame Virtual Line statistics
 - **channel** [pktport/trunk] reset channel statistics
 - **?** display reset help
- **add** add configuration parameter
 - **fr-route** [entry] add a Frame Relay Route entry
 - **vc** [name] add a Virtual Circuit to a FR Interface
 - **staticroute** [entry] add IP Static Route
 - **?** display add help
- **delete** delete configuration parameter
 - **fr-route** [entry] delete a Frame Relay Route entry
 - **vc** [name] delete a Virtual Circuit from a FR Interface
 - **staticroute** [entry] delete IP Static Route
 - **?** display delete help

- **set** set configuration parameter
 - fr-int [name] set Packet Line or Trunk Interface
 - lan set LAN port for Bridge/Router
 - wan [port] set WAN ports 1 to 8 for Bridge/Router
 - ip set IP Router global parameters
 - bridge set Bridge global parameters
 - defaultroute set IP Default Route
 - ? display set help
- **clear** clear status parameter
 - compression clear data compression status
 - ? display clear help
- **ping [IP address]** test whether an IP address is contactable
- **save** save configuration to Flash now
- **quit** quit the configure application
- **?** display configure help

System command

Type `system` at the `rm:telnet>` prompt to get the following information:

- RM Version
- CM Functionality
- Serial Number
- Build Standard
- Modification Level
- Options
- Device Type
- Number of 2MBit/s card
- Country Code
- MAC Address
- Chassis Type
- Slot Position
- Dram Size
- Flash Size
- Daughter Card
- Power Supply 1 State
- Power Supply 2 State
- Power Supply 1 48V State
- Power Supply 2 48V State
- Backplane Clock Status
- Number of Packet Ports
- Number of TimeFrame trunks
- Number of WAN ports.

Run commands

- `run` (with no parameters) show which image is currently active.
- `run <filename>` select a specific software image file in Flash to execute. **<filename>** must be a valid filename that exists in the RM flash file.
- `run clear` clear the selected image
- `run ?` get help on the run command.

User commands

- `user` display list of users
- `user add [name] [password]` add user
- `user delete [name]` delete user

Rebooting the RM

To reboot the Resource Manager, enter the command `reboot now` at the `rm:telnet>` prompt. The RM will shut down any current activity and reboot after a period of five seconds.

Appendix N

QSIG-ETSI supplementary services interworking

The following table provides a list of QSIG and ETSI Supplementary Services (SS) as defined for the public ISDN. The table also shows the level of interwork support provided for each service in the current implementation of QSIG. The QSIG column indicates whether the SS is defined for QSIG. The ETSI column indicates that the SS is defined for the public ETSI ISDN.

Some QSIG supplementary services are coded such that they are compatible with ETSI. In these cases no mapping is required for them to work.

Supplementary Service	QSIG	ETSI	Interwork support
Advice Of Charge (at start, during and at end of call)	Yes	Yes	No interwork support provided. Service will not work
Call Waiting	Note 1	Yes	No interwork support required. Service will work
Call Hold	Note 1	Yes	No interwork support required. Service will work
Call Transfer	Yes	Yes	No interwork support provided. Service will work between ETSI and QSIG networks. In this release, notification of transfer will not be carried between the networks.
Calling Line ID Presentation	Yes	Yes	Interwork support provided. Service may work. The E1M-QSIG allows the relevant information, if provided by the signalling, to pass unmodified between the two networks.
Calling Line ID Presentation Restriction	Yes	Yes	Interwork support provided. Service may work. The E1M-QSIG allows the relevant information, if provided by the signalling, to pass unmodified between the two networks but does not act on any restriction indication.
Connected Line ID Presentation	Yes	Yes	Interwork support provided. Service may work. The E1M-QSIG allows the relevant information, if provided by the signalling, to pass unmodified between the two networks.
Connected Line ID Presentation Restriction	Yes	Yes	Interwork support provided. Service may work. The E1M-QSIG allows the relevant information, if provided by the signalling, to pass unmodified between the two networks but does not act on any restriction indication.
Closed User Group	No	Yes	Not defined for QSIG. No interwork support required.
Call Completion No Reply	Yes	Yes	No interwork support provided. Service will not work.
Call Completion to Busy Subscriber	Yes	Yes	No interwork support provided. Service will not work.

Supplementary Service	QSIG	ETSI	Interwork support
Conference Call Add on	No	Yes	Not defined for QSIG. No interwork support required. Users in the QSIG network may only be remote users, that is, they may be added to the conference call but may not control it.
Meet Me Conference	No	Yes	Not defined for QSIG. No interwork support required. QSIG subscribers may use the public service.
Call Forward Unconditional	Yes	Yes	No interwork support provided. In general, CFU with join will work. CFU with re-routing may fail if the PINX connected to the device treats it as a gateway PINX. Diversion information will not be sent between the two networks, that is, the calling party is not informed of the diversion.
Call Forward on Busy	Yes	Yes	No interwork support provided. CFB with re-routing may fail if the PINX connected to the device treats it as a gateway PINX. Diversion information will not be sent between the two networks, that is, the calling party is not informed of the diversion.
Call Forward on No Reply	Yes	Yes	No interwork support provided. CFNR with re-routing may fail if the PINX connected to the device treats it as a gateway PINX. Diversion information will not be sent between the two networks, that is, the calling party is not informed of the diversion.
Call Deflection	Yes	Yes	No interwork support provided. CD with re-routing may fail if the PINX connected to the device treats it as a gateway PINX. Diversion information will not be sent between the two networks, that is, the calling party is not informed of the diversion.
Freephone	No	Yes	Not defined for QSIG. No interwork support required. QSIG subscribers may use the public service.
Malicious Call ID	No	Yes	Not defined for QSIG. No interwork support required. The public network leg of the call can be traced
Multiple Subscriber Numbers	Note 1	Yes	No interwork support required. Service will work.
Direct Dialling In	Yes	Yes	No interwork support required. Service will work.
Subaddressing	Note 1	Yes	No interwork support required. Service will work.
Terminal Portability	Note 1	Yes	Interwork support provided. Service will work. Notification of suspension/resumption is passed between the two networks.
Three Party Call	No	Yes	Not defined for QSIG. Service will work between ETSI and QSIG networks, The controlling user is always in the ETSI network.
User-User signalling	No	Yes	Not defined for QSIG. Service will not work.
Calling Name ID Presentation	Yes	No	Not defined for ETSI. Service will not work.

Supplementary Service	QSIG	ETSI	Interwork support
Connected Name ID Presentation	Yes	No	Not defined for ETSI. Service will not work.
Calling Name ID Presentation Restriction	Yes	No	Not defined for ETSI. Service works by default.
Connected Name ID Presentation Restriction	Yes	No	Not defined for ETSI. Service works by default.
Path Replacement	Yes	No	QSIG service. Service may fail within the QSIG network when invoked for a QSIG to ETSI gateway call.
Call offer	Yes	No	QSIG service. No interwork support provided. Service will not work.
Do Not Disturb	Yes	No	QSIG service. No interwork support provided. Service will work but ETSI subscriber may not receive inband tone/announcement for DND.
Do Not Disturb Override	Yes	No	QSIG service. No interwork support provided. Not affected as users in the ETSI network may not use this SS.
Call Intrusion	Yes	No	QSIG service. No interwork support provided. Service will not work.
Recall	Yes	No	QSIG service. Will work between QSIG and ETSI networks although users in the ETSI network will not receive notification of the recall.
Transit Counter	Yes	No	QSIG service. No interwork support provided. Our equipment does not generate default Transit Counter for gateway calls to the QSIG network.
Route Restriction Class	Yes	No	No interwork support provided.
Message Waiting	Yes	Yes	No interwork support provided. Service will work if the message centre is part of the QSIG network. QSIG users may not interrogate the public network and vice versa.
Call Interception Immediate	Yes	No	No interwork support provided. Service will not work if a PINX connected to the device sees it as an intercepting PINX. That is, it considers that the device will complete the call to the intercepted party. Interception information is not sent between the two networks, that is, the calling party is not informed of the interception.
Call Interception on Busy	Yes	No	
Call Interception on No Answer	Yes	No	
Call Interception on Congestion	Yes	No	
Call Interception on Progress	Yes	No	
Common information	Yes	No	QSIG service No interwork support provided. Service will not work.
PINX Clock synchronisation	Yes	No	QSIG service No interwork support provided. Service will not work.
CTM Local registration	Yes	No	Interwork not applicable

Supplementary Service	QSIG	ETSI	Interwork support
CTM Authorisation	Yes	No	Interwork not applicable
CTM Incoming call	Yes	No	No interworking support provided. Service will work providing that the device is not treated as a re-routing PINX.
CTM Outgoing call	Yes	No	No interwork support required

Note 1 *No QSIG specification is defined, but the service may be provided by the PINX.*

Appendix O

Channel Bank T1 and E1 Signalling

The T1M and E1M-2 are used in the Channel Bank application to connect to a core network. The T1M supports Robbed Bit Signalling (RBS) and the E1M-2 supports Channel Associated Signalling (CAS).

T1M signalling

Automatic Ring-Down (ARD) applications are supported through the emulation of the PLAR (Private Line Automatic Ring-Down) protocol.

Manual Ring-Down (MRD) applications are supported through the emulation of the E&M Immediate Start protocol.

E1M signalling

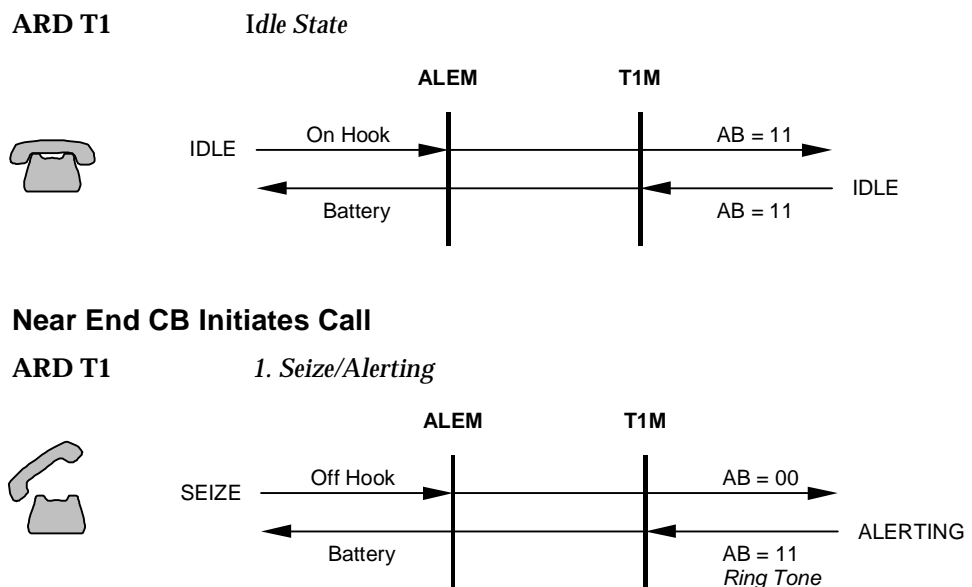
Both ARD and MRD applications are supported through the emulation of SSDC5A protocol (on-hook and off-hook states only).

T1M (ARD)

The T1M PLAR signalling protocol has the following RBS AB bit states.

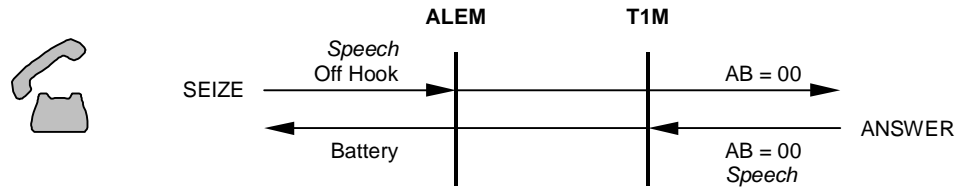
- On-hook Signal: AB = 11
- Off-hook Signal: AB = 00

The T1M only evaluates the input A bit for signal change.



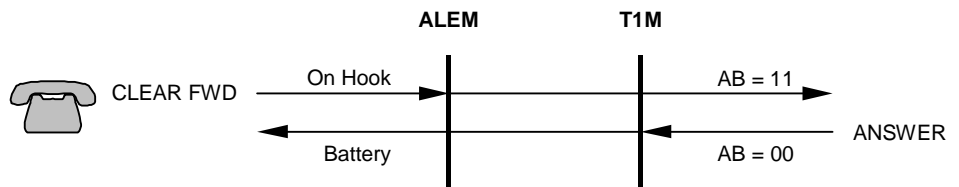
ARD T1

2. Answer/Active



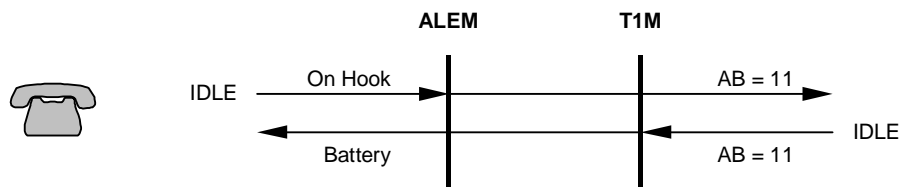
ARD T1

3a. Clear Forward (from Active)



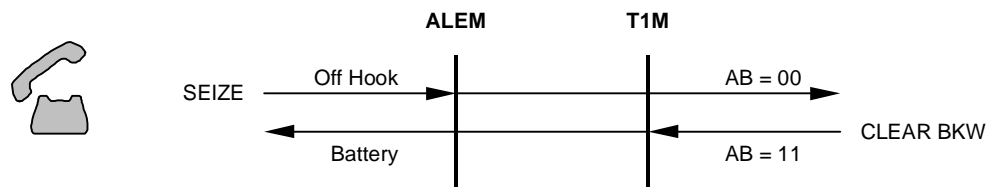
ARD T1

3b. Return to Idle



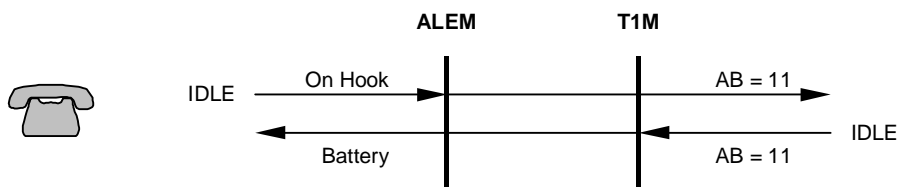
ARD T1

4a. Clear Backward (from Active)



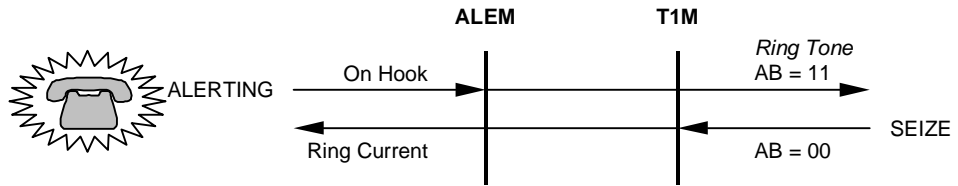
ARD T1

4b. Return to Idle

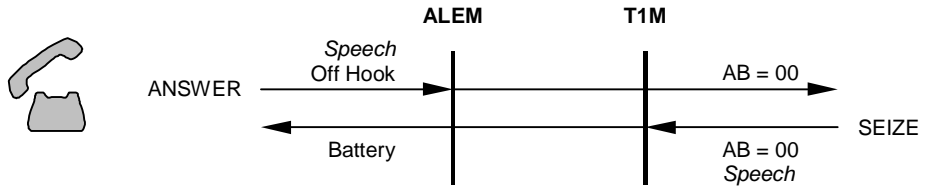


Far End CB Initiates Call

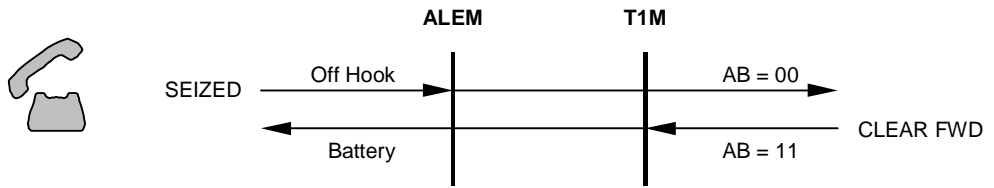
ARD T1 *1. Seize/Alerting*



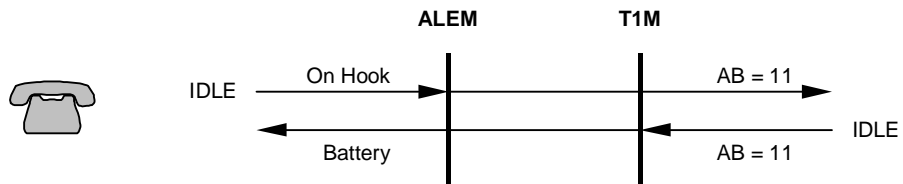
ARD T1 *2. Answer/Active*



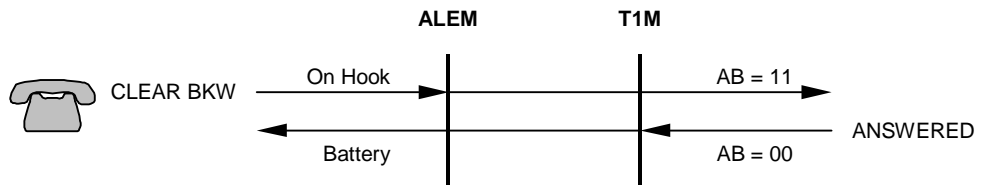
ARD T1 *3a. Clear Forward (from Active)*



ARD T1 *3b. Return to Idle*

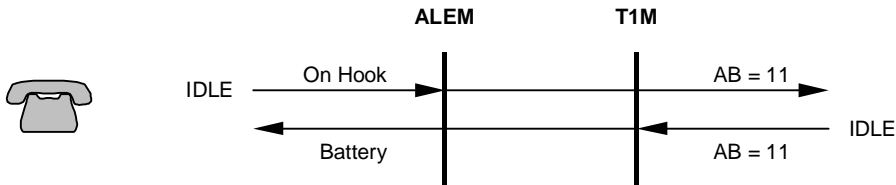


ARD T1 *4a. Clear Backward (from Active)*



ARD T1

4b. Return to Idle



E1M (ARD)

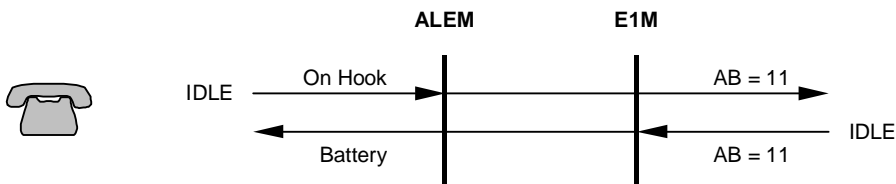
The E1M ARD signalling protocol has the following CAS AB bit states.

- On-hook Signal: AB = 11
- Off-hook Signal: AB = 01

The E1M only evaluates the input A bit for signal change.

ARD E1

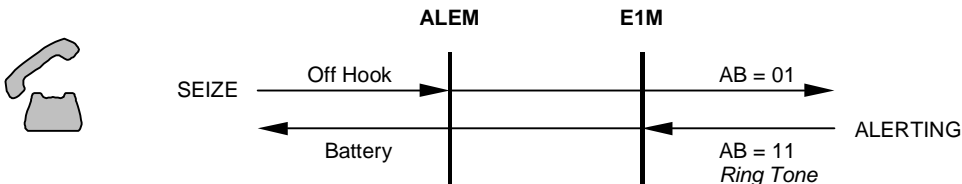
Idle State



Near End CB Initiates Call

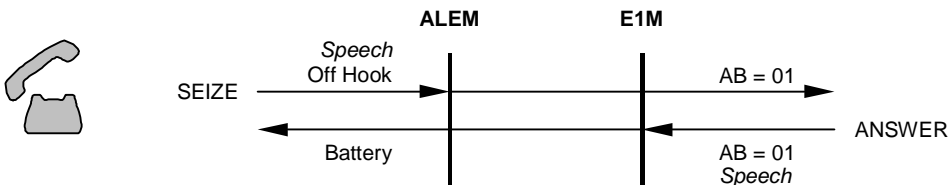
ARD E1

1. Seize/Alerting



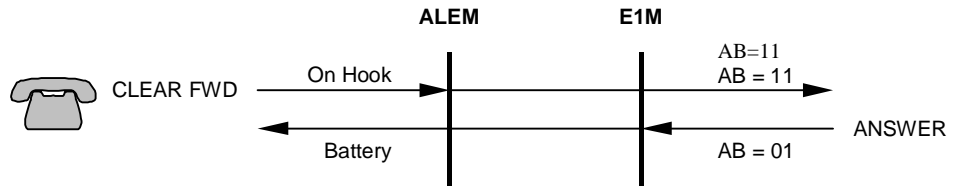
ARD E1

2. Answer/Active



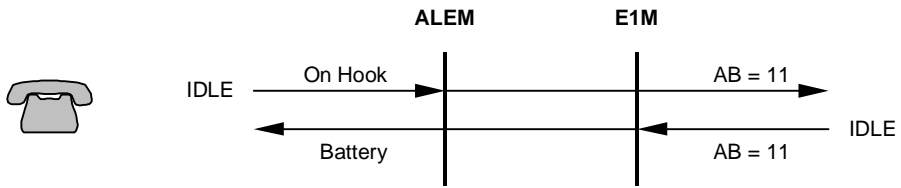
ARD E1

3a. Clear Forward (from Active)



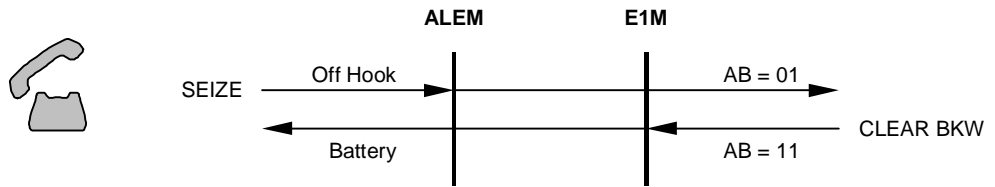
ARD E1

3b. Return to Idle



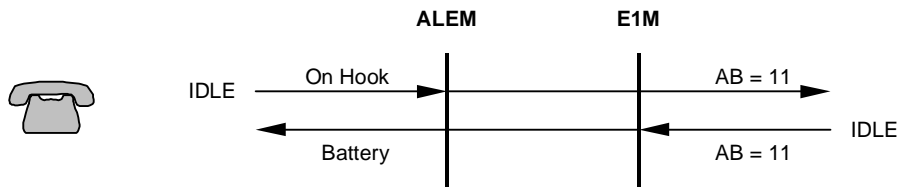
ARD E1

4a. Clear Backward (from Active)



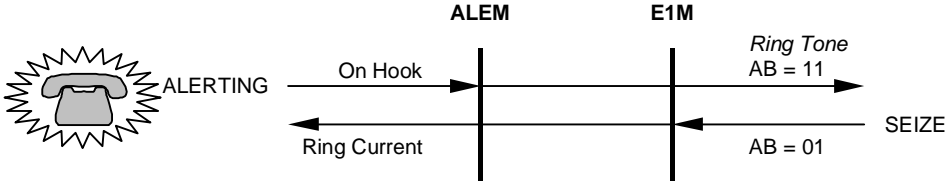
ARD E1

4b. Return to Idle

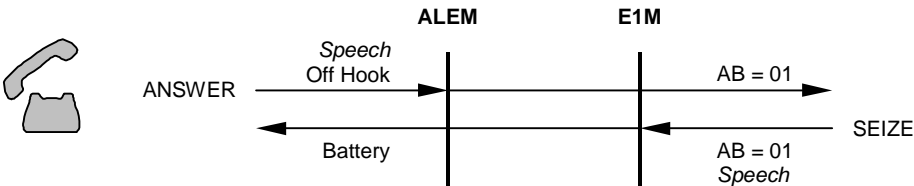


Far End CB Initiates Call

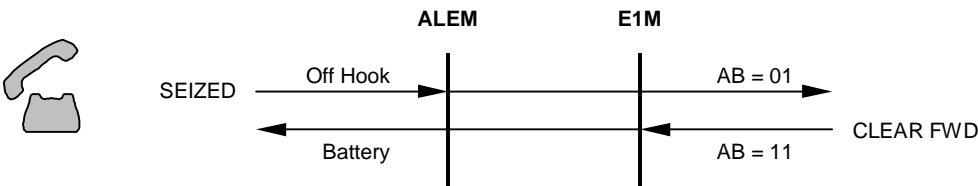
ARD E1 *1. Seize/Alerting*



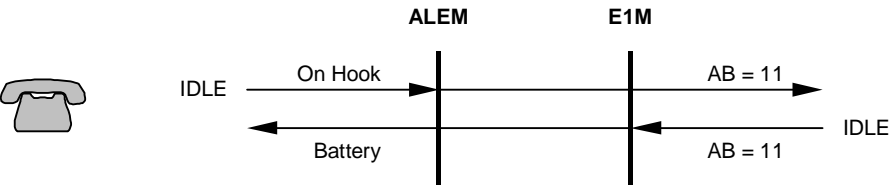
ARD E1 *2. Answer/Active*



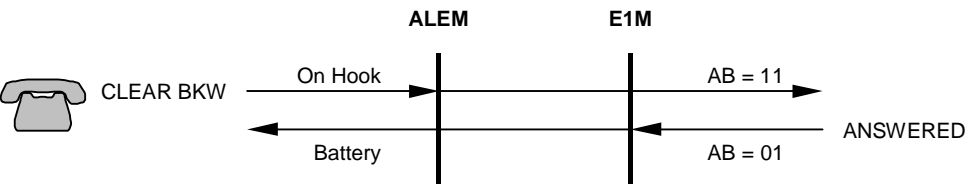
ARD E1 *3a. Clear Forward (from Active)*



ARD E1 *3b. Return to Idle*

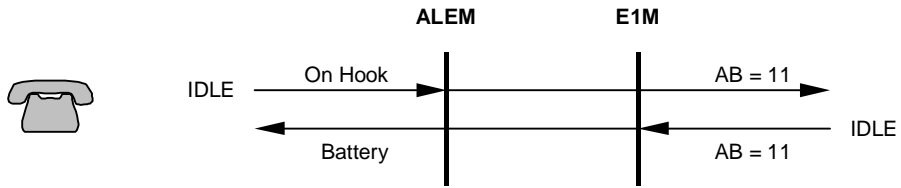


ARD E1 *4a. Clear Backward (from Active)*



ARD E1

4b. Return to Idle



T1M (MRD)

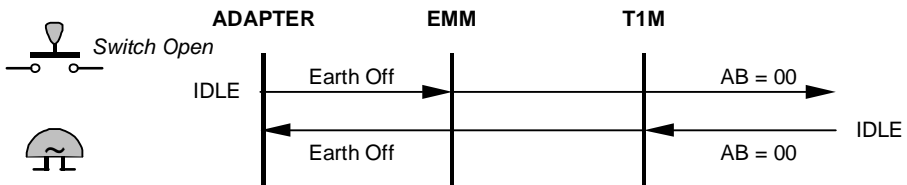
The T1M E&M Immediate Start signalling protocol has the following RBS AB bit states:

- On-hook Signal: AB = 00
- Off-hook Signal: AB = 11

The T1M only evaluates the input A bit for signal change.

MRD T1

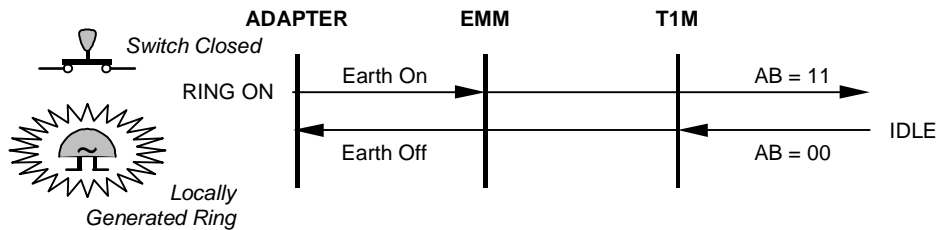
Idle State



Near End CB Initiates Call

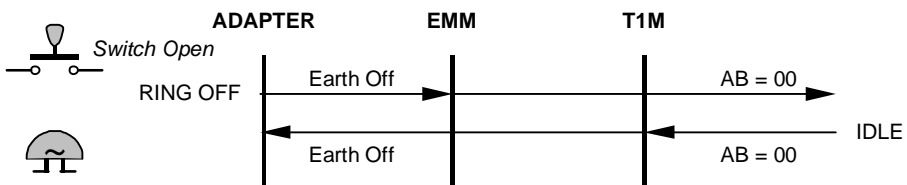
MRD T1

1. Ring On

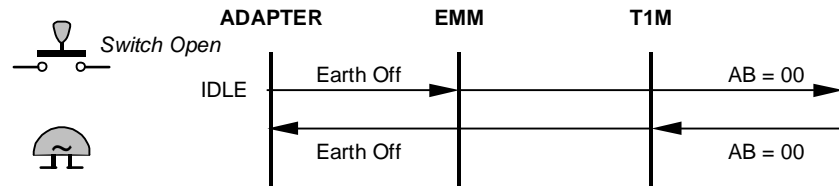


MRD T1

2. Ring Off

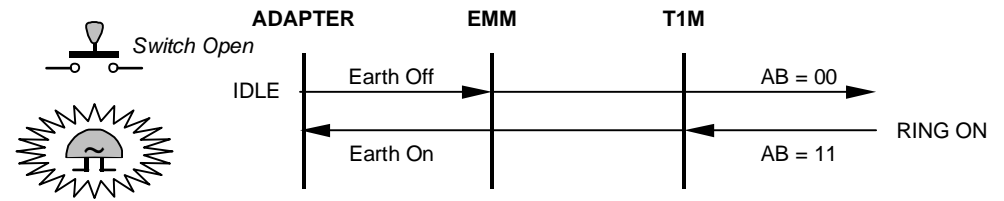


MRD T1 *3. Speech*

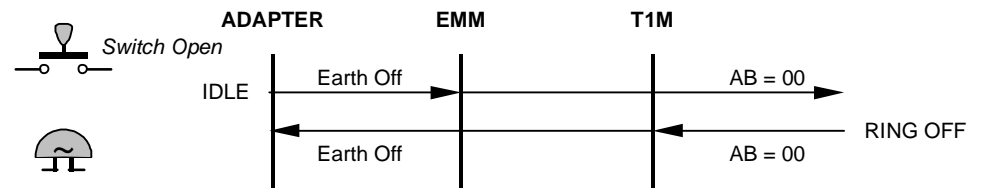


Far End CB Initiates Call

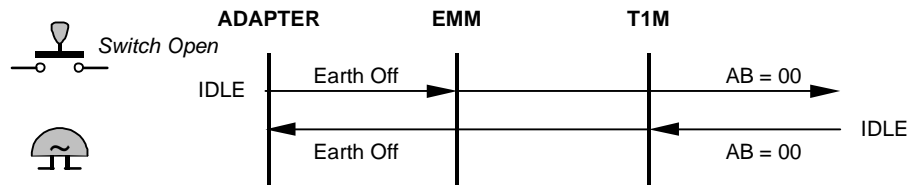
MRD T1 *1. Ring On*



MRD T1 *2. Ring Off*



MRD T1 *3. Speech*

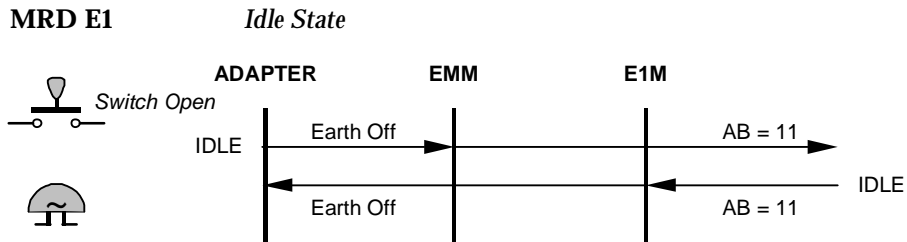


E1M (MRD)

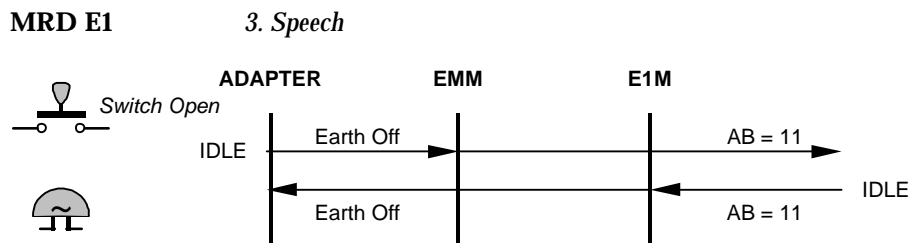
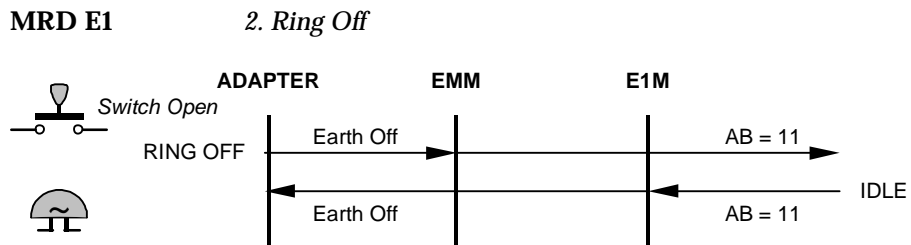
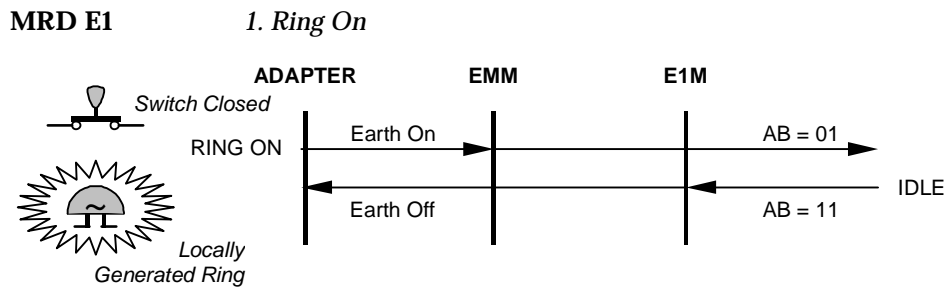
The E1M MRD signalling protocol has the following CAS AB bit states.

- On-hook Signal: AB = 11
- Off-hook Signal: AB = 01

The E1M only evaluates the input A bit for signal change.

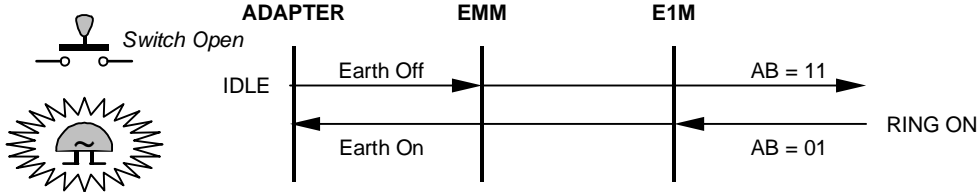


Near End CB Initiates Call

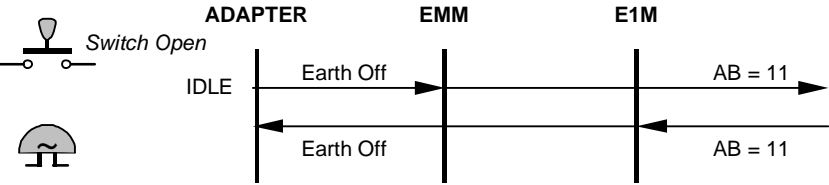


Far End CB Initiates Call

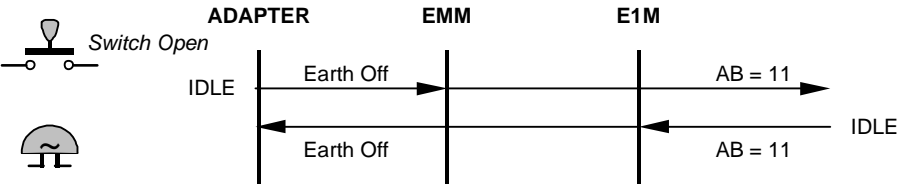
MRD E1 1. Ring On



MRD E1 2. Ring Off



MRD E1 3. Speech



Appendix P

PBX compatibility

This appendix details the PBXs that have been tested successfully with our equipment. The table below shows the PBX and its software release level against the Jtec equipment associated with the test. For J1000, the product release level is given. For J5000, the Resource Manager Version is provided.

PBX Type	PBX software level	Jtec Model		Protocol Tested
		J1000 Release	J5000 RM Version	
Nortel Meridian Option 51	16	9.1	N/A	DPNSS
Ericsson MD110	BCS 6.3.2	9.1	N/A	DPNSS
GPT SL1	X81	9.1	N/A	DPNSS
Nortel Meridian Option 61	21	9.1	N/A	DPNSS
Mitel SX 2000	M24.2.0	10.1	N/A	DPNSS
Nortel Meridian Option 11	18	10.1	N/A	DPNSS
Nortel Meridian Option 51	18	10.1	N/A	DPNSS
Ericsson MD110	BC 7.21	10.1	N/A	DPNSS
Siemens iSDX 3000	4.1	11.1	N/A	DPNSS
Nortel Meridian Option 11	20.19	11.1	N/A	DPNSS
Lucent Definity G3	V6i30.0.223.5	12.1	N/A	QSIG & DPNSS
		N/A	RM Ver. 2.16	QSIG
Ericsson MD110	BC9 & BC10	N/A	RM Ver. 1.56	DPNSS
Ericsson Mobility Server	1.4 & 2.0	N/A	RM Ver. 1.56	QSIG
Nortel Meridian Option 11C	22.46 & 23.47	N/A	RM Ver. 2.16	DPNSS
Nortel Meridian Option 81C	22.46	N/A	RM Ver. 2.16	DPNSS
Nortel Passport 6480	5.0.14	N/A	RM Ver. 2.16	QSIG
Siemens iSDX	6.1	N/A	RM Ver. 2.14	DPNSS
Siemens Hycom 300	3.51	N/A	RM Ver. 2.05	QSIG
Siemens Hycom 300	B3.513.16	N/A	RM Ver. 2.05	QSIG

Note *This table only supplies details of known systems against which our equipment has been tested. Please contact Jtec Customer Service and Support for information on systems not covered in this table.*

Appendix Q

Constant- and Variable-delay Subrate switches

This appendix details the number of constant- and variable-delay Subrate switches (SRS) available in each listed module:

Module	Version	No. of Constant-delay SRS	No. of Variable-delay SRS
SRMM	All	0	32
LDCM	All	0	12
MFCM	Pre V5.01	0	12
MFCM (with MLQ-CMP)	V5.01 onwards	12	0
MFCM (with HD-VCM)	All	16	0
COMBO	Pre V5.01	0	24
COMBO (with VC-G7231-2)	V5.01 onwards	24	0
COMBO (with HD-VCM)	All	16	0
HSDM	All	24	0
VCM	All	0	32

Appendix R

Working with V.90 technology

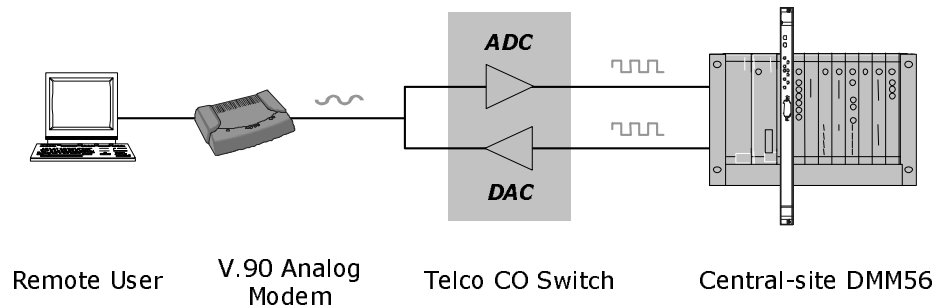
This appendix contains additional information on V.90 technology. It also provides tips for operating the DMM-56K module.

V.90 Technology

Compared to conventional modems, V.90 technology allows dial-up client modems to receive data from the PSTN at much higher speed (up to 56 kbit/s). This increase in performance results because the interior of the PSTN network is composed of all-digital circuits, which are much less prone to error in the presence of electrical noise than analog circuits. However by its very nature, the digital network limits the speed obtained in the reverse direction (typically to 33.6 kbit/s or less) due to the analog-to-digital conversion process required at the digital network edge.

Digital PSTN Networks

The PSTN network was originally designed for voice transmission and so consisted of analog lines and switches. However, over the last two decades, the interior of the PSTN network has been progressively upgraded to use all-digital, switched circuits (for example, ISDN B-channels). This means that the only remaining analog portion of the network is the copper line joining the customer premises to the central office (CO) switch in the local telephone exchange. A typical call connection from a user's modem to a digital V.90 modem system (such as a DMM-56K) centrally located in the PSTN network can therefore be shown as follows:



In the upstream direction (from the user modem to the central-site), the user signal is converted into tones and transmitted over the analog network (local loop) connection to the CO switch at the PSTN network edge. The CO switch samples the analog input 8000 times a second, and converts these samples to 8-bit PCM codes using an analog-to-digital converter (ADC). The resulting digital signal is transported over the digital portion of the PSTN network to the central-site V.90 modem.

Quantization Noise Limits Signalling Speed

When a continuous analog signal is converted to a discrete digital signal (in this case 8-bit PCM codes), information is lost from the analog signal. The difference between an original analog sample and its digitally encoded value is called quantization noise. This noise fundamentally limits the speed at which information can be sent over the channel. For this reason, V.90 analog modems use conventional modem modulation techniques to transmit in the upstream direction, leading to speeds of 33.6 kbit/s (or less depending on analog line noise) in this direction [1].

Achieving 56 kbit/s Speeds

From the above it is clear that analog-to-digital conversion introduces noise into the communications channel and therefore limits the maximum data speed obtained on the channel. However in the case of digital-to-analog conversion (DAC), where each unique 8-bit PCM code is converted into a correspondingly unique voltage level, no quantization noise is injected in the channel.

Referring to the figure above, we conclude that when the central-site V.90 modem is digitally connected to the CO switch, and there are no analog-to-digital conversions between the CO switch and the client modem's receiver, the communications path is not affected by quantization noise - these are the key requirements that enable a V.90 modem systems to achieve 56 kbit/s in the downstream direction [1,2].

Downstream V.90 Connections

In the downstream direction (from the central-site to the users modem) V.90 modems encode transmitted information using the same 256 (8-bit PCM) code words used for signalling across the digital portion of the PSTN network. At the CO switch, the DAC converts the received PCM codes into the (unique) voltage levels that are transmitted across the analog (local loop) circuit to the client modem receiver.

Under good line conditions, the transmitted voltage levels arrive at the client modem receiver without information loss. The client modem can therefore recover the digital PCM values transmitted by the central-site modem by examining the received levels. However, network impairments (such as noise) introduce decoding errors.

Network Impairments

In practice, error sources (for example, analog noise, echo, and crosstalk), inter-office PABX systems, and regulation of the maximum transmitted signal power allowed in some Telecommunications administrations, are the determining factors that limit the downstream speed to something less than 56 kbit/s.

Local Loop Noise

Network DACs follow non-linear conversion rules (μ -law PCM in the USA, A-law PCM in the rest of the world). Though adequate for voice communications, the non-linear rules present difficulties for V.90 modems in the presence of noise sources in the analog local loop.

The PCM codes representing low voltages produce smaller step changes in the DAC output voltages than PCM codes representing large values. Thus, analog noise in the local loop affects the closely spaced low voltage levels more than the larger levels. In practice, excess noise in the local loop means some proportion of the 255 PCM codes must be excluded from signalling scheme used by the V.90 central-site modem.

During call setup, the DMM-56K V.90 modems probe the downstream path to remove the PCM codes most susceptible to noise from the signalling scheme. The more noise encountered in the local noise, the greater the number of PCM codes eliminated. The result is a reduction in the maximum downstream speed (the rate is reduced by 1333 bit/s for each PCM code eliminated).

Local Equipment

Poor quality or noisy line terminations at the client end of the call can lead to further reductions in the maximum V.90 receive speeds achieved in practice. Equipment (for example, answering machines and cordless phones) sharing the same wall-socket as the client V.90 modem can be a source of additional noise in the local loop, and will therefore limit the maximum V.90 receive speed of the client modem. Disconnecting such devices may significantly improve the receive speed.

Office PABX Systems

Clients dialling through PABX devices often experience poor V.90 speeds. A PABX is a digital system that, apart from call switching, provides advanced features like voice mail, and call re-direction. However, these features come at a cost to the V.90 modem client, since a PABX uses analog-to-digital conversion on all inputs, and, in particular signals coming from a central-site V.90 modem.

During call setup, DMM-56K V.90 modems examine the downstream path to detect downstream analog-to-digital converters (such as those used in a PABX). When these are detected, the DMM-56K connects the call using a lower speed modem modulation standard (for example, V.34 or earlier modem standards).

FCC Regulation (USA only)

Due to FCC regulations, the maximum transmit speed in V.90 is limited to 53 kbit/s, even under the best line conditions. This is not a limit on speed as such - instead, to minimize crosstalk on the PSTN network, the FCC regulations restrict the amount of signal power transmitted to -12dBm. To meet this low power limit, V.90 central-site modems must use a subset (not all) of the 255 PCM codes, which limits the speed to around 53 kbit/s in FCC governed administrations.

Variable Network Paths

Modem users dialling in from a fixed location that ordinarily makes high-speed V.90 connections to the central-site, may at times experience poor V.90 connections to this same central-site. The pathway from the client-site through the digital portion of the PSTN network is not always the same. For example, the poorer quality pathway may include analog-to-digital conversions internal to the network (transcoding), different line balance amplifiers, and/or automatic network gain systems (digital pads), all of which cause lower speed V.90 connections.

Operating tips for the DMM-56K

The following paragraphs provide additional operating tips for the DMM-56K module.

Modem Call Setup

During call setup, the DMM-56K V.90 modems examine the downstream path to detect downstream analog-to-digital converters (such as those used in a PABX). When these are detected, the DMM-56K connects the call using a lower speed modem modulation standard (for example, V.34 or earlier modem standards). If the remote client modem requests a connection at V.34 (or lower standard) the DMM-56K attempts to connect the call using the requested modem standard.

Modem Connect Speeds

V.90 56 kbit/s speeds can only be achieved in the downstream direction (i.e., from the DMM-56K module to the remote user modem). In the reverse direction the data speed is typically limited to 33.6 kbit/s or less.

A modem usually produces a CONNECT XXXXX response once it connects and then negotiates the call parameters with a remote modem. Approximately 10 seconds into the call, the modems re-negotiate the call rate upward (or downward) based on their individual measurements of the maximum sustainable rate obtained by analysing the connection in data mode. The resulting call rate may differ from the rate displayed in the modem CONNECT XXXXX response.

Data Compression

Modem data compression methods (V.42 bis, MNP 5) cannot be used to achieve data rates greater than 57.6 kbit/s because the data rate across the backplane from a DMM-56K modem to an FSM is limited to 57.6 kbit/s. Therefore, remote client modems should disable data compression if the uncompressed rate could exceed the 57.6 kbit/s limit. Nonetheless, more efficient compression can be achieved across the link using the data compression features provided by the FSM (data compression at the PPP level, as controlled by the CCP - compression control protocol).

Windows Dial-Up Networking

If you enable the option 'bring up terminal screen after dialling' when using Windows 95/NT Dial-Up Network Client to connect to the central-site FSM, you may or may not see the characters '+++AT&V2' printed on the terminal screen prior to the FSM login prompt. Some login scripts may fail on seeing these characters. Well-designed scripts typically scan for the string 'ogin:' and will thus work as expected.

Appendix S

Management Calls across OnRamp Xpress

Telstra, Australia's OnRamp Xpress service provides only one in-dial number on the link when the Xpress tariff is applied to a Basic Rate service.

When using OnRamp Xpress with either TimeFrame or Subrate Virtual Lines, ensure that the Virtual Line has a **Line Number** with a **Subaddress**. This line number must be the in-dial number that is allocated to the Xpress service. The Subaddress can be arbitrarily allocated. The OmniVision **Manager Number** must then be set as the in-dial number of the Xpress service.

Incoming calls with the correct Subaddress will match onto the appropriate Virtual Line, while an incoming management call from OmniVision without a Subaddress will correctly match onto the Manager Number.

Subaddressing

Subaddressing is not available on the **Manager Number**, the **Router Table** or the **CLI/Number Checking**.

The first two restrictions imply that the **Manager Number** must always be the in-dial number without a **Subaddress**.

If you are using **CLI/Number Checking**, only the line number portion of the number can be changed, not the Subaddress. Likewise, when using TimeFrame and **CLI/Number Checking**, the number to be changed must be changed to the **Hotline Number** set in the TimeFrame Virtual Line Call Establishment Mode, excluding any Subaddress that may be used. This allows the calls to be matched and directed across the correct TimeFrame Virtual Line.

Glossary

2B+D ISDN transmission type comprising two 64 kbit/s B-channels and one 16 kbit/s D-channel.

30B+D ISDN transmission type comprising thirty 64 kbit/s B-channels and one 64 kbit/s D-channel.

ALEM Analog Line Exchange Module.

ALPM Analog Line Phone Module.

Advice Of Charge (AOC) a subscribed supplementary service supplied by the ISDN provider. The service provides the cost of nominated outgoing calls. Charging information is provided in the form of units, \$ per unit and packet charges on a per call basis.

analog (analogue) an electrical signal which varies constantly in voltage. This is unlike a digital signal which varies between two constant values, usually denoted as 0 and 1.

The value of the analog signal varies continually during transmission, whereas a digital signal only ever changes between two set values without any intermediate variations. See also *digital*.

Associated Lines List a list of all the lines which are members of a lineset.

area code telephone number prefix representing a geographic area.

asynchronous a way of transmitting data whereby each information character, or sometimes each word or small block, is individually synchronised. This is usually achieved by labelling the beginning and end of each character with additional bits, known as start and stop bits.

Also known as start/stop transmission. See also *synchronous*.

AT commands set of commands used by the Asynchronous Digital Line Module (ADLM V.24, QDLM V.24).

Austel (Australian Telecommunication Authority) national body responsible for setting and regulating telecommunications standards for network carriers and CPE within Australia.

Austpac a Telstra, Australia public packet switched data service. It is a flexible data network which can handle synchronous and asynchronous data streams of various speeds and protocols.

Auto Answer used when the called DTE is not using X.21 circuit switched call establishment and clearing procedures. An incoming call will be automatically answered.

BABT British Approvals Board for Telecommunications, the United Kingdom National approvals authority.

bandwidth range of signal frequencies that can be carried on a communication channel.

B-channel (also known as the bearer channel) an ISDN 64 kbit/s channel that carries customer information such as encoded voice, circuit-switched or packet-switched data.

base number can be defined when you configure IPMTs and BRMTs. It is subtracted from an incoming called party number and the call is directed to the line with the resulting number, for example:

4507020 (incoming) - 4507000 (base) = 20 (line called).

The base number is also added to the area code and line number to generate the 8 digit national significant number which is sent as the calling party number on outgoing calls, for example:

20 (line calling) + 4507000 (base) + 2 (area) = 24507020.

Basic Rate ISDN transmission type comprising two 64 kbit/s B-channels for user transmission and one 16 kbit/s D-channel for common channel signalling.

baud (Bd) a unit of signalling speed of a data transmission device. The speed expressed in bauds (Bd) reflects the number of discrete conditions or signal events per second.

BCAM B-Channel Aggregation Module.

bearer service the communication link of a service that provides for the transmission of digital information (for example, digitised voice and data) between user devices.

bidirectional a link whereby it is possible to transfer information simultaneously in both directions between two points.

bit (contraction of binary digit) a binary unit of information that can have one of two values — 0 or 1.

bit Rate rate of data transmission.

bit/s (also bps and bits/s) bits per second. Used to describe data transmission rate.

BOOTP Bootstrap Protocol.

BRA (Basic Rate) ISDN transmission type comprising two 64 kbit/s B-channels and one 16 kbit/s D-channel.

BRMN Basic Rate Module Network.

BRMT Basic Rate Module Terminal.

broadcast a unidirectional transmission to multiple receiving locations simultaneously.

byte a small group of bits, usually 8, that combine to represent a character or measurement of memory capacity.

call the temporary connection (or apparent connection, as perceived by the caller) of one terminal to another; for example, a telephone call.

Calling Line Identification and Verification Enhancement (CLIVE) an optional Jtec feature that can be fitted to any J1000/J5000 system.

Calling Party Number the number from which an incoming call originates.

Called Party Number the number to which an incoming call is directed.

call control a function performed by the network layer (layer 3) of communications. It provides call routing, number analysis, and call setup and clearance services.

Call Log call logging facility, which records details of sessions, module function and errors.

CAS (Channel Associated Signalling) a signalling method in which traffic signals carried by a single channel are transmitted in the channel itself or in a signalling channel permanently dedicated to it.

E1 modules incorporate a CAS option.

Timeslot 16 is used as the dedicated signalling channel for timeslots 1 - 15 and 17 - 31.

CCITT *see* ITU-T.

CCS Common Channel Signalling; for example, the D-channel of ISDN Primary Rate and Basic Rate interfaces.

CELP (Code Excited Linear Prediction) The voice compression coding technique used by the Low Delay CELP Module (LDCM).

channel a conceptual pipe that represents a specific portion of the information carrying capability of an interface, through which information (for example, control signals, voice, data) may be transported.

CLI/Number checking (Calling Line Identification or Calling Party Number checking). Means that calls can be accepted or rejected on the basis of the calling party's number.

CLI/Number list a list of calling party numbers against which incoming calls are checked. A list may be global, effecting all lines, or specific, affecting particular lines only.

CLIR (Call Line Identification Restriction) a Telstra service to which you can subscribe.

configuration set of values which control the operation of the device.

Configure mode mode which allows you to set or change values for the device parameters.

cyclic hunting a method of searching linesets for an appropriate line to which to direct a call. The search always starts one member further down the Associated Lines List than the previous call, regardless of whether that call was successful. This ensures that calls are distributed as evenly as possible amongst members, and that lines that are faulty but not blocked will not cause the service to fail continuously.

D-channel (also delta channel) an ISDN 64 kbit/s (Primary Rate) or 16 kbit/s (Basic Rate) channel that carries common channel signalling and user-to-user information.

data bit rate rate of data transmission (in bit/s).

Data Communications Equipment (DCE) equipment that establishes, maintains and terminates a connection.

Data Terminal Equipment (DTE) equipment which converts user information into data signals for transmission, and vice-versa, for example, a PC.

default a value which is present unless changed.

digital a signal that takes only one of two values, normally denoted as 0 or 1, during transmission. This is in contrast to an analog signal which can take any value between a minimum and maximum at any particular time during transmission. See also *analog*.

DLM Digital Line Module.

DOS Disk Operating System.

DTMF (Dual Tone Multiple Frequency) a signalling scheme that uses two audio tones to pass dialled numbers to the network.

DTR (Data Terminal Ready) control lead on V.24 and V.35 digital line cards. Defined as Interchange Circuit No.108.

duplex transmission data transmission in both directions, either simultaneously (full duplex) or alternately (half duplex).

E1M Exchange 1 Module.

E2E answer xfer end-to-end answering mechanism to support inter-PABX operation.

en bloc dialling a dialling system in which the number is transmitted as an entity.

end-to-end signalling signalling between parties at either end of a link, independently of the link signalling.

error a discrepancy between a computed, observed or measured value or condition and the true, specified or theoretically correct value or condition.

extension (Extn, Xtn) a locally provided terminal equipment service, for example, a telephone which is connected to a PABX.

facsimile (fax)

1. The process of scanning a document, converting the image into electrical signals for transmission to a remote receiver, and the conversion of the received signals into a copy of the original document.
2. A machine that performs the facsimile process.
3. The copy document that is produced by the facsimile process.

G.703 the ITU-T (CCITT) standard for the physical traits of transmissions over digital circuits.

gain increased signal power, usually after the signal passes through an amplifier, repeater or antenna. Normally expressed in decibels (dB).
The opposite of gain is loss (or attenuation), which is often described as negative gain.

group number allocated main directory number within the range allocated by ISDN provider. It is written on the label of the NT1.

Hotline dialling provides rapid call connection to the number sent in the setup message.

Hotline number a number which is sent in the setup message to provide rapid call connection.

I.460 ITU-T (CCITT) recommendation.

I.461 ITU-T (CCITT) recommendation – equivalent to X.30.

I.463 ITU-T (CCITT) recommendation – equivalent to V.110 and ECMA102.

icon a symbol used by an operating system or application to represent an event, object, tool, command, etc.

IDD (International Direct Dial) the facility whereby network service provider customers can place unassisted international calls to other customers.

Information Element (IE) information about the type of call and its destination is conveyed in messages to the network exchange (ISDN) by the user equipment. These messages are made up of various Information Elements, some of which are mandatory, others optional.

input/output (I/O) a general term for the equipment used to communicate with a computer; also the data involved in such communication.

interface a boundary between two systems, or two parts of one system, across which all of the signals that pass are carefully defined.
The definition includes the connector signal levels, impedance, timing, sequence of operation, and the meaning of signals.

IP Internet Protocol.

IPX Internet Packet Exchange.

IPMN ISDN Primary Rate Module Network.

IPMT ISDN Primary Rate Module Terminal.

ISDN Integrated Services Digital Network. An ITU-T (CCITT) standardised telecommunications service that provides for the transmission of voice, data and image, and a common channel digital signalling system. End-to-end digitised voice and data traffic is handled simultaneously on the same links via the same exchanges. Network access includes Basic Rate (2B+D ~ 144 kbit/s) and Primary Rate (30B+D ~ 2.048 Mbit/s).

ISDN is defined in the ITU-T (CCITT) I.400 series.

ISDN port interface to the ISDN.

ITU-T International Telecommunications Union - Telecommunications, formerly the CCITT (Comité Consultatif International de Télégraphie et de Téléphonie). The International Telecommunications Union is a part of the United Nations. It publishes a set of recommendations, which are used by equipment providers and service providers worldwide. Examples of ITU-T recommendations are X.25, V.24 and G.703.

ITU-T recommendations recommendations of the International Telecommunications Union, formerly the CCITT (Comité Consultatif International de Télégraphie et de Téléphonie). Relevant recommendations are shown in *Appendix B*.

kbit/s (also called kbps and kbits/s) kilobits per second (thousand bits per second). Used to describe data transmission rate.

layer a collection of related network-processing functions that together comprise one step of a hierarchy of functions. For example, the OSI reference model has seven layers.

LED (Light Emitting Diode) a semiconductor junction diode that emits light in either the visible or non-visible range. Uses include alphanumeric displays, status indicators (for example, power on/off, alarm reported, etc.) and light sources for fibre optic communications.

line a concept. Not necessarily a physical port, and not necessarily on the user or network side. An ALEM has four lines and an E1M has 31. We use a physical line number (for example, ALEM lines 1 top to 4 bottom), and a Line Number entered in the Line Configuration dialog box which is used to match the phone number of the line.

linear hunting a method of searching linesets. The search always starts at the top of the Associated Lines List. The call is directed to the first idle member on the list.

lineset a range of ISDN numbers associated with a group of ports. Called numbers in the range are directed to one of the ports. One or many numbers can be associated with one or many ports.

local device one to which the NMS computer is connected via the serial port on the Resource Manager, not over the ISDN.

local number telephone number without area code, for example, 450 7000.

log record of events.

loopback a diagnostic facility to assist in the determination of data transfer faults. The transmit and receive channels of a device are effectively connected, such that all received information is immediately re-transmitted to the source. Often used in conjunction with BERT (Bit Error Rate Testing) in digital systems.

Manager number a number defined in the CM and RM configuration which is matched to the device in order to receive D-channel calls. If absent, all incoming B-channel and D-channel calls are assumed to be for establishing a session between OmniVision and the device, to allow configuration of an unconfigured chassis. Must be numbered like other lines within the allocated ISDN range.

Mbit/s (also Mbps and Mbits/s) megabits per second (million bits per second). Used to describe data transmission rate.

Microsoft Windows an operating system which provides a graphical user interface for PCs.

mode a type of operation, for example, configure, monitor, etc.

modem (modulator/demodulator) device which transforms (modulates) a digital bit stream into an analog signal, and vice-versa.

monitor port interface to a PC running Jtec's Protocol Analyser Package (PAP).

MS-DOS Microsoft Disk Operating System.

ms (millisecond) one thousandth of a second.

Multiple Subscriber Numbering (MSN) a numbering system used in the U.K.

multiplex the process of transmitting multiple signals from different sources over a common cable, transmission line, or channel.

national significant number telephone number including the area code without the first leading zero.

network a series of points, or nodes, connected by communications channels.

Network Termination 1 (NT1) device which terminates the ISDN on your premises. The ISDN equivalent of a telephone socket.

Network Termination 2 (NT2) device between terminal equipment (TE) and the network termination 1 (NT1) which provides switching and/or multiplexing facilities.

Network Independent Clock (NIC) the clock of a digital synchronous device which is carried over a synchronous network and reproduced at the other end. The digital device clock is independent of the network clock. The ISDN low layer compatibility element (LLC) allows this to be configured. Jtec does not provide this feature, but provides access to the LLC to allow external devices to provide clock buffering.

node a point on a network that demands or supplies services, or where transmission paths are interconnected, typically by means of switching.

non-volatile memory retains information even if the storage device is switched off.

NRZ Non Return to Zero. NRZ signals maintain constant voltage levels with no signal transitions (no return to a zero voltage level) during a bit interval. See also *NRZI*.

NRZI Non Return to Zero Inverted. NRZI signals maintain constant voltage levels with no signal transitions (no return to a zero voltage level) but interpret the presence of data at the beginning of a bit interval as a signal transition, and the absence of data as no transition. See also *NRZ*.

octet a unit of data consisting of 8 bits (binary digits).

offline if NMS is offline, it is accessing the database file stored in the PC. There are no active sessions.

OmniVision™ Jtec's Network Management Program. A Windows™ 95, 98 or NT application for configuring and managing Jtec's products.

OmniVision Help online Help for OmniVision.

online a device is online if a session is established with the PC running OmniVision.

Optus an Australian carrier.

out-of-band signalling a signalling system in which signals are conveyed outside the band of frequencies normally used for message transmission, but are carried on (or are associated with) the transmission channel.

parity refers to the state of bits of data. Can be odd, even or none.

password the password required to leave Monitor mode, and to establish a session with an unconfigured device is Jtec (case sensitive). The password can be changed via OmniVision using either the Mode Menu, Set Mode Password command, or the Device Menu, Set Password command.

PC (Personal Computer) an IBM or compatible PC is needed to run the NMS software.

point-to-point a system in which all connections are permanent and direct.

port the physical entry and exit point (termination point) for information going into and out of a module, that is, the user interface.

PPP Point-to-Point Protocol.

Primary Rate (also PRA and 30B+D) an ISDN term that describes the 30B+D interface, that is thirty 64 kbit/s transmission links and one 64 kbit/s signalling channel. These are known as bearer links and the delta channel. The ITU-T (CCITT) recommendation is the I.430 series.

Private Automatic Branch Exchange (PABX) (also PBX) a type of Customer Switching System that connects to voice services via trunk lines that are provided by the network service provider. Allows optionally unattended incoming, outgoing and internal extension calls.

protocol a set of rules governing the information flow within a communications infrastructure, otherwise known as data link control. Protocols control format, timing, error correction and running order. Suites of protocols are often used in networks, with each protocol responsible for one part of a communications function.

Protocol Analyser Package (PAP) a Jtec package which runs on a PC and provides a display analysis of the ISDN layer 2 and layer 3 protocols and reports on the status of layer 1.

PSM Power Supply Module.

Public Switched Telephone Network (PSTN) The complete public telephone system, including telephones, local and trunk lines, and exchanges. Links are only held as long as a call is in progress. Local lines to telephones are usually analog in nature.

RAM (Random Access Memory) a storage location where information can be entered, stored and removed (written, held and overwritten), as required. A power source is required to maintain the contents of the memory.

re-boot restart a PC, forcing DOS to be re-loaded.
To re-boot, press the **Ctrl**, **Alt** and **Delete** keys simultaneously.

relative level (dBr) the relative (power) level of a point in a transmission system is the nominal power gain at the reference frequency, from a reference point to the point considered. Relative levels can be used to compare two or more points of a network with respect to power. One point of a network is usually defined as the reference point at 0dBr, from which other measurement points are derived.

remote device a device to which the NMS computer is connected over the ISDN.

RM Resource Manager.

retry try again.

S interface *see* ISDN port.

semipermanent connection (SPC) the equivalent of a tie line between two parties through the ISDN, set up by the ISDN provider. Can be deactivated to free the B-channel for other traffic. Applicable in Australia only.

semipermanent backup attempts a Hotline connection if the semipermanent connection drops out for more than three seconds. OmniVision gives no indication whether a semipermanent or switched call is operating. If a Hotline call is established, it must be manually cleared. Backup should therefore be used with care. Applicable in Australia only.

session refers to the relationship between the NMS PC and the device.

session status an indication of the relationship between the NMS PC and the chassis, displayed in the window title bar:

Online	session established between Omnivision and the chassis
Offline	no session established between OmniVision and the chassis
Uploading	configuration being sent from the chassis to OmniVision
Downloading	database file being sent from OmniVision to the chassis
Connecting	making connection between OmniVision and the chassis

signal a physical phenomenon whose characteristics may vary to represent information.

SNMP Simple Network Management Protocol.

SPC (semipermanent connection) ISDN equivalent of a leased line. Applicable in Australia only.

software lock a device which must be in place in the PC parallel port for OmniVision to run. Prevents the unauthorised use of software.

SRMM Subrate Multiplexer Module.

stop bit bit that comes at the end of a character.

STP Spanning Tree Protocol.

subaddress a suffix to a line or Hotline number which is not interpreted by the network. If a called party subaddress number Information Element (IE) is received on an incoming call, the subaddress is checked against the configured line subaddress. The line subaddress is sent, if configured, as the calling party subaddress IE on outgoing calls.

subrack the chassis which functions as the equipment housing. 15 slot chassis are designed to fit into standard 19 inch equipment racks. They measure 7 IU (that is, 7 x 44.5 mm) in height. Installation of a 1 IU vent panel above each subrack is recommended to allow for adequate airflow.

Subrate and subchannel a rate or channel of less than 64 kbit/s, in multiples of 8 kbit/s, for example, 8, 32 and 56 kbit/s.

supplementary service a type of telecommunication service that can only be used in conjunction with one or more bearer or teleservice. A subscription service provided by the ISDN provider.

synchronous a data transmission method whereby there is a constant interval between the transmitted bits, obviating the need for start and stop bits that are necessary for asynchronous transmission. Its sophistication allows for faster transmission rates. The method does, however, demand that the two devices involved in the interchange must be synchronised. This is achieved by one device providing the clock (usually the DCE) and the other synchronising to it. The ISDN is a synchronous network. See also *asynchronous*.

T309 an ISDN layer 3 timer. If T309 is enabled, calls are automatically cleared after a certain time following a layer 2 failure. Primary Rate calls are cleared after 15 seconds. Basic Rate calls are cleared after 2 seconds. The T309 disable option in OmniVision is only available for IPMTs and BRMTs with a version number greater than 3.0.

Telstra an Australian carrier.

telecommunications any transmission and/or emission, and reception of signals that represent signs, writing, images and sounds, or intelligence of any nature by wire, radio, optical or other electromagnetic systems.

telephony a form of telecommunication that allows the transfer of speech, bidirectionally and in real time.

telex teleprinter exchange network. System of text communication between text-only teleprinters at 50 bit/s (6 characters per second).

Terminal Adaptor (TA) device that adapts non-ISDN equipment to a standard ISDN interface, for example, a device that converts RS-232 to ISDN.

Terminal Equipment (TE) equipment that makes use of the ISDN. TE1 has a standard ISDN interface (for example, a digital telephone). TE2 does not have a standard ISDN interface (for example, an analog telephone), and requires a Terminal Adaptor (TA).

test number defined in RM configuration, for diagnostic loopback testing of B-channel connections to and from the device.

TFTP Trivial File Transfer Protocol.

tie line (also leased line) a private-line type communications channel that is used to link two CCSs. Provides interworking and/or traffic transfer capabilities. Also known as Tie Trunk and Leased Line.

Timelink™ tariff system with high flag fall and small increment provided by Telecom Australia on subscription. Mainly used for data transfer.

transmission the transfer of information from one point to another point, or points, by means of signals.

unidirectional pertaining to a link where the transfer of information is possible in one pre-assigned direction only.

V.24 asynchronous an asynchronous protocol recommended by ITU-T (CCITT) for the transmission of data over telephone circuits via modems. *see* asynchronous.

V.24 synchronous a synchronous protocol recommended by ITU-T (CCITT) for the transmission of data over telephone circuits via modems. *see* synchronous.

V.35 a protocol recommended by ITU-T (CCITT) for the transmission of data over telephone circuits via modems.

Virtual Line a named group of lines which can be treated as one for call control purposes. It has no physical attributes itself, but is defined by its members. The four types of Virtual Line are:

- subrate multiplexed
- non-subrate multiplexed
- broadcast
- TimeFrame.

Windows Help comprehensive on-screen help provided with Microsoft Windows packages.

X.21 a data protocol recommended by ITU-T (CCITT) to provide an interface for circuit switched data networks and data terminal.

XSPC Switched Semipermanent Connection. A facility which attempts to automatically re-establish a failed Hotline call at timed intervals.

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