Digital Voice
NetPerformer® System Reference

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NetPerformer Support of Digital Voice

1.1 Overview

This chapter introduces NetPerformer switched digital voice functions, including:

- Digital interfaces (“Digital Interfaces” on page 1-4)
- Types of digital voice connections supported (“Types of Digital Voice Connections Supported” on page 1-6)
- ISDN signaling (“NetPerformer Support of ISDN Signaling” on page 1-9)
- QSIG signaling (“NetPerformer Support of QSIG Signaling” on page 1-11)
- CAS signaling (“Channel Associated Signaling (CAS) on Digital Interface Cards” on page 1-15)
- Installation requirements (“Installation Requirements” on page 1-18).

1.1.1 Finding Detailed Information

In this Module

Turn to the following chapters for further information on setting up a NetPerformer network that supports switched digital voice:

- “Configuring Digital Voice Connections” on page 2-1
- “Modem Passthru” on page 3-1

NOTE: Special considerations concerning Modem Passthru in a SIP VoIP network are included in this chapter.

- “Monitoring Digital Voice Connectionsl” on page 4-1
- “Application Examples” on page 5-1.

In Other Modules of this Series

- To configure the Voice Mapping Table (**SETUP/MAP** parameters), turn to the chapter *Configuring the Voice Mapping Table* in the *Analog Voice* module of this document series
- To configure and monitor NetPerformer SIP VoIP functions, refer to the *Voice over IP (VoIP) Option* module
- To configure and monitor NetPerformer switched analog voice functions, refer to the *Analog Voice* module of this document series

NOTE: **The *Analog Voice* module also provides reference information on:**

  - All **LINK** and **CHANNEL** parameters
  - The NetPerformer Signaling Engine
  - DSP operations
  - Analog connections on digital interface cards
  - Tones generated by the NetPerformer
  - Line activation types
  - Hoot & Holler and Push To Talk applications.
- Information on advanced NetPerformer voice features is provided in the *Advanced Voice Features* module, which addresses:
  - Voice Traffic Control
  - Enhanced Dialing
  - Domain Dialing
  - Hunt Forwarding and Hunt Group Rules
  - Voice Traffic Routing
  - Supplementary Services on Analog Connections
  - Backup ISDN Phones
  - Custom Signaling.

1.2 Digital Interfaces

The NetPerformer optionally provides channelized digital connections that can be used to transport switched voice traffic between a central location and multiple remote sites. Each interface supplies a number of individual channels, each of which can be terminated in a separate digital circuit at a different remote site.

A channelized digital connection on the NetPerformer can be established using:

- An E1/T1 port on:
  - A built-in *span* on the SDM-9210, SDM-9606 RTM and SDM-9620 RTM
  - An E1/T1 interface card installed in a slot on the SDM-9220 and SDM-9230
- An ISDN-BRI port on an ISDN-BRI S/T interface card, which can be installed on the SDM-9220 and SDM-9230 only.

For a detailed description of these interfaces and interface card installation procedures, consult the *Hardware Installation Guide* for your NetPerformer product.

1.2.1 T1 Interface

NOTE: To select a T1 interface on an E1/T1 port, set the *Framer Type* port parameter to **T1**. Refer to "Configuring a T1 Physical Port (LINK)" on page 2-4.

- Provides a 1.544 Mbps interface with 24 timeslots, supporting up to 24 user channels per physical port
- A separate voice call can be set up on each user channel
- Each timeslot operates at 64 Kbps, or 56 Kbps if Robbed-bit Signaling is configured on the physical port (or **LINK**).

Consult the chapter *NetPerformer Support of Digital Connections* in the *Digital Data* module of this document series for information on:

- T1 Channel Assignment
- T1 Error Detection
- T1 Coding Scheme.

1.2.2 E1 Interface

NOTE: To select an E1 interface on an E1/T1 port, set the *Framer Type* port parameter to **E1**. Refer to "Configuring an E1 Physical Port (LINK)" on page 2-6.

- Provides a 2.048 Mbps interface with 32 timeslots, supporting up to 32 user channels per physical port
- A separate voice call can be set up on each user channel
- Each timeslot operates at 64 Kbps.

Two types of E1 interfaces are available, depending on the hardware installed:

- **E1-120:** 2.048Mbps interface at 120 Ohms. Requires no extra hardware or hardware strapping on the port.
- **E1-75:** 2.048Mbps interface at 75 Ohms. Requires installation of an RJ48 to dual BNC adaptor and hardware strapping on the port, or a G.703 balun. For details, consult the *Hardware Installation Guide* for your NetPerformer product.

Consult the chapter *NetPerformer Support of Digital Connections* in the *Digital Data* module of this document series for information on:

- E1 Channel Assignment
- E1 Error Detection
- E1 Coding Scheme.

1.2.3 ISDN-BRI S/T Interface Card

- Provides 2 Basic Rate Interfaces (or physical ports), each of which supports 2 bearer channels (or user channels) per port
- Operates at 64 Kbps per bearer channel
- Each port can be configured with 3 Multiple Subscriber Numbers on which incoming ISDN connections can be accepted.

1.3 Types of Digital Voice Connections Supported

Switched digital voice traffic can be carried over a variety of digital connections and signaling types:

- Trunk connections (see next section)
- Drop & Insert Mode (see "Drop & Insert Mode" on page 1-6)
- Transparent signaling, also referred to as DCME (see "Transparent Signaling" on page 1-7)
- Channel Associated Signaling (CAS) signaling (see "Channel Associated Signaling (CAS)" on page 1-7)
- Common Channel Signaling (CCS), including ISDN and QSIG (see "Common Channel Signaling (CCS)" on page 1-7).

1.3.1 Trunk Connections

A voice connection over the WAN can be configured on a digital interface by configuring a timeslot with a voice protocol. The NetPerformer supports:

- ACELP-CN
- G.711 a-law and μ-law (PCM64K)
- G.723
- G.726 with 16, 24, 32 or 40 Kbps packetization
- G.729 (Optional: Available only with SIP license activated on the unit.)
- MELP.

Voice compression algorithms are loaded in the DSPs as the voice protocols are configured on the digital channels. The voice traffic is transported over the WAN via the physical T1, E1 or ISDN-BRI S/T link.

1.3.2 Drop & Insert Mode

In Drop & Insert mode, timeslots on one digital interface can pass traffic directly through to other timeslots on another digital interface without interpretation or processing. Drop & Insert mode allows information to be exchanged between units with minimal software involvement. It is supported when the link is configured for CAS, CCS or no signaling (data only). Set the channel *Protocol* parameter to **D&I**.

Drop & Insert can be used in a digital PBX application when timeslots need to be divided between:

- Timeslots that must be processed (e.g. compression) and passed through a packet transport cloud (e.g. Frame Relay), and
- Others timeslots that must be routed directly to the PSTN.

1.3.3 Transparent Signaling

Transparent signaling is used when two PBXs have to communicate using different types of CCS signaling (e.g. ISDN-PRI, ISDN-BRI, QSIG). It is available on NetPerformer T1, E1 or ISDN-BRI S/T interfaces.

Transparent Signaling can be used only in applications where the NetPerformer units have a point-to-point connection, for example, between two digital PBXs. It allows the PBXs to send their signaling information in a transparent way over the NetPerformers. The advantage of this is that the NetPerformers do not have to interpret the PBX signaling, freeing up processing time and resources for voice/fax traffic.

In transparent signaling the timeslot that carries the signaling is passed transparently to the remote unit. For E1 this is timeslot 16, and for T1 it is timeslot 24. The information contained in the signaling timeslot can be passed as either uncompressed or compressed traffic, depending on the configuration. For further details and an example configuration, refer to “Transparent Signaling for Digital Voice” on page 5-5.

1.3.4 DCME

A special application of transparent signaling is DCME (Digital Circuit Multiplication Equipment). In a DCME application, the two NetPerformer units communicate directly to each other on dedicated, permanent connections. The digital channels are configured as R2, ISDN or Transparent connections. One unit simply switches all the traffic on a particular digital channel to the corresponding channel on the other unit, with very little mapping and no call routing required.

NOTE: When R2 is used, the connection goes up and down with the signals, and can handle different telephone lines on the same configured channel.

1.3.5 Channel Associated Signaling (CAS)

In this method, signaling bits are allocated throughout the 24-frame multiframe, rather than to a specific timeslot. There are four different signaling bits in the multiframe: A, B, C and D. Channel associated signaling can provide 4-state, 2-state or 1-state signaling, based on the number of independent signaling bits used.

The NetPerformer supports CAS on the T1 and E1 interface cards.

1.3.6 Common Channel Signaling (CCS)

This method uses different channels (either physical or logical) to convey the signaling information and the actual user information to the connected devices. The NetPerformer supports two Common Channel Signaling (CCS) methods:

- **QSIG:** An open standard for inter-PBX signaling, designed as a global signaling system for corporate networking and sophisticated communication services. QSIG is supported on the T1, E1 and ISDN-BRI S/T interface cards.

NOTE: On the NetPerformer, QSIG operates in fully switched mode, allowing every connection to support supplementary services (both channel associated and non-channel associated) even when connecting to different remote units.

- **Integrated Services Digital Network (ISDN):** A worldwide telecommunication service that uses digital transmission and switching technology to support voice and digital data communication. Different types of ISDN are used in Europe, Japan, and North America, and are supported on different NetPerformer interface cards, as described in the next section.

1.4 NetPerformer Support of ISDN Signaling

ISDN support is available for both voice and data traffic on a NetPerformer installed with an E1, T1 or ISDN-BRI S/T interface card. For voice applications, ISDN provides connectivity to user PBX equipment using standardized signaling methods.

NOTE: ISDN features are discussed further in the chapter *Data Transport Using ISDN*, in the *Digital Data* module of this document series.

The NetPerformer supports the following ISDN signaling types on its digital interface cards:

ISDN Type (Location)	NetPerformer Interface Card	LINK *Signaling mode* Parameter Values
European ISDN	E1 (PRI)	EURO-ISDN
	ISDN-BRI S/T	EURO-ISDN
Japanese ISDN	T1 (PRI)	NTT, KDD
	ISDN-BRI S/T	INS-NET, KDD
National ISDN (North American)	T1 (PRI)	4ESS, 5ESS, DMS100, NI2
	ISDN-BRI S/T	NI1, NI2, 5ESS, DMS100

Table 1-1: NetPerformer Support of ISDN Signaling

1.4.1 European ISDN Support

The NetPerformer supports ISDN used in Europe on both the E1 (PRI) and ISDN-BRI S/T interface cards. It can handle both voice applications and data connectivity using the same hardware interfaces.

To configure the physical port (LINK) with European ISDN signaling, execute the **SLOT/LINK** submenu of the **SETUP** command, and set the *Signaling type* on the LINK to **EURO-ISDN**.

1.4.2 Japanese ISDN Support

The NetPerformer supports ISDN used in Japan on both the T1 (PRI) and ISDN-BRI S/T interface cards. It can handle both voice applications and data connectivity using the same hardware interfaces.

To configure the physical port (LINK) with Japanese ISDN signaling, execute the **SLOT/LINK** submenu of the **SETUP** command, and set the *Signaling type* on the LINK to **NTT** (on T1 card only), **INS-NET** (on ISDN-BRI S/T card only), or **KDD**.

1.4.3 National ISDN Support

The NetPerformer supports National ISDN used in North America on both the T1 (PRI) and ISDN-BRI S/T interface cards. It can handle both voice applications and data connectivity using the same hardware interfaces.

To configure the physical port (LINK) with National ISDN signaling, execute the **SLOT/ LINK** submenu of the **SETUP** command, and set the *Signaling type* on the LINK to **4ESS** (on T1 card only), **NI1** (on ISDN-BRI S/T card only), **5ESS**, **DMS100**, or **NI2**.

1.5 NetPerformer Support of QSIG Signaling

1.5.1 QSIG Features

QSIG is a modern, powerful and intelligent inter-PBX signaling protocol designed as a global signaling system for corporate networking using sophisticated communication services. QSIG is an open standard that is supported by the world's leading PBX suppliers. It offers:

- Guaranteed interoperability between products from different PBX vendors
- Very flexible interconnection possibilities
- Additional services for corporate users.

The term *QSIG* is the name given to a family of standards that provides signaling across the Q-reference point. The Q-reference point is the logical signaling point between two PBXs configured as part of the same network. QSIG signaling is carried on timeslot 16 of the E1 interface, and on timeslot 24 of the T1 interface. The other timeslots are used as bearer channels.

1.5.2 Corporate Networking

Large companies often operate from multiple locations which are sometimes spread around the world. A sound and efficient communication infrastructure that shares information between the various offices is essential.

- To fulfill these communication needs, corporate networks are being created to allow for the exchange of information between all global locations. To reach a certain level of efficiency, these corporate networks require enhanced telecommunication services, which are usually unavailable from the PSTN alone.
- In many cases, high speed digital links have replaced analog "tie lines" between remotely connected PBXs. Furthermore, PBXs now need to support much more than voice traffic.
- PBXs now commonly support LAN, video-conferencing and fax connections. This creates networks such as the one shown in Figure 1-1.

To create multi-vendor networks, a standard was required to define a common signaling protocol between PBXs. QSIG (also known as PSS1) was defined by the European Computer Manufacturers Association (ECMA) for this purpose, and offers the following advantages:

- Permits the creation of networks using multi-vendor equipment
- Provides a platform for future development supported by leading PBX manufacturers
- Compatible with public ISDN as well as business applications developed for public ISDN
- Offers a wide range of supplementary services to enhance business communications.

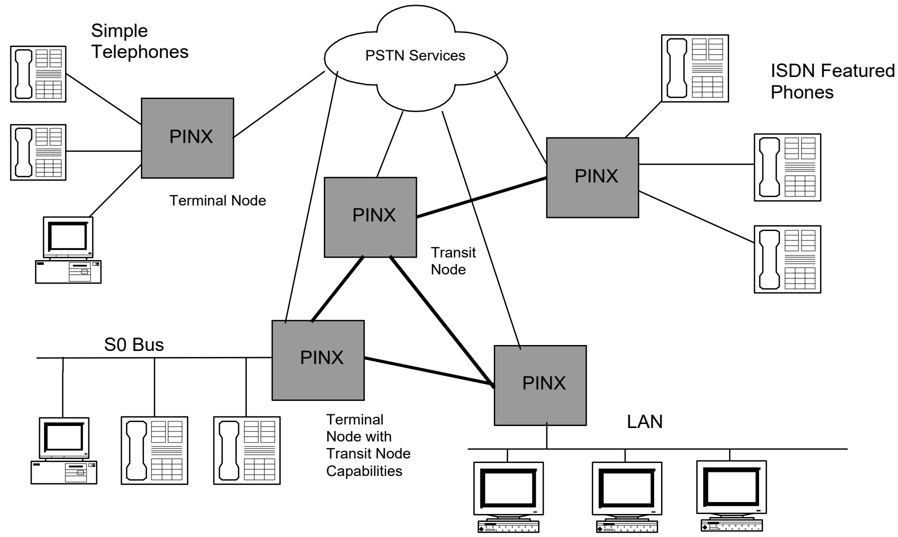

Figure 1-1: Typical Corporate Network

NOTE: The term *PINX* in Figure 1-1 refers to Private Integrated Services Network Exchange, a generic term for various types of corporate networking equipment such as PBX, multiplexer, CENTREX and so on.

When PBXs are linked together using QSIG, the facilities and services traditionally available between extensions of one PBX become available to all extensions of all PBXs in the network.

- This includes services such as Conferencing, Intrusion, Do Not Disturb and Call Back When Free Or Next Used.
- Without QSIG, these services are available only if they are part of a proprietary solution offered by the PBX manufacturer, and are often limited to the manufacturer's product line alone.

1.5.3 QSIG Protocol

The QSIG signaling protocol enables PBXs from different manufacturers to communicate intelligently, independently of the network carrier facilities. QSIG supports virtually any topology, and does not impose any limits on the number of nodes (PBXs) included in the network. Nodes can be connected in meshed or star topologies and can be used as:

- **Terminal Nodes:** Usually located at the customer premises, these nodes service a number of users, devices and LANs.

- **Transit Nodes:** Used as networking devices, these nodes route the information transparently from one node to the other. Information elements unknown by a transit node are forwarded to the destination with no interpretation required. This ensures compatibility between products and versions, and allows for equipment scalability.

Figure 1-1 shows both Terminal Nodes and Transit Nodes.

Besides defining the basic connectivity between nodes, QSIG includes a wide set of Supplementary Services that extend the capabilities of the overall network. The PBXs must send signaling messages to each other for management of the connection. Supplementary Services have to do with the management of the voice channels. For example, Supplementary Services are used to:

- Divert calls
- Display information about the calling and called user
- Notify that a specified user has just become available.

NOTE: When an analog phone is used the voice messages will be transferred from one end of the connection to the other, but the Supplementary Services will not go through. This is also the case for analog NetPerformer voice connections that call into a QSIG PBX. ANI information from a basic call setup can be transferred from R2 interfaces as well as FXS to an ISDN/QSIG voice connection.

The protocol stack model used by QSIG contains some similarities with ISDN. In particular, the physical and link layers are similar. The NetPerformer takes advantage of these similarities, and offers QSIG support on the same interface cards as ISDN-PRI (E1 or T1) and ISDN-BRI S/T.

1.5.4 QSIG Functionality on the NetPerformer

On the NetPerformer SDM-9230, SDM-9360, SDM-9380 and SDM-9585, QSIG is deployed as a software-configurable signaling mode option on the E1-120, E1-75, T1 and ISDN-BRI S/T interface cards.

1. QSIG can be used between a NetPerformer unit running V9.x and higher and a legacy NetPerformer unit running V7.2.X/V7.3.X, since their basic voice connections are fully compatible. **However, the support of QSIG supplementary services in V9.x and higher is not compatible with the method used in V7.x. For this reason, QSIG may not provide satisfactory results if supplementary services are required between, for example, an SDM-9585 running V9.x and higher and an SDM-9400 running V7.x.**

With QSIG support the NetPerformer offers a multitude of possibilities for connecting networking devices:

- For example, it allows for the connection of multiple PBXs so that they appear as a single (larger) PBX to their users. This creates an enhanced communication infrastructure for businesses whose offices are located in diverse locations.
- QSIG devices can communicate through a NetPerformer-managed WAN link transparently, providing a cost-effective solution to a complex network topology.

In Figure 1-2, all QSIG messages originating at one end of the connection are carried through the WAN connection to reach their destination at the remote site.

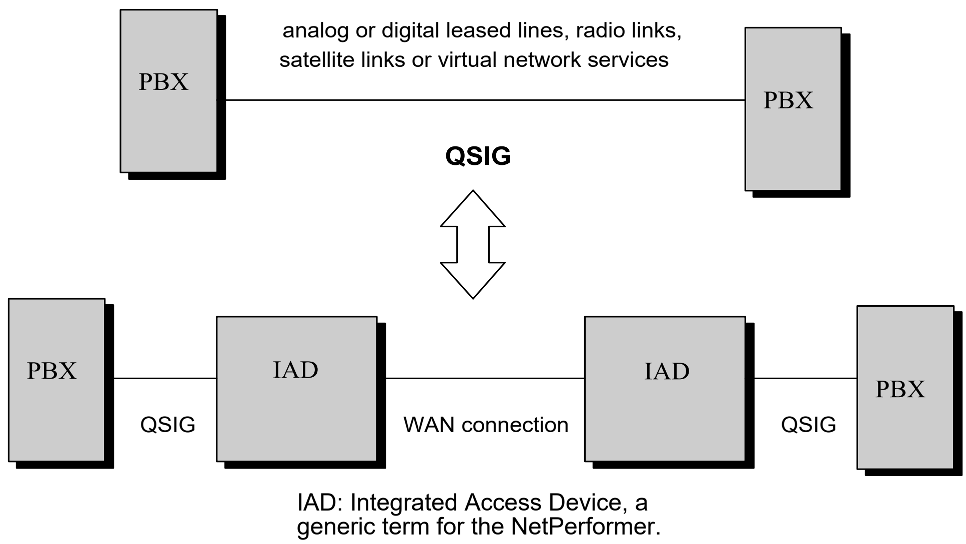

Figure 1-2: Basic QSIG Functionality in the NetPerformer

1.6 Channel Associated Signaling (CAS) on Digital Interface Cards

In addition to supporting digital voice connections, a T1 or E1 port on a NetPerformer digital interface card can also connect to an analog telephone or PBX, using Channel Associated Signaling (CAS).

- The *Signaling mode* on the physical port (**LINK**) must be set to:
  - **NONE** or **ROB BIT** on a T1 interface card
  - **NONE** or **CAS** on an E1 interface card
- The *Protocol* on the logical channel must be set to **ACELP-CN**, **G729**, **G729a**, **G723**, **G726** or **PCM64K**.

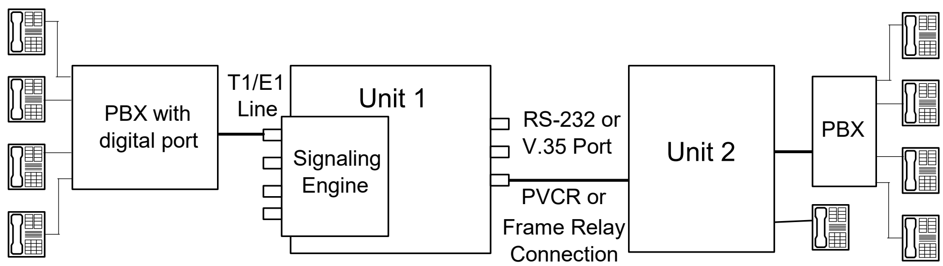

Figure 1-3: Digital Voice Support Through the Signaling Engine

1.6.1 Signaling Variations Supported

Depending on type of interface card and type of PBX, various signaling methods can be used:

- **E&M Immediate Start:** An E&M type where transmission takes place immediately. This is the industry standard for E&M operation
- **E&M Wink Start:** An E&M type where the unit toggles the A/B-lead before the PBX will transmit dial digits
- **FXO:** CAS emulation of FXO
- **FXS:** CAS emulation of FXS
- **GND FXO:** CAS emulation of an FXO type that uses ground start instead of loop start signaling
- **GND FXS:** CAS emulation of an FXS type that uses ground start instead of loop start signaling
- **PLAR:** Private Line Automatic Ringdown, a CAS signaling type used with certain types of channel banks
- **R2:** (on E1 interface card only) A CAS signaling type found in Europe that is similar to loop or ground start signaling in North American installations

- **R2-CHINA:** (on E1 interface card only) A CAS signaling type found in China that is similar to loop or ground start signaling in North American installations
- **CUSTOM:** To fine-tune the NetPerformer for non-standard signaling types. For details, turn to the chapter *Custom Signaling* in the *Advanced Voice Features* module of this document series.

Card	Signaling	Ports per Card	Channels per Card
T1	IMMEDIATE START, FXO, FXS, GND FXO, GND FXS, PLAR, WINK START, CUSTOM	1	24 (data or digital voice)
E1	IMMEDIATE START, FXO, FXS, GND FXO, GND FXS, PLAR, WINK START, R2, R2-CHINA, CUSTOM	1	30 (digital voice), 31 (data)

Table 1-2: CAS support on digital interface cards

1.6.2 Hybrid Voice and Data on Digital Link

A hybrid digital environment can support both voice and data connections simultaneously on the same digital interface.

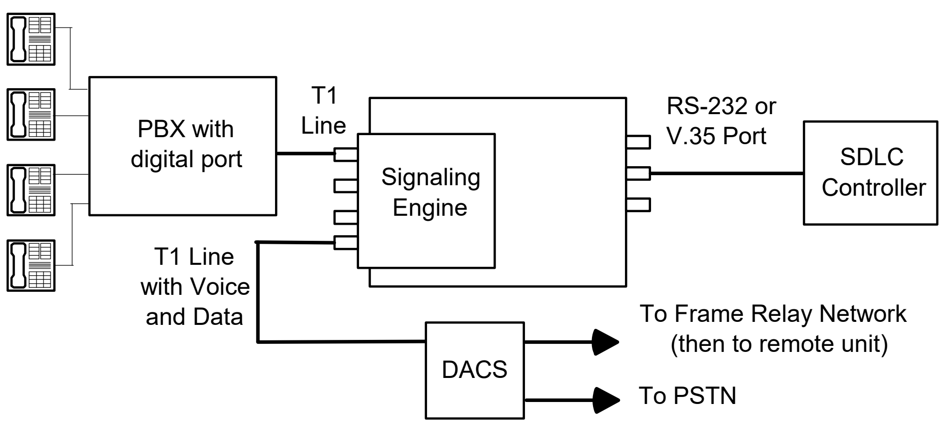

Figure 1-4: Hybrid Digital Environment

In the example in Figure 1-4, two T1 interface cards are used.

- The first T1 port connects the NetPerformer unit to the PBX.

  The PBX is configured so that some timeslots are assigned to external calls, using the PSTN, while the remaining timeslots are used for inter-office voice connections using Frame Relay.

- The second T1 port is connected to the DACS.

  The Signaling Engine switches the *external calls* timeslots on the PBX from the first T1 port (connected to the PBX), to this T1 port.

Information on these timeslots eventually ends up at the central office, for public telephone calls.

- For the remaining *inter-office* timeslots on the T1 connection with the PBX, the NetPerformer unit is responsible for transferring the voice information to the destination units.

  Voice traffic exchanged over these timeslots is sent to the remote destination through the NetPerformer-managed Frame Relay or PVCR links.

- Connection to the Frame-Relay network is established through data channels using the remaining timeslots on the T1 port connected to the DACS.
- Timeslot assignment must also be configured at the DACS to ensure that some timeslots go over the PSTN while others go over the Frame Relay network.

1.7 Installation Requirements

To permit digital voice connections to multiple remote locations using T1, E1 or ISDN-BRI interface cards, you must install the interface card appropriate to your application in a slot on the unit chassis. For complete instructions, consult the *Hardware Installation Guide* for your product.

- Each interface card supplies a number of individual channels, each of which can be terminated in a separate digital circuit at a different remote site:
  - **ISDN-BRI S/T interface card:** Provides two ISDN-BRI connections with a total of 4 bearer channels
  - **E1-75 or E1-120 interface card:** Provides one CAS, ISDN-PRI or QSIG connection with up to 30 B-channels and 1 D-channel
  - **T1 interface card:** Provides one CAS, ISDN-PRI or QSIG connection with up to 23 B-channels and 1 D-channel.
- Both ISDN-PRI and CAS signaling are supported on the same interface cards. However, the two signaling types cannot coexist on a single interface.
- No external CSU/DSUs or associated cabling are required at the NetPerformer site.

2

Configuring Digital Voice Connections

2.1 Configuration Overview

To configure the NetPerformer for a digital voice application:

1. Configure the physical port on the digital interface card, using the **SLOT/LINK** option of the **SETUP** command.

   The *Signaling mode* parameter on the physical port (**LINK**) determines the signaling type:

   - **For a digital voice application, the *Signaling mode* must *not* be set to NONE (the default value).**
   - **To support an analog telephone or PBX, the *Signaling mode* parameter *must* be set to one of the following values:**
     - **NONE**, the default value
     - **ROB BIT**, available on the T1 interface card
     - **CAS**, available on the E1 interface card

   The other parameters listed at the console are slightly different for the three digital interface cards (T1, E1 or ISDN-BRI S/T) and the various signaling types. Refer to:

   - "Configuring a T1 Physical Port (LINK)" on page 2-4
   - "Configuring an E1 Physical Port (LINK)" on page 2-6
   - "Configuring an ISDN-BRI S/T Physical Port (LINK)" on page 2-8.

2. Set up all required voice channels using the **SLOT/CHANNEL** option of the **SETUP** command (see "Configuring the Digital Voice Channels (CHANNEL)" on page 2-11)

3. Define the Voice Mapping Table with all required speed dial numbers, remote locations and calling characteristics, using the **MAP** option of the **SETUP** command. For details, refer to the chapter *Configuring the Voice Mapping Table* in the *Analog Voice* module of this document series.

NOTE: The Voice Mapping Table for a unit installed with the SIP VoIP licensed software option is detailed in the *Voice over IP (VoIP) Option* module of this document series.

Neither Phone profiles nor Caller IDs are required for a voice application using ISDN.

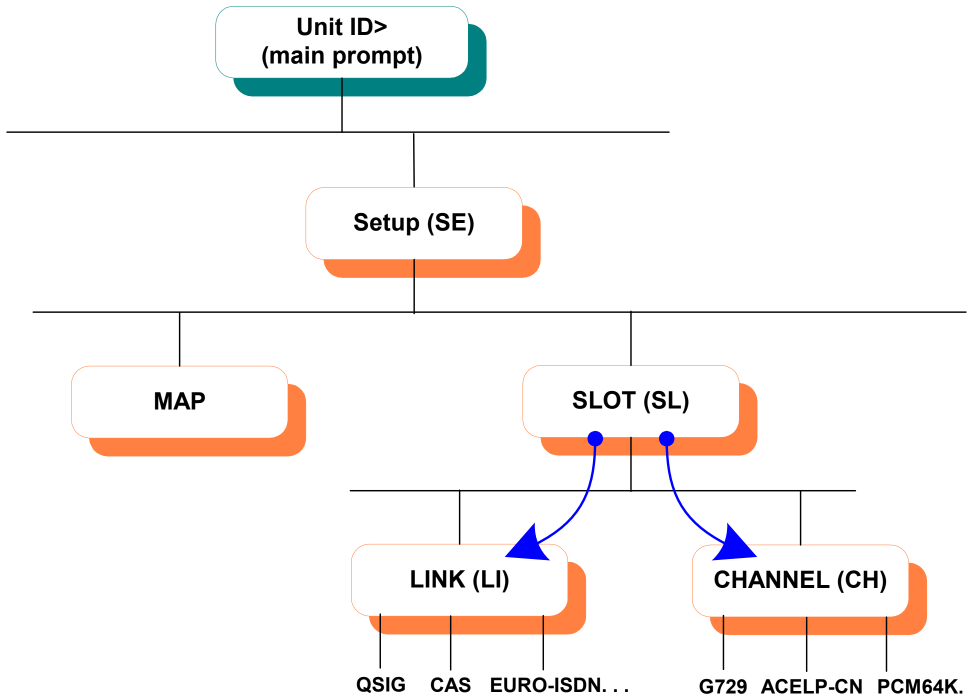

Figure 2-1: SETUP Command Paths in the CLI Tree for Digital Voice Support

2.2 Configuring a T1 Physical Port (LINK)

To define the physical port on a T1 interface card:

1. Enter the menu sequence: **SE** ↵ **SLOT**
2. Select the *Slot number*
3. Enter **LINK**
4. Set the *Status* to **ENABLE** to activate the physical link

NOTE: If the physical link is disabled, all voice channels associated with this port are disabled as well. You can continue with channel configuration, and then enable the link at a later time.

5. Set the *Signaling mode* to:
   - **NTT** or **KDD** for ISDN signaling in Japan
   - **4ESS**, **5ESS**, **DMS100** or **NI2** for ISDN signaling in North America (National ISDN)
   - **QSIG** for QSIG signaling
   - **ROB BIT** for T1 Robbed-bit signaling (can support an analog telephone or PBX)
   - **TRSP-ORIG** (originate) or **TRSP-ANSW** (answer) for transparent HDLC-based or PCM64K-based transport
   - **NONE** for support of an analog telephone or PBX, or a data connection. This is the default value. **It does *not* support a digital voice connection.**

NOTE: For further information on digital data connections, consult the *Digital Data* module of this document series.

6. Change the other digital link parameters from their default values, if desired.

SE/SLOT/#/LINK example: on a T1 interface card, for digital voice

```
SDM-9230>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ?
Item (LINK/CHANNEL,def:CHANNEL) ? LINK
PORT 100> Status (def:ENABLE) ?
PORT 100> Clock recovery (def:DISABLE) ?
PORT 100> Digital port clock source (def:INTERNAL) ?
```

```
PORT 100> Signaling mode (def:NONE) ? QSIG
PORT 100> QSIG master (def:NO) ?
PORT 100> Channel selection mode (def:PREFERRED) ?
PORT 100> Local number (def:) ?
PORT 100> Local subaddress (def:) ?
PORT 100> Calling number type of network  (def:000 (Unknown)) ?
PORT 100> Calling number numbering plan  (def:0000 (Unknown)) ?
PORT 100> Called number type of network  (def:000 (Unknown)) ?
PORT 100> Called number numbering plan  (def:0000 (Unknown)) ?
PORT 100> Pcm encoding law (def:MU-LAW) ?
PORT 100> Hunt Group Sorting (def:RRA) ?
PORT 100> Idle code (def:7F) ?
PORT 100> Zero suppression mode (def:B8ZS) ?
PORT 100> Gain limit (def:-30DB) ?
PORT 100> Framing mode (def:ESF) ?
PORT 100> Line Build Out (def:0-133FT) ?
PORT 100> Loopback (def:DISABLE) ?
```

NOTE: When the *Signaling mode* is set to **QSIG**, set *QSIG master* to **YES** if this side connects to equipment that is defined as slave. Otherwise, set it to **NO**. **The two sides of a QSIG connection must have opposite values for this parameter.**

SE/SLOT/#/ LINK example: on T1 interface card, for analog connection

```
SDM-9230>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ?
Item (LINK/CHANNEL,def:LINK) ? LINK
PORT 100> Status (def:DISABLE) ? ENABLE
PORT 100> Clock recovery (def:DISABLE) ?
PORT 100> Digital port clock source (def:INTERNAL) ?
PORT 100> Signaling mode (def:NONE) ? ROB BIT
PORT 100> Pcm encoding law (def:MU-LAW) ?
PORT 100> Idle code (def:7F) ?
PORT 100> Zero suppression mode (def:B8ZS) ?
PORT 100> Gain limit (def:-30DB) ?
PORT 100> Framing mode (def:ESF) ?
PORT 100> Line Build Out (def:0-133FT) ?
PORT 100> Loopback (def:DISABLE) ?
```

All **LINK** parameters on a T1 interface card are detailed in the appendix *SE/SLOT/#/ LINK Configuration Parameters* in the *Digital Data* module of this document series. *Hunt Group Sorting* is addressed in the chapter *Hunt Forwarding* of the *Advanced Voice Features* module.

2.3 Configuring an E1 Physical Port (LINK)

To define the physical port on an E1 interface card:

1. Enter the menu sequence: **SE ↵ SLOT**
2. Select the *Slot number*
3. Enter **LINK**
4. Set the *Status* to **ENABLE** to activate the physical link

NOTE: If the physical link is disabled, all voice channels associated with this port are disabled as well. You can continue with channel configuration, and then enable the link at a later time.

5. Set the *Signaling mode* to:
   - **CAS** for CAS signaling (can support an analog telephone or PBX)
   - **EURO-ISDN** for ISDN signaling in Europe
   - **QSIG** for QSIG signaling
   - **TRSP-ORIG** (originate) or **TRSP-ANSW** (answer) for transparent HDLC-based or PCM64K-based transport
   - **NONE** for support of an analog telephone or PBX, or a data connection. This is the default value. **It does *not* support a digital voice connection**.

NOTE: For further information on digital data connections, consult the *Digital Data* module of this document series.

6. Change the other digital link parameters from their default values, if desired.

SE/SLOT/#/ LINK example: on an E1 interface card, for digital voice

```
SDM-9230>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ? 2
Item (LINK/CHANNEL,def:CHANNEL) ? LINK
PORT 200> Status (def:ENABLE) ?
PORT 200> Clock recovery (def:DISABLE) ?
PORT 200> Digital port clock source (def:INTERNAL) ?
PORT 200> Signaling mode (def:NONE) ? EURO-ISDN
PORT 200> CCS side (def:USER) ?
PORT 200> Channel selection mode (def:PREFERRED) ?
```

```
PORT 200> Local number (def:) ?
PORT 200> Local subaddress (def:) ?
PORT 200> Calling number type of network  (def:000 (Unknown)) ?
PORT 200> Calling number numbering plan  (def:0000 (Unknown)) ?
PORT 200> Called number type of network  (def:000 (Unknown)) ?
PORT 200> Called number numbering plan  (def:0000 (Unknown)) ?
PORT 200> Pcm encoding law (def:A-LAW) ?
PORT 200> Hunt Group Sorting (def:RRA) ?
PORT 200> Idle code (def:7E) ?
PORT 200> Zero suppression mode (def:HDB3) ?
PORT 200> Gain limit (def:-12DB) ?
PORT 200> CRC4 mode (def:ENABLE) ?
PORT 200> International bit (def:ENABLE) ?
PORT 200> ETS 300 011 mode (def:DISABLE) ?
PORT 200> Generate ring back locally (def:DISABLE) ?
PORT 200> Loopback (def:DISABLE) ?
```

NOTE: When the *Signaling mode* is set to **EURO-ISDN**, set the *CCS side* (SNMP: *ifwanCcsSide*) to **USER** if this channel faces the network side of the connection. Set it to **NETWORK** when the channel faces the user side of the application. **The two sides of a CCS connection must have opposite values for this parameter.**

SE/SLOT/#/ LINK example: on E1 interface card, for analog connection

```
SDM-9230>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ? 2
Item (LINK/CHANNEL,def:CHANNEL) ? LINK
PORT 200> Status (def:DISABLE) ? ENABLE
PORT 200> Clock recovery (def:DISABLE) ?
PORT 200> Digital port clock source (def:INTERNAL) ?
PORT 200> Signaling mode (def:NONE) ? CAS
PORT 200> Pcm encoding law (def:A-LAW) ?
PORT 200> Idle code (def:7E) ?
PORT 200> Zero suppression mode (def:HDB3) ?
PORT 200> Gain limit (def:-12DB) ?
PORT 200> CRC4 mode (def:ENABLE) ?
PORT 200> International bit (def:ENABLE) ?
PORT 200> ETS 300 011 mode (def:DISABLE) ?
PORT 200> Loopback (def:DISABLE) ?
```

All **LINK** parameters on an E1 interface card are detailed in the appendix *SE/SLOT/#/ LINK Configuration Parameters* in the *Digital Data* module of this document series.

2.4 Configuring an ISDN-BRI S/T Physical Port (LINK)

To define a physical port on an ISDN-BRI S/T interface card:

1. Enter the menu sequence: **SE** ↵ **SLOT**
2. Select the *Slot number*
3. Select the *Port number*

   The port number indicator in subsequent parameters is **00** when *Port number* is set to **1**, or **50** when *Port number* is set to **2**. For example, **PORT 300** indicates the physical link on Slot 3, Port 1.
4. Enter **LINK**
5. Set the *Status* to **ENABLE** to activate the physical link

   If the physical link is disabled, all voice channels associated with this port are disabled as well. You can continue with channel configuration, and then enable the link at a later time.
6. Set the *Signaling mode* to:
   - **EURO-ISDN** for ISDN signaling in Europe
   - **INS-NET** or **KDD** for ISDN access in Japan
   - **NI1**, **NI2**, **5ESS** or **DMS100** for ISDN access in North America (National ISDN)
   - **QSIG** for QSIG signaling
   - **NONE** for a data connection. This is the default value. **It does *not* support a digital voice connection**.

   For further information on digital data connections, consult the *Digital Data* module of this document series.
7. Change the other digital link parameters from their default values, if desired.

SE/SLOT/#/ LINK example: on an ISDN-BRI S/T interface card

```
CHICAGO>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2,def:1) ? 2
Port number (1/2,def:1) ?
Item (LINK/CHANNEL,def:LINK) ?
PORT 200> Status (def:DISABLE) ? ENABLE
PORT 200> Clock recovery (def:DISABLE) ?
PORT 200> Digital port clock source (def:INTERNAL) ?
PORT 200> Signaling mode (def:NONE) ? KDD
PORT 200> CCS side (def:USER) ? NETWORK
PORT 200> Channel selection mode (def:PREFERRED) ?
PORT 200> Local number (def:) ? 6595090
PORT 200> Local subaddress (def:) ? 01
PORT 200> Calling number type of network (def:000 (Unknown)) ?
PORT 200> Calling number numbering plan (def:0000 (Unknown)) ?
```

```
PORT 200> Called number type of network (def:000 (Unknown)) ?
PORT 200> Called number numbering plan (def:0000 (Unknown)) ?
PORT 200> Terminal Endpoint Identifier (TEI) (def:AUTOMATIC) ?
PORT 200> Pcm encoding law (def:A-LAW) ?
PORT 200> Hunt Group Sorting (def:RRA) ?
PORT 200> Power Mode (def:OFF) ? PHANTOM
PORT 200> Generate ring back locally (def:DISABLE) ?
PORT 200> Loopback (def:DISABLE) ?
```

When the *Signaling mode* is set to **NI1**, **NI2**, **5ESS**, **DMS100** or **QSIG**, two additional parameters are presented to define the Service Profile Identifiers (SPIDs): *Local SPID 1* and *Local SPID 2*:

```
CHICAGO>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ? 3
Port number (1/2,def:1) ? 2
Item (LINK/CHANNEL,def:LINK) ? LINK
PORT #350> Status (def:DISABLE) ? ENABLE
PORT #350> Clock recovery (def:DISABLE) ?
PORT #350> Digital port clock source (def:INTERNAL) ?
PORT #350> Signaling mode (def:NONE) ? NI1
...
PORT #350> Local subaddress (def:) ?
PORT #350> Local SPID 1 (def:) ?
PORT #350> Local SPID 2 (def:) ?
PORT #350> Terminal Endpoint Identifier (TEI) (def:AUTOMATIC) ?
...
```

The *Power Mode* parameter, shown in the example on "SE/SLOT/#/LINK example: on an ISDN-BRI S/T interface card" on page 2-8, is listed **only when the ISDN-BRI S/T port is set to NT termination mode**. If the ISDN-BRI S/T port is set to TE termination mode, the power source is automatically disabled by the hardware.

To view the current NT/TE termination mode:

1. At the NetPerformer command prompt, enter **DS** ↵ **SLOT**
2. Select the *Slot number*.

   The *Interface* statistic will read either:

   - **BRI-NT**, for NT termination mode
   - **BRI-TE**, for TE termination mode

DS/SLOT example: with current NT/TE termination mode

```
CHICAGO>DS
DISPLAY STATES
Item (GLOBAL/PORT/PU/PVC/SLOT/SVC/VLAN,def:GLOBAL) ? SLOT
SLOT> Slot number (1/2/3/4/ALL,def:1) ? 3
SLOT 3 - PORT 1>
PORT #300> Protocol.............................ISDN BRI
PORT #300> State................................OUT OF SYNC
PORT #300> D-Channel state......................DOWN
PORT #300> Interface............................BRI-NT

SLOT 3 - PORT 2>
PORT #350> Protocol.............................ISDN BRI
PORT #350> State................................OUT OF SYNC
PORT #350> D-Channel state......................DOWN
PORT #350> Interface............................BRI-TE

Modem signals:  d(S)r  d(T)r  (D)cd  (R)ts  (C)ts  r(I)  (-)off
```

To change the current NT/TE termination mode for the physical ISDN-BRI S/T port, refer to the *Hardware Installation Guide* for your particular NetPerformer model.

2.5 Configuring the Digital Voice Channels (CHANNEL)

The digital channels of a T1/E1 or ISDN-BRI S/T interface can transport data or voice, depending on:

- The *Signaling mode* set on the physical link. Refer to “Configuring a T1 Physical Port (LINK)” on page 2-4 for T1 signaling modes, “Configuring an E1 Physical Port (LINK)” on page 2-6 for E1 signaling modes, and “Configuring an ISDN-BRI S/T Physical Port (LINK)” on page 2-8 for ISDN-BRI S/T signaling modes.
- The *Protocol* set on the **CHANNEL**. The protocols that are available depend on the *Signaling mode* set on the **LINK**, as explained below.
- For a CAS connection, the *Signaling type* required. Digital interfaces offer the following *Signaling type* selections:
  - T1 interface: **IMMEDIATE START**, **FXO**, **FXS**, **GND FXO**, **GND FXS**, **PLAR**, **WINK START**, **CUSTOM**
  - E1 interface: **IMMEDIATE START**, **R2**, **FXO**, **FXS**, **GND FXO**, **GND FXS**, **PLAR**, **WINK START**, **R2-CHINA**, **CUSTOM**

  Refer to “Channel Associated Signaling (CAS) on Digital Interface Cards” on page 1-15 for an explanation of analog connections that are supported by digital interfaces.

To define a digital voice channel:

NOTE: Digital voice channels are configured in the same way on all types of digital interfaces. Refer to Figure 2-1.

1. Enter the menu sequence: **SE** ↵ **SLOT**
2. Select the *Slot number*
3. For a dual port interface, select the *Port number*
4. Enter **CHANNEL**
5. Select the *Channel Number*, e.g. **101**, where the first digit indicates the slot or span on the NetPerformer chassis, and the last two digits indicate the channel
6. Set the *Protocol* to a voice protocol: **ACELP-CN**, **G723**, **G726 16K**, **G726 24K**, **G726 32K**, **G726 40K**, **G729**, **G729A, MELP** or **PCM64K**
7. Select the *Timeslot*

   Unlike a digital data channel which can contain multiple contiguous timeslots, a digital voice channel has only one timeslot.

NOTE: The *Timeslot* is set automatically on an ACELP-CN channel.

8. Change the other digital channel parameters from their default values, if desired. The parameters required for each of the available voice protocols are listed in the next section.

SE/SLOT/#/ CHANNEL example: on a dual-port interface

```
NP3>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/SS7/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ? 1
Port number (1/2,def:1) ? 1
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Channel Number (101-131/ALL,def:101) ? 101
VOICE 101> Protocol (def:OFF) ? ?
CHOICE: OFF            ACELP-CN       D&I            FR-NET
        FR-USER        G723           G726 16K       G726 24K
        G726 32K       G726 40K       G729           G729A
        HDLC           MELP           PASSTHRU       PASSTHRUOFR
        PCM64K         PPP            PVCR           SS7
        SS7MTP2        SS7MTP2 ISUP-A SS7 ISUP-A     TRANSPARENT

VOICE 101> Protocol
        (Default value:OFF, Current value:OFF) ? ACELP-CN
VOICE 101> Timeslot.................................1


VOICE 101> DSP packets per frame              1234
VOICE 101> 8K packetization selection (Y/N) ? YNNN
VOICE 101> DSP packets per frame              12345
VOICE 101> 6K packetization selection (Y/N) ? NNNNN
VOICE 101> Comfort noise level (def:10) ?
VOICE 101> Signaling type (def:NONE) ?
VOICE 101> Hoot & Holler application (def:NO) ?
VOICE 101> Local inbound voice level (db) (def:0) ?
VOICE 101> Local outbound voice level (db) (def:-3) ?
VOICE 101> Priority Level (0-10,def:0) ?
VOICE 101> Echo canceler (def:ENABLE) ?
VOICE 101> Double talk threshold (db) (def:6) ?
VOICE 101> Echo suppressor (def:ENABLE) ?
VOICE 101> Activation type (def:PREDEFINED) ?
VOICE 101> Link down busy (def:NO) ?
VOICE 101> TONE type: (def:DTMF) ?
VOICE 101> TONE regeneration: (0-255,def:1) ?
VOICE 101> TONE ON (ms) (30-1000,inc:10,def:100) ?
VOICE 101> TONE OFF (ms) (30-1000,inc:10,def:100) ?
VOICE 101> Pulse make/break ratio (30-50,inc:4,def:34) ?
VPORT 101> (UHFTONEDETECTION) UHF Radio tone detection
(def:DISABLE) ?
VOICE 101> Fax relay (def:FAX) ?
VOICE 101> Maximum fax rate (def:14400) ?
VOICE 101> ECM mode (def:DISABLE) ?
VOICE 101> Modem relay (def:NONE) ?
VOICE 101> Remote unit (def:NONE) ?
VOICE 101> Remote port number (1-65534,def:101) ?
VOICE 101> DTMF power ratio (5-100,def:5) ?
```

```
VOICE 101> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 101> Egress ANI operation mode (def:NONE) ?
VOICE 101> Egress CHANNEL ANI digits (def:) ?
VOICE 101> Ingress ANI operation mode (def:NONE) ?
VOICE 101> Ingress CHANNEL ANI digits (def:) ?
VOICE 101> Redundant channel (def:NO) ?
VOICE 101> V110 mode (def:DISABLE) ?
```

2.6 Voice Protocol Configuration

Any channel on a T1, E1 or ISDN-BRI S/T interface can be configured with the following voice protocols:

- ACELP-CN (see next section)
- G723 (see “G723 Protocol” on page 2-16)
- G726 16K, G726 24K, G726 32K and G726 40K (see “G726 16K, G726 24K, G726 32K and G726 40K Protocols” on page 2-17)
- G729 (see “G729 Protocol” on page 2-18)
- G729A (see “G729A Protocol” on page 2-19)
- MELP (see “MELP Protocol” on page 2-20)
- PCM64K (see “PCM64K Protocol” on page 2-21).

2.6.1 ACELP-CN Protocol

The following parameters are presented at the NetPerformer console for configuration of the ACELP-CN protocol:

SE/SLOT/#/ CHANNEL example: with ACELP-CN

```
NP3>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/SS7/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ?
Port number (1/2,def:1) ?
Item (LINK/CHANNEL,def:CHANNEL) ?
SLOT> Channel Number (101-131/ALL,def:103) ? 104
VOICE 104> Protocol (def:OFF) ? ACELP-CN
VOICE 104> Timeslot.................................4
VOICE 104> DSP packets per frame                1234
VOICE 104> 8K packetization selection (Y/N) ? YNNN
VOICE 104> DSP packets per frame                12345
VOICE 104> 6K packetization selection (Y/N) ? NNNNN
VOICE 104> Comfort noise level (def:10) ?
VOICE 104> Signaling type (def:NONE) ?
VOICE 104> Hoot & Holler application (def:NO) ?
VOICE 104> Local inbound voice level (db) (def:0) ?
VOICE 104> Local outbound voice level (db) (def:-3) ?
VOICE 104> Priority Level (0-10,def:0) ?
VOICE 104> Echo canceler (def:ENABLE) ?
VOICE 104> Double talk threshold (db) (def:6) ?
VOICE 104> Echo suppressor (def:ENABLE) ?
VOICE 104> Activation type (def:PREDEFINED) ? AUTODIAL
VOICE 104> Speed dial number (def:NONE) ? 435
VOICE 104> Link down busy (def:NO) ?
VOICE 104> TONE type: (def:DTMF) ?
VOICE 104> TONE regeneration: (0-255,def:1) ?
VOICE 104> TONE ON (ms) (30-1000,inc:10,def:100) ?
```

```
VOICE 104> TONE OFF (ms) (30-1000,inc:10,def:100) ?
VOICE 104> Pulse make/break ratio (30-50,inc:4,def:34) ?
VPORT 104> (UHFTONEDETECTION) UHF Radio tone detection
(def:DISABLE) ?
VOICE 104> Fax relay (def:FAX) ?
VOICE 104> Maximum fax rate (def:14400) ?
VOICE 104> ECM mode (def:DISABLE) ?
VOICE 104> Modem relay (def:NONE) ?
VOICE 104> Hunt Group active (def:NONE) ?
VOICE 104> Port extension number (def:104) ?
VOICE 104> Fwd digits (def:NONE) ?
VOICE 104> DTMF power ratio (5-100,def:5) ?
VOICE 104> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 104> Egress ANI operation mode (def:NONE) ?
VOICE 104> Egress CHANNEL ANI digits (def:) ?
VOICE 104> Ingress ANI operation mode (def:NONE) ?
VOICE 104> Ingress CHANNEL ANI digits (def:) ?
VOICE 104> Redundant channel (def:NO) ?
VOICE 104> V110 mode (def:DISABLE) ?
```

All voice **CHANNEL** parameters are detailed in the appendix *SE/SLOT/#/CHANNEL Configuration Parameters* in the *Analog Voice* module of this document series.

NOTE: When the *Signaling mode* on the **LINK** is set to **CAS** or **ROB BIT**, the *Signaling type* on the channel can be set to **IMMEDIATE START**, **FXO**, **FXS**, **GND FXO**, **GND FXS**, **PLAR**, **WINK START** or **CUSTOM** (default: **IMMEDIATE START**).

If the *Signaling type* is set to **FXO** or **GND FXO**, additional parameters are presented for definition of the *FXO seizure delay* and *FXO timeout*:

```
NP3>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/SS7/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ? 2
Port number (1/2,def:1) ?
Item (LINK/CHANNEL,def:CHANNEL) ?
SLOT> Channel Number (201-231/ALL,def:201) ? 201
VOICE 201> Protocol (def:OFF) ? ACELP-CN
VOICE 201> Timeslot.................................1
VOICE 201> DSP packets per frame              1234
VOICE 201> 8K packetization selection (Y/N) ? YNNN
VOICE 201> DSP packets per frame              12345
VOICE 201> 6K packetization selection (Y/N) ? NNNNN
VOICE 201> Comfort noise level (def:10) ?
VOICE 201> Signaling type (def:IMMEDIATE START) ? FXO
VOICE 201> FXO seizure delay (def:DISABLE) ?
VOICE 201> FXO timeout (s) (6-99,def:30) ?
```

...

2.6.2 G723 Protocol

The following parameters are presented at the NetPerformer console for the G723 protocol:

SE/SLOT/#/ CHANNEL example: with G723

```
NP3>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/SS7/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ?
Port number (1/2,def:1) ?
Item (LINK/CHANNEL,def:CHANNEL) ?
SLOT> Channel Number (101-131/ALL,def:105) ? 106
VOICE 106> Protocol (def:OFF) ? G723
VOICE 106> Timeslot (def:6) ?
VOICE 106> DSP packets per frame                  123
VOICE 106> 6.4K packetization selection (Y/N) ? YNN
VOICE 106> DSP packets per frame                  123
VOICE 106> 5.3K packetization selection (Y/N) ? NNN
VOICE 106> Comfort noise level (def:OFF) ?
VOICE 106> Signaling type (def:NONE) ?
VOICE 106> Hoot & Holler application (def:NO) ?
VOICE 106> Local inbound voice level (db) (def:0) ?
VOICE 106> Local outbound voice level (db) (def:-3) ?
VOICE 106> Priority Level (0-10,def:0) ?
VOICE 106> Echo canceler (def:ENABLE) ?
VOICE 106> Double talk threshold (db) (def:6) ?
VOICE 106> Echo suppressor (def:ENABLE) ?
VOICE 106> Activation type (def:PREDEFINED) ?
VOICE 106> Link down busy (def:NO) ?
VOICE 106> TONE type: (def:DTMF) ?
VOICE 106> TONE regeneration: (0-255,def:1) ?
VOICE 106> TONE ON (ms) (30-1000,inc:10,def:100) ?
VOICE 106> TONE OFF (ms) (30-1000,inc:10,def:100) ?
VOICE 106> Pulse make/break ratio (30-50,inc:4,def:34) ?
VPORT 106> (UHFTONEDETECTION) UHF Radio tone detection
(def:DISABLE) ?
VOICE 106> Fax relay (def:FAX) ?
VOICE 106> Maximum fax rate (def:14400) ?
VOICE 106> ECM mode (def:DISABLE) ?
VOICE 106> Modem relay (def:NONE) ?
VOICE 106> Remote unit (def:NONE) ?
VOICE 106> Remote port number (1-65534,def:106) ?
VOICE 106> DTMF power ratio (5-100,def:5) ?
VOICE 106> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 106> Egress ANI operation mode (def:NONE) ?
VOICE 106> Egress CHANNEL ANI digits (def:) ?
VOICE 106> Ingress ANI operation mode (def:NONE) ?
VOICE 106> Ingress CHANNEL ANI digits (def:) ?
VOICE 106> Redundant channel (def:NO) ?
```

```
VOICE 106> V110 mode (def:DISABLE) ?
```

All voice **CHANNEL** parameters are detailed in the appendix *SE/SLOT/#/CHANNEL Configuration Parameters* in the *Analog Voice* module of this document series.

NOTE: The buffering scheme is based on 6.4K and 5.3K packetization, and provides fallback levels up to 3.

The *Signaling type* can be set to **IMMEDIATE START**, **R2**, **FXO**, **FXS**, **GND FXO**, **GND FXS**, **PLAR**, **WINK START**, **R2-CHINA** or **CUSTOM**.

This parameter is available only when the *Signaling mode* on the **LINK** is set to **CAS** or **ROB BIT**.

2.6.3 G726 16K, G726 24K, G726 32K and G726 40K Protocols

The G726 16K, G726 24K, G726 32K and G726 40K protocols are all configured in the same way, except for the *Protocol* setting. The following parameters are presented at the NetPerformer console:

SE/SLOT/#/ CHANNEL example: with G726 16K

```
NP3>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/SS7/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,def:1) ?
Port number (1/2,def:1) ?
Item (LINK/CHANNEL,def:CHANNEL) ?
SLOT> Channel Number (101-131/ALL,def:106) ? 107
VOICE 107> Protocol (def:OFF) ? G726 16K
VOICE 107> Timeslot (def:7) ?
VOICE 107> Signaling type (def:NONE) ?
VOICE 107> Hoot & Holler application (def:NO) ?
VOICE 107> Silence suppression level (1-5,def:1) ?
VOICE 107> Local inbound voice level (db) (def:0) ?
VOICE 107> Local outbound voice level (db) (def:-3) ?
VOICE 107> Priority Level (0-10,def:0) ?
VOICE 107> Echo canceler (def:ENABLE) ?
VOICE 107> Double talk threshold (db) (def:6) ?
VOICE 107> Echo suppressor (def:ENABLE) ?
VOICE 107> Activation type (def:PREDEFINED) ?
VOICE 107> Link down busy (def:NO) ?
VOICE 107> TONE type: (def:DTMF) ?
VOICE 107> TONE regeneration: (0-255,def:1) ?
VOICE 107> TONE ON (ms) (30-1000,inc:10,def:100) ?
VOICE 107> TONE OFF (ms) (30-1000,inc:10,def:100) ?
VOICE 107> Pulse make/break ratio (30-50,inc:4,def:34) ?
VPORT 107> (UHFTONEDETECTION) UHF Radio tone detection
```

```
(def:DISABLE) ?
VOICE 107> Fax relay (def:FAX) ?
VOICE 107> Maximum fax rate (def:14400) ?
VOICE 107> ECM mode (def:DISABLE) ?
VOICE 107> Modem relay (def:NONE) ?
VOICE 107> Remote unit (def:NONE) ?
VOICE 107> Remote port number (1-65534,def:107) ?
VOICE 107> DTMF power ratio (5-100,def:5) ?
VOICE 107> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 107> Egress ANI operation mode (def:NONE) ?
VOICE 107> Egress CHANNEL ANI digits (def:) ?
VOICE 107> Ingress ANI operation mode (def:NONE) ?
VOICE 107> Ingress CHANNEL ANI digits (def:) ?
VOICE 107> Redundant channel (def:NO) ?
VOICE 107> V110 mode (def:DISABLE) ?
```

All voice **CHANNEL** parameters are detailed in the appendix *SE/SLOT/#/CHANNEL Configuration Parameters* in the *Analog Voice* module of this document series.

NOTE: There is no buffering (*packetization selection*) or *Comfort Noise* configuration on a G726 voice channel.

The *Silence suppression level* can be adjusted on a G726 voice channel. This parameter is also requested when the protocol is set to **G729**, **G729A** or **PCM64K**.

2.6.4 G729 Protocol

The following parameters are presented at the NetPerformer console for the G729 protocol:

SE/SLOT/#/ CHANNEL example: with G729

```
CHICAGO>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3/4,def:1) ?
Item (LINK/CHANNEL,def:CHANNEL) ?
SLOT> Channel Number (101-131/ALL,def:111) ? 112
PORT 112> Protocol (def:OFF) ? G729
PORT 112> Timeslot (1-31,def:12) ?
VOICE 112> DSP packets per frame           12345678
VOICE 112> Packetization selection (Y/N) ? YNNNNNNN
VOICE 112> Signaling type (def:IMMEDIATE START) ?
VOICE 112> Silence suppression level (1-5,def:1) ?
VOICE 112> Local inbound voice level (db) (def:0) ?
VOICE 112> Local outbound voice level (db) (def:-3) ?
VOICE 112> Priority Level (0-10,def:0) ?
```

```
VOICE 112> Echo canceler (def:ENABLE) ?
VOICE 112> Double talk threshold (db) (def:6) ?
VOICE 112> Country settings (def:USA) ?
VOICE 112> Activation type (def:PREDEFINED) ?
VOICE 112> Link down busy (def:NO) ?
VOICE 112> TONE type: (def:DTMF) ?
VOICE 112> TONE regeneration: (0-255,def:255) ?
VOICE 112> TONE ON (ms) (30-1000,inc:10,def:100) ?
VOICE 112> TONE OFF (ms) (30-1000,inc:10,def:100) ?
VOICE 112> Pulse make/break ratio (30-50,inc:4,def:34) ?
VOICE 112> Fax relay (def:FAX) ?
VOICE 112> Maximum fax rate (def:14400) ?
VOICE 112> Modem relay (def:NONE) ?
VOICE 112> DTMF power ratio (5-100,def:5) ?
VOICE 112> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 112> Remote unit (def:NONE) ?
VOICE 112> Remote port number (1-65534,def:112) ?
VOICE 112> Redundant channel (def:NO) ?
VOICE 112> Egress ANI operation mode (def:NONE) ?
VOICE 112> Egress CHANNEL ANI digits (def:) ?
VOICE 112> Ingress ANI operation mode (def:NONE) ?
VOICE 112> Ingress CHANNEL ANI digits (def:) ?
```

All voice **CHANNEL** parameters are detailed in the appendix *SE/SLOT/#/CHANNEL Configuration Parameters* in the *Analog Voice* module of this document series.

NOTE: The buffering scheme is based on 8K packetization, and provides fallback levels up to 8.

The *Signaling type* can be set to **IMMEDIATE START**, **R2**, **FXO**, **FXS**, **GND FXO**, **GND FXS**, **PLAR**, **WINK START**, **R2-CHINA** or **CUSTOM**.

This parameter is available only when the *Signaling mode* on the **LINK** is set to **CAS** or **ROB BIT**.

2.6.5 G729A Protocol

The following parameters are presented at the NetPerformer console for the G729A protocol:

SE/SLOT/#/ CHANNEL example: with G729A

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3/4,def:3) ?
Item (LINK/CHANNEL,def:CHANNEL) ?
SLOT> Channel Number (301-324/ALL,def:306) ? 307
```

```
PORT 307> Protocol (def:OFF) ? G729A
PORT 307> Timeslot (1-31,def:12) ?
VOICE 307> DSP packets per frame           12345678
VOICE 307> Packetization selection (Y/N) ? YNNNNNNN
VOICE 307> Signaling type (def:IMMEDIATE START) ?
VOICE 307> Silence suppression level (1-5,def:1) ?
VOICE 307> Local inbound voice level (db) (def:0) ?
VOICE 307> Local outbound voice level (db) (def:-3) ?
VOICE 307> Priority Level (0-10,def:0) ?
VOICE 307> Echo canceler (def:ENABLE) ?
VOICE 307> Double talk threshold (db) (def:6) ?
VOICE 307> Country settings (def:USA) ?
VOICE 307> Activation type (def:PREDEFINED) ?
VOICE 307> Link down busy (def:NO) ?
VOICE 307> TONE type: (def:DTMF) ?
VOICE 307> TONE regeneration: (0-255,def:255) ?
VOICE 307> TONE ON (ms) (30-1000,inc:10,def:100) ?
VOICE 307> TONE OFF (ms) (30-1000,inc:10,def:100) ?
VOICE 307> Pulse make/break ratio (30-50,inc:4,def:34) ?
VOICE 307> Fax relay (def:FAX) ?
VOICE 307> Maximum fax rate (def:14400) ?
VOICE 307> Modem relay (def:NONE) ?
VOICE 307> DTMF power ratio (5-100,def:5) ?
VOICE 307> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 307> Remote unit (def:NONE) ?
VOICE 307> Remote port number (1-65534,def:112) ?
VOICE 307> Redundant channel (def:NO) ?
VOICE 307> Egress ANI operation mode (def:NONE) ?
VOICE 307> Egress CHANNEL ANI digits (def:) ?
VOICE 307> Ingress ANI operation mode (def:NONE) ?
VOICE 307> Ingress CHANNEL ANI digits (def:) ?
```

All voice **CHANNEL** parameters are detailed in the appendix *SE/SLOT/#/CHANNEL Configuration Parameters* in the *Analog Voice* module of this document series.

2.6.6 MELP Protocol

The following parameters are presented at the NetPerformer console for the MELP protocol:

SE/SLOT/#/ CHANNEL example: with MELP

```
NP3>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/SS7/USER/VLAN,
def:SLOT) ?
SLOT> Slot number (1/2/3,def:3) ?
Port number (1/2,def:2) ? 1
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Channel Number (301-302/ALL,def:301) ?
PORT 301> Protocol (def:OFF) ? MELP
VOICE 301> Timeslot (def:1) ?
VOICE 301> DSP packets per frame                12345678
VOICE 301> 2.4K packetization selection (Y/N) ? YNNNNNNN
```

```
VOICE 301> Signaling type...........................DEFAULT
VOICE 301> Hoot & Holler application (def:NO) ?
VOICE 301> Silence suppression level (1-5,def:1) ?
VOICE 301> Local inbound voice level (db) (def:0) ?
VOICE 301> Local outbound voice level (db) (def:-3) ?
VOICE 301> Priority Level (0-10,def:0) ?
VOICE 301> Echo canceller (def:ENABLE) ?
VOICE 301> Double talk threshold (db) (def:6) ?
VOICE 301> Echo suppressor (def:ENABLE) ?
VOICE 301> Activation type (def:PREDEFINED) ?
VOICE 301> Link down busy (def:NO) ?
VOICE 301> TONE type: (def:DTMF) ?
VOICE 301> TONE regeneration: (0-255,def:1) ?
VOICE 301> TONE ON (ms) (30-1000,inc:10,def:100) ?
VOICE 301> TONE OFF (ms) (30-1000,inc:10,def:100) ?
VOICE 301> Pulse make/break ratio (30-50,inc:4,def:34) ?
VPORT 301> (UHFTONEDETECTION) UHF Radio tone detection
(def:DISABLE) ?
VOICE 301> Fax relay (def:FAX) ?
VOICE 301> Maximum fax rate (def:14400) ?
VOICE 301> ECM mode (def:DISABLE) ?
VOICE 301> Modem relay (def:NONE) ?
VOICE 301> Remote unit (def:NONE) ?
VOICE 301> Remote port number (1-65534,def:301) ?
VOICE 301> DTMF power ratio (5-100,def:5) ?
VOICE 301> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 301> Egress ANI operation mode (def:NONE) ?
VOICE 301> Egress CHANNEL ANI digits (def:) ?
VOICE 301> Ingress ANI operation mode (def:NONE) ?
VOICE 301> Ingress CHANNEL ANI digits (def:) ?
VOICE 301> Redundant channel (def:NO) ?
VOICE 301> V110 mode (def:DISABLE) ?
```

All voice **CHANNEL** parameters are detailed in the appendix *SE/SLOT/#/CHANNEL Configuration Parameters* in the *Analog Voice* module of this document series.

2.6.7 PCM64K Protocol

The following parameters are presented at the NetPerformer console for the PCM64K protocol:

SE/SLOT/#/CHANNEL example: with PCM64K

```
CHICAGO>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3/4,def:3) ?
Item (LINK/CHANNEL,def:CHANNEL) ?
SLOT> Channel Number (301-324/ALL,def:305) ? 306
PORT 306> Protocol (def:OFF) ? PCM64K
PORT 306> Timeslot (1-24,def:6) ?
VOICE 306> Signaling type (def:IMMEDIATE START) ? FXO
VOICE 306> FXO seizure delay (def:DISABLE) ?
VOICE 306> FXO timeout (s) (6-99,def:30) ?
```

```
VOICE 306> Silence suppression level (1-5,def:1) ?
VOICE 306> Local inbound voice level (db) (def:0) ?
VOICE 306> Local outbound voice level (db) (def:-3) ?
VOICE 306> Priority Level (0-10,def:0) ?
VOICE 306> Echo canceler (def:ENABLE) ?
VOICE 306> Double talk threshold (db) (def:6) ?
VOICE 306> Country settings (def:USA) ?
VOICE 306> Pulse frequency (pps) (def:10) ?
VOICE 306> Activation type (def:PREDEFINED) ?
VOICE 306> Link down busy (def:NO) ?
VOICE 306> TONE type: (def:DTMF) ? ?
VOICE 306> TONE type: (DTMF/MF,def:DTMF) ?
VOICE 306> TONE regeneration: (0-255,def:255) ?
VOICE 306> TONE ON (ms) (30-1000,inc:10,def:100) ?
VOICE 306> TONE OFF (ms) (30-1000,inc:10,def:100) ?
VOICE 306> Pulse make/break ratio (30-50,inc:4,def:34) ?
VOICE 306> Fax relay (def:FAX) ?
VOICE 306> Maximum fax rate (def:14400) ?
VOICE 306> Modem relay (def:NONE) ?
VOICE 306> Remote unit (def:NONE) ?
VOICE 306> Remote port number (1-899,def:306) ?
VOICE 306> Redundant channel (def:NO) ?
VOICE 306> Egress ANI operation mode (def:NONE) ?
VOICE 306> Egress CHANNEL ANI digits (def:) ?
VOICE 306> Ingress ANI operation mode (def:NONE) ?
VOICE 306> Ingress CHANNEL ANI digits (def:) ?
```

All voice **CHANNEL** parameters are detailed in the appendix *SE/SLOT/#/CHANNEL Configuration Parameters* in the *Analog Voice* module of this document series.

NOTE: The *Signaling type* can be set to **IMMEDIATE START**, **FXO**, **FXS**, **GND FXO**, **GND FXS**, **PLAR**, **WINK START** or **CUSTOM**.

This parameter is available only when the *Signaling mode* on the **LINK** is set to **CAS** or **ROB BIT**.

2.7 Push to Talk over E1

NetPerformer v10.4.2 provides transparent open channel Push-to-Talk (PTT) signaling transport (off-hook/on-hook transitions) through E1 as well as E&M ports. Prior to v10.4.2, this ability to provide Push-to-Talk transparent signaling across two NetPerformer units was available on the E&M voice ports only.

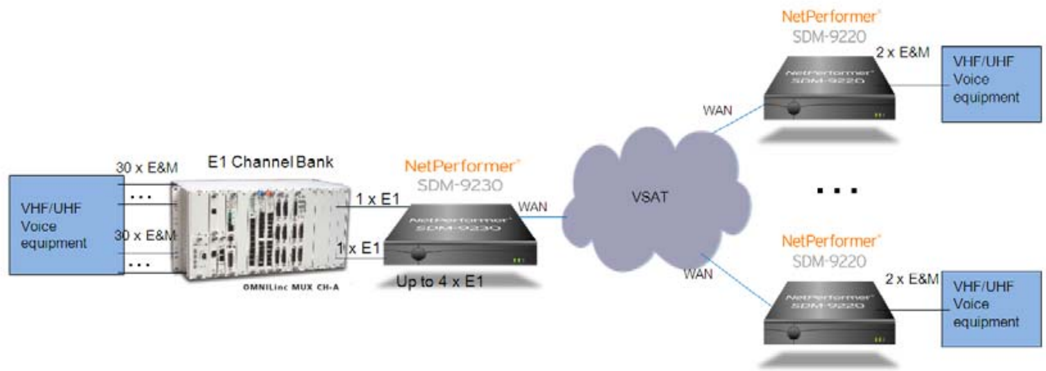

Figure 2-1: Push to Talk over E1

2.7.1 Parameters on E1 voice channels for Push-to-Talk Support

Console	SNMP	Text-based Config
Slot channel x / Push to Talk application	AnalogEmPuskTalk	[ifvce #] AnalogEmPuskTalk

Description: Push to talk application mode.

Values: DISABLE, PTT CONTROL, PTT ANSWER; default DISABLE

For more details about NetPerformer Push-to-Talk support, refer to the SDM-9220/9230 Hardware Installation Guide (Doc. # 1588) in the following sections:

- 9.3 - About Push To Talk (PTT)
- 9.4 - Configuring Push To Talk

3

Modem Passthru

3.1 About Modem Passthru

The *Modem Passthru* function allows a modem connection to be established without using compression, echo cancelling or any other DSP processing of the traffic stream. This simplifies the traffic over a PCM64K connection, permitting higher modem connection speeds.

NOTE: **Modem Passthru is available on the NetPerformer base product as well as any NetPerformer product installed with the SIP VoIP licensed software option.**

The characteristics of this feature are:

- Modem signal is sampled using the PCM64K codec algorithm
- No DSP echo cancellation
- No DSP speed control
- No DSP compression or decompression of the traffic.

Modem Passthru behaves differently on the NetPerformer base product (see next section) versus the NetPerformer equipped with the SIP VoIP licensed software option (see "Modem Passthru on the NetPerformer with SIP VoIP License" on page 3-7).

3.2 Modem Passthru on the NetPerformer Base Product

3.2.1 Call Setup Procedure

On the NetPerformer base product (SIP VoIP software license not activated), a Modem Passthru application starts with a normal voice call setup procedure.

- Once the voice path is fully established, a modem tone is sent from the calling modem to the called modem
- The called NetPerformer (with DSP application running) recognizes the modem tone and switches its transmitted packets over the WAN to a new packet ID. This new ID identifies the traffic as Modem Passthru traffic
- The DSP on the called NetPerformer then switches itself into **PCM64K** voice protocol, disables the echo canceller and indicates to the Signaling Engine that it is in Modem Passthru mode:
  - The DSP adjusts automatically, and there is no interaction with the host
  - **PCM64K** offers the highest sampling rate possible, including all modem speeds that may be required. **It is not necessary to specify the maximum modem speed in PASSTHRU mode.**
- The Modem Passthru mode indication is then sent to the host in an **INFO** message
- The calling NetPerformer (with DSP application running) detects the new packet ID
- The DSP on the calling NetPerformer switches into **PCM64K** voice protocol and disable the echo canceller
- At this point, the Modem Passthru connection is fully operational in both directions.

3.2.2 Configuring a NetPerformer Base Product for Modem Passthru

Configure Modem Passthru using the **SETUP/SLOT/CHANNEL** submenu.

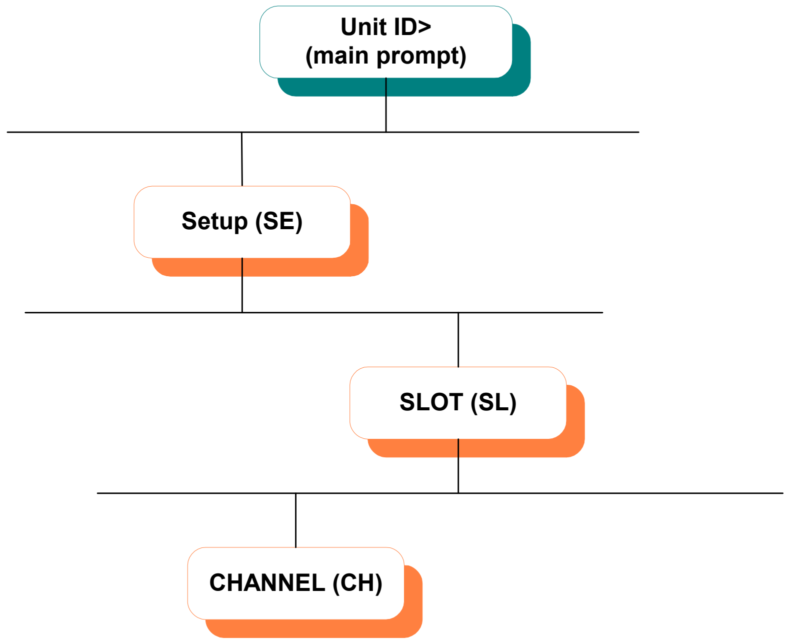

Figure 3-1: SETUP/SLOT/CHANNEL Path on the CLI Tree

NOTE: The following procedure assumes that you have already configured the physical interface using the **SE** ↵ **SLOT** ↵ **LINK** menu sequence. For details, consult the *NetPerformer User Guide*.

To configure Modem Passthru on the NetPerformer base product (SIP VoIP not installed):

1. At the NetPerformer command line prompt, enter the menu sequence: **SE** ↵ **SLOT**
2. Enter the *Slot number*
3. Enter **CHANNEL**
4. Enter the *Port number* to select the channel
5. Set the *Protocol* to a voice algorithm, e.g. **G729** or **PCM64K**
6. Set the *Modem relay* parameter to **PASSTHRU**
7. Change the other parameters from their default values, if desired.

SE/SLOT/ CHANNEL example: on NetPerformer base product

```
CHICAGO>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3/4,def:3) ? 2
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Port number (1-2,def:1) ? 1
VOICE 201> Protocol (def:OFF) ? G729
VOICE 201> DSP packets per frame           12345678
VOICE 201> Packetization selection (Y/N) ? YNNNNNNN
VOICE 201> Silence suppression level (1-5,def:1) ?
VOICE 201> Local inbound voice level (db) (def:0) ?
VOICE 201> Local outbound voice level (db) (def:-3) ?
VOICE 201> Priority Level (0-10,def:0) ?
VOICE 201> Echo canceler (def:ENABLE) ?
VOICE 201> Double talk threshold (db) (def:6) ?
VOICE 201> Country settings (def:USA) ?
VOICE 201> Pulse frequency (pps) (def:10) ?
VOICE 201> Activation type (def:PREDEFINED) ?
VOICE 201> Link down busy (def:NO) ?
VOICE 201> TONE type: (def:DTMF) ?
VOICE 201> TONE regeneration: (0-255,def:1) ?
VOICE 201> TONE ON (msec) (30-1000,inc:10,def:100) ?
VOICE 201> TONE OFF (msec) (30-1000,inc:10,def:100) ?
VOICE 201> Pulse make/break ratio (30-50,inc:4,def:34) ?
VOICE 201> Fax relay (def:NONE) ?
VOICE 201> Modem relay (def:NONE) ? PASSTHRU
VOICE 201> Remote unit (def:NONE) ?
VOICE 201> Remote port number (1-65534,def:201) ?
VOICE 107> Redundant channel (def:NO) ?
VOICE 107> Egress ANI operation mode (def:NONE) ?
VOICE 107> Egress CHANNEL ANI digits (def:) ?
VOICE 107> Ingress ANI operation mode (def:NONE) ?
VOICE 107> Ingress CHANNEL ANI digits (def:) ?
```

3.2.3 Modem relay

Console	SNMP	Text-based Config
Modem relay	ifvceModemRelay	[ifvce#] ModemRelay

Enables or disables handling of modem calls on the voice channel using either Modem Relay or Modem Passthru, as follows:

- **NONE:** Traffic across this channel is treated as regular voice traffic. Modem connections are not detected. Both Modem Relay and Modem Passthru are disabled.
- **MODEM:** *(available on the NetPerformer base product only)* Activates Modem Relay, the modem spoofing application available in NetPerformer versions prior to V10.1.

NOTE: If this value is selected, the *Maximum modem rate* parameter appears at the console for setting the maximum rate permitted for Modem Relay (SNMP: *ifvceMaxModemRate*).

- **PASSTHRU:** Activates Modem Passthru. No other parameters are required in this case.

Values: NONE, MODEM, PASSTHRU

Default: NONE

3.3 Modem Passthru on the NetPerformer with SIP VoIP License

3.3.1 Call Setup Procedure

On a NetPerformer installed with the SIP VoIP software license, the Modem Passthru application starts with a normal voice call setup procedure.

- Once the voice path is fully established, a modem tone is sent from the calling modem to the called modem
- The called NetPerformer (with DSP application running) recognizes the modem tone
- The DSP on the called NetPerformer then switches itself into **PCM64K** voice protocol, disables the echo canceller and indicates to the Signaling Engine that it is in Modem Passthru mode
- The called NetPerformer sends a connection modification in a **SIP INVITE** message to the calling NetPerformer. This **SIP INVITE** contains a new media descriptor to start the Modem Passthru connection
- Once the new SIP connection has been accepted by the calling NetPerformer, it sends back a **SIP 200 OK** message to confirm the connection
- At this point, the Modem Passthru connection is fully operational in both directions.

3.3.2 Modem Passthru and SIP Voice Codec Negotiation

For Modem Passthru on a NetPerformer installed with the SIP VoIP licensed option, **only the G711 codec is offered during codec negotiation.**

- You can set the *Protocol* parameter on the voice channel to any voice algorithm
- The change to G711 (PCM64K) will be carried out automatically and transparently between the two NetPerformer units participating in the negotiation process.

An attribute has been added to the NetPerformer *general purpose media descriptor* that specifies that the new media is Modem Passthru traffic.

When the NetPerformer detects that a Modem Passthru session has been requested from a SIP device:

- It offers a SIP **INVITE** that lists the G711 codec (including the specific G711 law) as the preferred codec choice
- The called unit determines whether it can accept this codec by comparing it with its preferred codec (the *Protocol* and *Passthru Codec law* configured on the voice **CHANNEL**) and the codecs listed in its Codec Negotiation Table
- The called unit will negotiate the codec, if required, and load the G711 codec on the voice channel.

TIPS: If you intend to activate Modem Passthru using codec negotiation:

- Ensure that all codec negotiation parameters (**SE/SIP/CODEC NEGO**) are set to **YES**. For further information, refer to the chapter *Codec Negotiation* in the *Voice over IP (VoIP) Option* module of this document series.

NOTE: Modem Passthru using codec negotiation is useful if you want to offer the G711 codec with both a-law and µ-law.

- If you are following the negotiation process from the console, ensure that it is operating at the correct baud rate (19,200 Kbps).

3.3.3 Configuring a SIP-enabled NetPerformer for Modem Passthru

Configure Modem Passthru using the **SETUP/SLOT/CHANNEL** submenu (see Figure 3-1).

NOTE: The following procedure assumes that you have already configured the physical interface using the **SE** ↵ **SLOT** ↵ **LINK** menu sequence. For details, consult the *NetPerformer User Guide*.

To configure Modem Passthru on the NetPerformer base product (SIP VoIP not installed):

1. At the NetPerformer command line prompt, enter the menu sequence: **SE** ↵ **SLOT**
2. Enter the *Slot number*
3. Enter **CHANNEL**
4. Enter the *Port number* to select the channel
5. Set the *Protocol* to any voice algorithm
6. Set the *Modem relay* parameter to **PASSTHRU**
7. Set the *Passthru Codec law* parameter to the correct codec law for your application: **G711_ULAW** or **G711_ALAW**
8. Change the other parameters from their default values, if desired.

SE/SLOT/ CHANNEL example: with Modem PASSTHRU on NetPerformer with SIP VoIP licensed option

```
PHOENIX>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/IP/IPX/MAP/
PHONE/PORT/
PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SIP/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1,def:1) ?
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Port number (1-4/ALL,def:1) ?
VOICE 101> Protocol (def:OFF) ? G723
VOICE 101> DSP packets per frame                123
VOICE 101> 6.4K packetization selection (Y/N) ? YNN
VOICE 101> DSP packets per frame                123
VOICE 101> 5.3K packetization selection (Y/N) ? NYN
VOICE 101> Comfort noise level (def:OFF) ?
VOICE 101> Local inbound voice level (db) (def:0) ?
VOICE 101> Local outbound voice level (db) (def:-3) ?
VOICE 101> Echo canceler (def:ENABLE) ?
VOICE 101> Double talk threshold (db) (def:6) ?
VOICE 101> Country settings (def:USA) ?
VOICE 101> Pulse frequency (pps) (def:10) ?
VOICE 101> Activation type (def:PREDEFINED) ? SWITCHED
VOICE 101> TONE type: (def:DTMF) ?
VOICE 101> TONE regeneration: (0-255,def:1) ?
VOICE 101> TONE ON (msec) (30-1000,inc:10,def:100) ?
VOICE 101> TONE OFF (msec) (30-1000,inc:10,def:100) ?
VOICE 101> Pulse make/break ratio (30-50,inc:4,def:34) ?
VOICE 101> Fax relay (def:T.38) ?
VOICE 101> Maximum fax rate (def:14400) ?
VOICE 101> Fax redundancy level (0-3,def:3) ?
VOICE 101> Modem relay (def:NONE) ? PASSTHRU
VOICE 101> Passthru Codec law (def:G711_ULAW) ? G711_ALAW
VOICE 101> Fwd digits (def:ALL) ?
VOICE 101> Fwd type (def:TONE) ?
VOICE 101> Fwd delay (msec) (0-10000,inc:250,def:0) ?
VOICE 101> DTMF power ratio (5-100,def:5) ?
VOICE 101> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 101> Redundant channel (def:NO) ?
VOICE 101> Egress ANI operation mode (def:NONE) ?
VOICE 101> Egress CHANNEL ANI digits (def:) ?
VOICE 101> Ingress ANI operation mode (def:NONE) ?
VOICE 101> Ingress CHANNEL ANI digits (def:) ?
VOICE 101> Caller ID (ANI) transmission protocol (def:OFF) ?
VOICE 101> Billing signals (def:DISABLE) ?
```

3.3.4 Modem relay

Console	SNMP	Text-based Config
Modem relay	ifvceSipModemRelay	[ifvce#] SipModemRelay

Enables or disables handling of Modem Passthru calls over this channel, as follows:

- **NONE:** Traffic across this channel is treated as regular voice traffic. Modem connections are not detected. Both Modem Relay and Modem Passthru are disabled.
- **PASSTHRU:** Activates Modem Passthru.

Values: NONE, PASSTHRU

Default: NONE

3.3.5 Passthru Codec law

Console	SNMP	Text-based Config
Passthru Codec law	ifvcePassthruCodecLaw	[ifvce#] PassthruCodecLaw

Determines the PCM encoding method to be applied to the G711 traffic. The NetPerformer supports the following:

- **G711_ULAW:** This value refers to µlaw (*mu-law*), which is typically used with T1 lines for installations in North America
- **G711_ALAW:** Specifies alaw (*A-law*) encoding, which is typically used with E1 lines for installations in Europe.

Values: G711_ALAW, G711_ULAW

Default: G711_ULAW

4

Monitoring Digital Voice Connectionsl

4.1 About Digital Voice Connection Commands

Several commands are available from the console to view the status of digital voice connections on the NetPerformer:

- The Signaling Engine Information (**SEI**) command identifies what kind of interface cards are installed in the slots
- Use the **SLOT** option of the Display Counters (**DC**) command to view the traffic counters for the digital channels
- The **SLOT** option of the Display Errors (**DE**) command shows the values of the error counters
- To view the current status of a voice channel, including its *DSP relay mode*, use the **SLOT** option of the Display States (**DS**) command
- For a display of channel status in real time, execute the Display Call States (**DCS**) command
- View any alarms that may have occurred using the Display Alarms (**DA**) command.

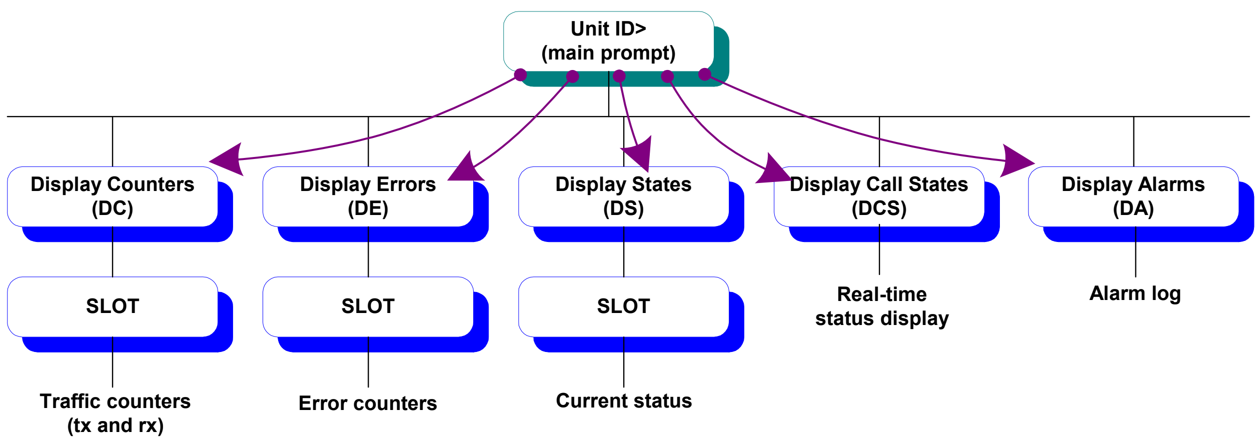

Figure 4-1: Display Commands in the CLI Tree for Digital Connections

4.2 Identifying the Interface Cards

To view what interface cards are installed in the unit:

- Enter **SEI** at the console command prompt.

  The available slots and their contents are listed after the Signaling Engine software information.

SEI example

```
SDM-9230>SEI
SIGNALING ENGINE INFORMATION

SIGNALING ENGINE SOFTWARE:
 Signaling Engine vX.X.X
 State: RUNNING

 SLOT 1> T1 Interface
 SLOT 2> E1 Interface - 120 ohms
 SLOT 3> E1 Interface - 75 ohms

DSP SOFTWARE:    QBXXX.BIN (X.X.X)
 Subfile 0x0078: TMSXXXVCXXXX code - 120/10MHz bootstrap
 Subfile 0x0001: TMSXXXVCXXXX code - ACELP-CN

   DSP SIMM         Interface 1        Interface 2        Interface
3
 1 Running(1)       7  No DSP          13 No DSP          19 No DSP
 2 Running(1)       8  No DSP          14 No DSP          20 No DSP
 3 Running(1)       9  No DSP          15 No DSP          21 No DSP
 4 Running(1)       10 No DSP          16 No DSP          22 No DSP
 5 Running(1)       11 No DSP          17 No DSP          23 No DSP
 6 Running(1)       12 No DSP          18 No DSP          24 No DSP
```

4.3 Displaying the Traffic Counters

To view all traffic counters for the digital channels:

- At the console command prompt, enter the menu sequence: **DC** ↵ **SLOT**

  All counters for all slots are displayed at the console screen. Press **<Enter>** to scroll through the display.

DC/SLOT example

```
SDM-9230>DC
DISPLAY COUNTERS
Item (BOOTP/CONFIG/DNS/IP/NAT/PORT/PVC/Q922/Q933/QOS/SLOT/SVC/
TIMEP,
def:BOOTP) ? SLOT
Counters (MEAN/PEAK,def:MEAN) ?
SLOT 3>
PORT 303> Transmitter rate......................0    % (M)
PORT 303> Receiver rate.........................0    % (M)
PORT 303> Number of frames transmitted..........1313
PORT 303> Number of frames received.............0
PORT 303> Number of octets transmitted..........64337
PORT 303> Number of octets received.............0

PORT 310> Transmitter rate......................0    % (M)
PORT 310> Receiver rate.........................0    % (M)
PORT 310> Number of frames transmitted..........86
PORT 310> Number of frames received.............146
PORT 310> Number of octets transmitted..........1118
PORT 310> Number of octets received.............16352

PORT 314> Transmitter rate......................0    % (M)
PORT 314> Receiver rate.........................0    % (M)
PORT 314> Number of frames transmitted..........1317
PORT 314> Number of frames received.............0
PORT 314> Number of octets transmitted..........64533
PORT 314> Number of octets received.............0

PORT 315> Transmitter rate......................0    % (M)
PORT 315> Receiver rate.........................95   % (M)
PORT 315> Number of frames transmitted..........0
PORT 315> Number of frames received.............120342
PORT 315> Number of octets transmitted..........0
PORT 315> Number of octets received.............11552832

PORT 316> Transmitter rate......................0    % (M)
PORT 316> Receiver rate.........................95   % (M)
PORT 316> Number of frames transmitted..........0
PORT 316> Number of frames received.............117878
PORT 316> Number of octets transmitted..........0
PORT 316> Number of octets received.............11316288
```

4.4 Displaying the Error Counters

To view all error counters:

1. At the console command prompt, enter the menu sequence: **DE ↵ SLOT**
2. Select the *Slot number*.

For SNMP access to these statistics, use the variables pertaining to the *statIfwan* category.

DE/SLOT example

```
SDM-9230>DE
DISPLAY ERRORS
Item (BOOTP/CHANNEL/DICT/GROUP/NAT/PORT/PU/PVC/Q922/SLOT/SVC/
TIMEP,
def:BOOTP) ? SLOT
SLOT> Slot number (1/2/3/ALL,def:1) ? 3
SLOT 3>
PORT 300> Number of errored seconds.............0
PORT 300> Number of severely errored seconds....0
PORT 300> Number of severely errored frames.....0
PORT 300> Number of unavailable seconds.........922
PORT 300> Number of controlled slip seconds.....0
PORT 300> Number of path code violations........0
PORT 300> Number of line errored seconds........0
PORT 300> Number of bursty errored seconds......0
PORT 300> Number of degraded minutes............0
PORT 300> Number of line code violations........0

PORT 300> Number of CRC errors..................0
PORT 300> Number of frame length errors.........0
PORT 300> Number of abort sequences.............0
PORT 300> Number of non-aligned octets..........0
PORT 300> Number of HDLC framing errors.........0
PORT 300> Number of returned rx buffers.........0
PORT 300> Number of channel restarts............0

PORT 303> Number of bad frames..................0       ------
PORT 303> Number of underruns...................0
PORT 303> Number of retries.....................0
PORT 303> Number of restarts....................35
PORT 303> Number of frames discarded (overrun)..0
PORT 303> Number of octets discarded (bad)......0
PORT 303> Number of octets discarded (overrun)..0

PORT 310> Number of bad frames..................0       ------
PORT 310> Number of underruns...................0
PORT 310> Number of retries.....................0
PORT 310> Number of restarts....................91
PORT 310> Number of frames discarded (overrun)..0
PORT 310> Number of octets discarded (bad)......0
PORT 310> Number of octets discarded (overrun)..0

Bad flags: U:Bad LENGTH  Q:Overflow  F:Flush  S:Overrun  B:Bad CRC
A:Abort
```

4.5 Displaying the Status Information

To view the current state of the digital link and channels:

1. At the console command prompt, enter the menu sequence: **DS ↵ SLOT**
2. Select the *Slot number*.

For SNMP access to these statistics, use the variables pertaining to the *statIfwan* category.

DS/SLOT example

```
CHICAGO>DS
DISPLAY STATES
Item (GLOBAL/PORT/PU/PVC/SIP/SLOT/SVC/VLAN,def:GLOBAL) ? SLOT
SLOT> Slot number (1/2/3/4/ALL,def:1) ? 2
SLOT 2>
PORT 200> Protocol.............................ANALOG FXS
PORT 200> State................................ENABLE

VOICE 201> State...............................CONNECTED
VOICE 201> Protocol............................G729
VOICE 201> Last error..........................NONE
VOICE 201> Fax State...........................CONNECTED
VOICE 201> DSP relay rate......................
VOICE 201> DSP relay mode......................MPAS

VOICE 301> State...............................IDLE
VOICE 301> Protocol............................ACELP-CN
VOICE 301> Last error..........................CALL BUSY
VOICE 301> DSP relay rate......................NO DSP
VOICE 301> DSP relay mode......................NO DSP

VOICE 302> State...............................IDLE
VOICE 302> Protocol............................PCM64K
VOICE 302> Last error..........................CANCELLED BY HOST
VOICE 302> DSP relay rate......................NO DSP
VOICE 302> DSP relay mode......................NO DSP
```

NOTE: The *DSP relay mode* statistic indicates **MPAS** when Modem Passthru is active. Other values include:

MOD: Modem spoofing mode (NetPerformer base product only), also referred to as Modem Relay

FAX: Fax mode: standard fax on the NetPerformer base product or T.38 fax on the SIP VoIP licensed option.

4.6 Real-time Status Display

To view a continuously updated display of the digital link and channel states:

- Enter **DCS** at the console command prompt.

  The various statistics displayed on the screen are updated dynamically.

- To quit from this command, press any key other than **<Home>**, **<End>** or the up and down arrow keys.

DCS example

```
---------------------------- SLOT 2 : T1 -----------------------------------
 # Status  DNIS              Rate       | # Status  DNIS              Rate
01 CONNECT 1234567890123456 MPAS       |13 IDLE    1234567890123456 G726 32Kx2
02 SETUP   1234567890123456 G726 16Kx2 |14 PROGRES 1234567890123456 G729A 8Kx4
03 PROGRES 1234567890123456 G729A 8Kx4 |15 PROGRES 1234567890123456 G729A 8Kx4
04 CONNECT 1234567890123456 MPAS       |16 PROGRES 1234567890123456 G729A 8Kx4
05 DISCONN 1234567890123456 G729A 8Kx4 |17 PROGRES 1234567890123456 G729A 8Kx4
06 OFF                                 |18 OFF
07 OFF                                 |19 OFF
08 OFF                                 |20 OFF
09 OFF                                 |21 OFF
10 OFF                                 |22 OFF
11 OFF                                 |23 OFF
12 OFF                                 |24 OFF
-----------------------------------------------------------------------------
```

NOTE: The *Rate* column provides the same information as the *DSP relay mode* statistic in the **DS/SLOT** example on “DS/SLOT example” on page 4-6.

4.7 Displaying the Alarms

To view any alarms that may have occurred on the digital links and channels:

- At the console command prompt, enter **DA**.

DA example

```
SDM-9380>DA
DISPLAY ALARMS
SDM-9380 vX.X.X Memotec Inc. (c) 2004
Signaling Engine vX.X.X Memotec Inc. (c) 2004
DSP code version: X.X.X
Console connected on port CSL
Time> WED   2004/04/14 13:34:22
Alarm>    BACKUP CALL, LINK 101              WED   2004/04/14
12:30:18
Alarm>    SOFT START (RST)                   WED   2004/04/14
12:26:25
Alarm>   SETUP RESET                       WED   2004/04/14 12:25:10
Alarm>    SOFT START (RST)                   WED   2004/04/14
9:49:42
Alarm>    FIRMWARE STORED                    WED   2004/04/14
9:48:20
Alarm>    SOFT START (PWR)                   MON   2004/04/12
9:09:33
Alarm>    LINK 101 DOWN (SDM-9380)           THU   2004/04/12
7:24:43
```

5

Application Examples

5.1 Digital Voice over T1 using QSIG Signaling

This application uses QSIG signaling on a T1 line to transport voice traffic from unit B to unit C via the central unit A.

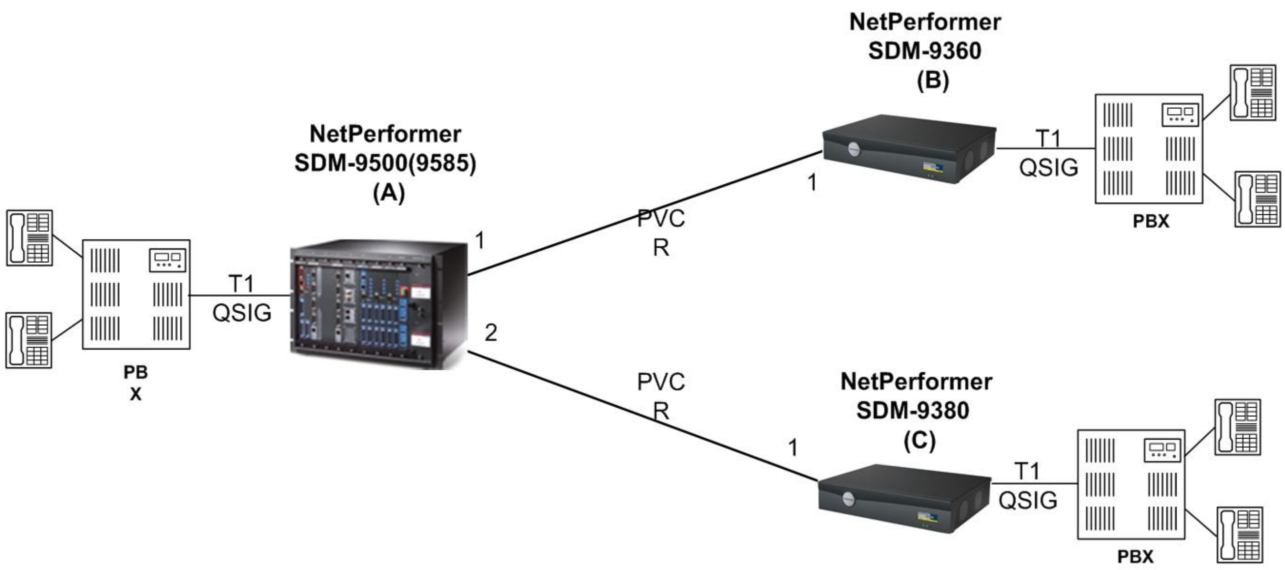

Figure 5-1: Digital Voice over T1 using QSIG Application

NOTE: Only the central site configuration (unit A) is provided here. The configuration required at the remote sites (units B and C) is similar to that for the central site.

5.1.1 Global Parameters

```
GLOBAL> Unit name..............................CENTRAL-A
GLOBAL> Unit routing version...................1
GLOBAL> Contact name...........................Memotec
GLOBAL> Unit location..........................Unknown
GLOBAL> Loopback...............................NO
GLOBAL> Link timeout delay.....................0
GLOBAL> Transit delay (sec)....................4
GLOBAL> Daylight saving time...................NO
GLOBAL> Default IP address.....................000.000.000.000
GLOBAL> Default IP mask (number of bits)........0
{000.000.000.000}
GLOBAL> Default gateway........................000.000.000.000
GLOBAL> SNMP trap: IP address #1...............000.000.000.000
GLOBAL> SNMP trap: IP address #2...............000.000.000.000
GLOBAL> SNMP trap: IP address #3...............000.000.000.000
GLOBAL> SNMP trap: IP address #4...............000.000.000.000
GLOBAL> Frame relay status change trap.........DISABLE
```

```
GLOBAL> Watch power supplies and fans..........NONE
GLOBAL> Local unit DLCI address................0
GLOBAL> Extension number (no. of digits)........3
GLOBAL> Country code...........................1
GLOBAL> Jitter buffer (msec)...................40
GLOBAL> Enable voice/fax log...................YES
GLOBAL> Dial timer (sec).......................2
GLOBAL> High priority voice class..............YES
GLOBAL> Global CIR for FR over IP..............64000
GLOBAL> Timer in ms for FR over IP.............50
GLOBAL> Max number of voice channels over IP....10000
GLOBAL> Delay generated by a comma (ms).........250
GLOBAL> Auto save configuration delay (sec) ....10
GLOBAL> Enable VTR (Voice Traffic Routing)......NO
GLOBAL> Enable Domain Dialing..................YES
GLOBAL> Enable hunt forwarding.................YES
GLOBAL> Enable user access logging.............NO
GLOBAL> Exclusive access to console............DISABLE
```

5.1.2 Port Parameters (Ports 1 and 2)

```
PORT 1-2> Protocol.............................PVCR
PORT 1-2> Port speed (bps).....................1534000
PORT 1-2> Interface............................DCE-V35
PORT 1-2> Clocking mode........................INTERNAL
PORT 1-2> Mode.................................DEDICATED
PORT 1-2> IP address...........................000.000.000.000
PORT 1-2> Subnet mask (number of bits)..........0
{000.000.000.000}
PORT 1-2> IP RIP...............................V1
PORT 1-2> IP RIP TX/RX.........................DUPLEX
PORT 1-2> OSPF.................................DISABLE
PORT 1-2> IP multicast active..................NO
PORT 1-2> IP multicast protocol................NONE
PORT 1-2> NAT enable...........................NO
PORT 1-2> IPX RIP..............................DISABLE
PORT 1-2> IPX SAP..............................DISABLE
PORT 1-2> IPX network number...................00000000
PORT 1-2> Compression..........................YES
PORT 1-2> Remote unit name.....................REMOTE-B & REMOTE-
C
PORT 1-2> Timeout (msec).......................1000
PORT 1-2> Number of retransmission retries.......100
PORT 1-2> Maximum number of voice channels.......10000
PORT 1-2> Maximum Voice Channels If High Priority Data  10000
PORT 1-2> Cell Packetization...................YES
```

5.1.3 Slot 1 Parameters

```
PORT #100> Status..............................ENABLE
PORT #100> Clock recovery......................DISABLE
PORT #100> Digital port clock source...........INTERNAL
PORT #100> Signaling mode......................QSIG
PORT #100> QSIG master.........................NO
PORT #100> Channel selection mode..............PREFERRED
PORT #100> Local number........................
```

```
PORT #100> Local subaddress....................
PORT #100> Pcm encoding law....................MU-LAW
PORT #100> Idle code...........................7F
PORT #100> Zero suppression mode...............B8ZS
PORT #100> Gain limit..........................-30DB
PORT #100> Framing mode........................ESF
PORT #100> Line Build Out......................0-133FT
PORT #100> Loopback............................DISABLE

PORT #101-123> Protocol............................ACELP-CN
PORT #101-123> Timeslot............................1-23
VOICE #101-123> DSP packets per frame             1234
VOICE #101-123> 8K packetization selection (Y/N)...YNNN
VOICE #101-123> DSP packets per frame             12345
VOICE #101-123> 6K packetization selection (Y/N)...NNNNN
VOICE #101-123> Comfort noise level................0
VOICE #101-123> Local inbound voice level (db)......0
VOICE #101-123> Local outbound voice level (db).....0
VOICE #101-123> Priority Level.....................0
VOICE #101-123> Echo canceler......................ENABLE
VOICE #101-123> Double talk threshold (db).........6
VOICE #101-123> Pulse frequency (pps)..............10
VOICE #101-123> Activation type....................SWITCHED
VOICE #101-123> Link down busy.....................NO
VOICE #101-123> TONE type:.........................DTMF
VOICE #101-123> TONE regeneration:.................255
VOICE #101-123> TONE ON (msec).....................100
VOICE #101-123> TONE OFF (msec)....................100
VOICE #101-123> Pulse make/break ratio.............34
VOICE #101-123> Fax/modem relay....................FAX
VOICE #101-123> Maximum fax/modem rate.............14400
VOICE #101-123> Hunt Group active..................A
VOICE #101-123> Delete digits......................0
VOICE #101-123> Port extension number..............101-123
VOICE #101-123> Fwd digits.........................NONE
VOICE #101-123> Enable DTMF ON-TIME configuration...NO
```

5.1.4 Map Parameters

```
MAP #1> Entry digits...........................200
MAP #1> Destination Name.......................REMOTE-C
MAP #1> Destination Extension number source.....HUNT
MAP #1> Destination Extension number............A
MAP #1> Extended digits source..................NONE
MAP #1> Extended digits to forward.............NONE
MAP #1> Use SVC connection.....................NO

MAP #2> Entry digits...........................100
MAP #2> Destination Name.......................REMOTE-B
MAP #2> Destination Extension number source.....HUNT
MAP #2> Destination Extension number............A
MAP #2> Extended digits source..................NONE
MAP #2> Extended digits to forward.............NONE
MAP #2> Use SVC connection.....................NO
```

5.2 Transparent Signaling for Digital Voice

This feature permits the transport of PBX signaling packets in transparent mode between NetPerformer units over a PowerCell connection (WAN link).

- The PBX signaling information is transported transparently through the NetPerformer, and is typically located in timeslot 16 of an E1 interface card, or channel 24 of a T1 interface card.

  On the NetPerformer, transparent signaling can be configured on any E1 or T1 timeslot.

- All voice calls are permanently active and must be configured for **AUTODIAL** or **PREDEFINED** line activation.

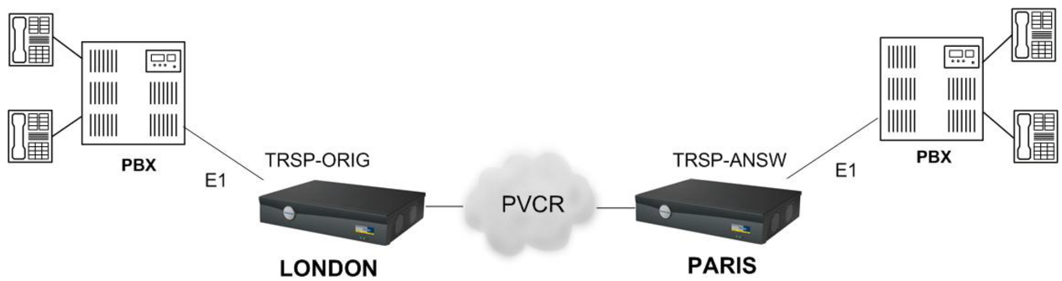

Figure 5-2: Transparent Signaling Application

With transparent signaling, one timeslot is used to provide a common channel signal at a rate of 64 Kbps. A T1 or E1 port on one NetPerformer is defined as the originating side (**TRSP-ORIG**), and a T1 or E1 port on the other NetPerformer is defined as the answering side (**TRSP-ANSW**). The **TRSP-ORIG** port initiates the PowerCell connection.

The T1 or E1 card sets the AB bits to generate **OFF HOOK/ON HOOK** conditions to all active timeslots until the PowerCell connection is made. The connection window is determined from the global *Link timeout delay* parameter. If a connection is not made within this period the T1/E1 port will time out, go **ON HOOK** for another timeout period, then back **OFF HOOK** again. This sequence repeats until the PowerCell connection is successful.

5.2.1 Configuring the Originating Side (TRSP-ORIG)

When configuring the T1/E1 port on the originating side, the link *Signaling mode* must be set to **TRSP-ORIG**, using the **SETUP/SLOT/#/CHANNEL** submenu.

SE/SLOT/#/ LINK example: TRSP-ORIG mode

```
LONDON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2,def:1) ?
Item (LINK/CHANNEL,def:LINK) ?
PORT 100> Status (def:DISABLE) ? ENABLE
PORT 100> Clock recovery (def:DISABLE) ?
PORT 100> Digital port clock source (def:INTERNAL) ?
PORT 100> Signaling mode (def:NONE) ? TRSP-ORIG
PORT 100> Pcm encoding law (def:A-LAW) ?
PORT 100> Hunt Group Sorting (def:RRA) ?
PORT 100> Idle code (def:7E) ?
PORT 100> Zero suppression mode (def:HDB3) ?
PORT 100> Gain limit (def:-12DB) ?
PORT 100> CRC4 mode (def:ENABLE) ?
PORT 100> International bit (def:ENABLE) ?
PORT 100> ETS 300 011 mode (def:DISABLE) ?
PORT 100> Loopback (def:DISABLE) ?
```

NOTE: You must also set the global *Link timeout delay* (**GLOBAL** submenu of the **SETUP** command) to allow sufficient time for the T1/E1 interface to fake an **OFF HOOK** condition to all 30 DS0s (default 10 seconds).

5.2.2 Configuring the Answering Side (TRSP-ANSW)

On the answering side, the T1 or E1 link must be defined with the **TRSP-ANSW** *Signaling mode*.

SE/SLOT/#/ LINK example: TRSP-ANSW mode

```
PARIS>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2,def:1) ?
Item (LINK/CHANNEL,def:LINK) ?
PORT 100> Status (def:DISABLE) ? ENABLE
PORT 100> Clock recovery (def:DISABLE) ?
PORT 100> Digital port clock source (def:INTERNAL) ?
PORT 100> Signaling mode (def:TRSP-ORIG) ? TRSP-ANSW
PORT 100> Pcm encoding law (def:A-LAW) ?
PORT 100> Hunt Group Sorting (def:RRA) ?
PORT 100> Idle code (def:7E) ?
```

```
PORT 100> Zero suppression mode (def:HDB3) ?
PORT 100> Gain limit (def:-12DB) ?
PORT 100> CRC4 mode (def:ENABLE) ?
PORT 100> International bit (def:ENABLE) ?
PORT 100> ETS 300 011 mode (def:DISABLE) ?
PORT 100> Loopback (def:DISABLE) ?
```

NOTE: The global *Link timeout delay* parameter is not applied on the answering side.

5.2.3 Configuring the Voice Channels

All voice channels must be defined as follows (**SETUP/SLOT/#/CHANNEL** submenu):

- Set the *Signaling type* set to **IMMEDIATE START**
- Set the *Activation type* to **PREDEFINED** or **AUTODIAL**
  - The default settings for cards in matching slots permit connection in **PREDEFINED** mode
  - The **AUTODIAL** activation type can also be used, but this requires that speed dial numbers be configured in the Voice Mapping Table. Refer to the chapter *Configuring the Voice Mapping Table* in the *Analog Voice* module of this document series.
- Define the *Remote unit* and *Remote port number* on the originating side to reach the correct T1 or E1 channel on the answering side.

SE/SLOT/#/ CHANNEL example: Voice channel

```
PARIS>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2,def:1) ?
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Channel Number (101-131/ALL,def:101) ?
PORT 101> Protocol (def:OFF) ? ACELP-CN
VOICE 101> Timeslot (def:1) ?
VOICE 101> DSP packets per frame              1234
VOICE 101> 8K packetization selection (Y/N) ? YNNN
VOICE 101> DSP packets per frame              12345
VOICE 101> 6K packetization selection (Y/N) ? NNNNN
VOICE 101> Comfort noise level (def:0) ?
VOICE 101> Signaling type (def:IMMEDIATE START) ?
VOICE 101> Hoot & Holler application (def:NO) ?
VOICE 101> Local inbound voice level (db) (def:0) ?
VOICE 101> Local outbound voice level (db) (def:-3) ?
VOICE 101> Priority Level (0-10,def:0) ?
VOICE 101> Echo canceler (def:ENABLE) ?
VOICE 101> Double talk threshold (db) (def:6) ?
VOICE 101> Activation type (def:PREDEFINED) ?
VOICE 101> Link down busy (def:NO) ?
```

```
VOICE 101> TONE type: (def:DTMF) ?
VOICE 101> TONE regeneration: (0-255,def:1) ?
VOICE 101> TONE ON (msec) (30-1000,inc:10,def:100) ?
VOICE 101> TONE OFF (msec) (30-1000,inc:10,def:100) ?
VOICE 101> Pulse make/break ratio (30-50,inc:4,def:34) ?
VOICE 101> Fax relay (def:FAX) ?
VOICE 101> Maximum fax rate (def:14400) ?
VOICE 101> ECM mode (def:DISABLE) ?
VOICE 101> Modem relay (def:NONE) ?
VOICE 101> Remote unit (def:NONE) ? LONDON
VOICE 101> Remote port number (1-65534,def:101) ?
VOICE 101> DTMF power ratio (5-100,def:5) ?
VOICE 101> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 101> Redundant channel (def:NO) ?
VOICE 101> Egress ANI operation mode (def:NONE) ?
VOICE 101> Egress CHANNEL ANI digits (def:) ?
VOICE 101> Ingress ANI operation mode (def:NONE) ?
VOICE 101> Ingress CHANNEL ANI digits (def:) ?
```

5.2.4 Configuring a Transparent Signaling Channel

In addition to voice channels, you must define a transparent signaling channel to carry the PBX signaling information through the NetPerformer without interpretation or processing.

NOTE: **Signaling is no longer reserved to channel 16 on an E1 card, or channel 24 on a T1 card.** Any channel on the NetPerformer T1/E1 cards can be configured to carry signaling information transparently.

The *Protocol* of the signaling channel is typically set to **TRANSPARENT**, but other possibilities exist:

- On the SDM-9230, the **PASSTHRU** protocol provides the same type of transport as the **TRANSPARENT** protocol, but without the use of DSP resources
- The **SS7** protocol should be selected to transport SS7 signaling. With this protocol, the NetPerformer provides ISU spoofing, which improves performance and bandwidth utilization for SS7 applications.

SE/SLOT/#/CHANNEL example: Transparent channel

```
PARIS>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2,def:1) ?
Item (LINK/CHANNEL,def:CHANNEL) ?
SLOT> Channel Number (101-131/ALL,def:102) ? 116
VOICE 116> Protocol (def:OFF) ? TRANSPARENT
VOICE 116> Timeslot (def:2) ? 16
VOICE 116> Remote unit (def:NONE) ? LONDON
VOICE 116> Remote port number (1-65534,def:116) ?
```

```
VOICE 116> Redundant channel (def:NO) ?
```

TIP: With Transparent signaling, all T1/E1 channels are always up. To minimize the bandwidth used, set the *Silence suppression level* or *Comfort noise level* parameter to a value higher than **0** (default setting).

Using ACELP-CN 8Kbps single buffering, this provides 112 packets per second of full-duplex voice traffic when voice is detected. If there is no voice traffic, the DSP will send only 1 packet per second to keep the voice channel alive.

6

SE/SLOT/#/LINK Configuration Parameters

6.1 T1 Port

6.1.1 Common Parameters

These parameters are common to all T1 channels, regardless of the *Signaling mode* defined. Some of the parameters detailed in this section also apply to configuration of an E1 or ISDN-BRI S/T port.

Framer Type

Console	SNMP	Text-based Config
Framer Type	ifwanE1T1Type	[ifwan#] E1T1Type

For an E1/T1 interface only (Dual Framer)
Sets the type of framer used on this physical port.

- **E1:** The physical link uses E1 framing (32 channels)
- **T1:** The physical link uses T1 framing (24 channels).

Values: E1, T1

Default: E1

Status

Console	SNMP	Text-based Config
Status	ifwanT1E1Status	[ifwan#] T1E1Status

Sets the activation status of this physical port.

- **ENABLE:** Activates the physical link
- **DISABLE:** The physical link is not activated.

NOTE: This means that all data/voice channels associated with this port are disabled, but the channel configuration is not lost.

Values: DISABLE, ENABLE

Default: DISABLE

Clock recovery

Console	SNMP	Text-based Config
Clock recovery	ifwanClockRecovery	[ifwan#] ClockRecovery

Specifies whether the clock received on this digital port can be used as the master clock for all digital ports or as one of the backup clocks.

- **ENABLE:** The clock received on this port provides the clock for other digital ports
- **DISABLE:** The clock received on this port is not used elsewhere.

Values: DISABLE, ENABLE

Default: DISABLE

Digital port clock source

Console	SNMP	Text-based Config
Digital port clock source	ifwanDigitalPortClkSrc	[ifwan#] DigitalPortClkSrc

Specifies the source of the master clock for digital ports. This parameter indicates whether the clock should be generated internally, or taken from one of the interface cards installed in the NetPerformer unit. *Digital port clock source* can be used in conjunction with *Clock recovery* to optimize the clock distribution. Set *Digital port clock source* to:

- **Numeric value:** The slot containing the digital interface card that is connected to the network (**1**, **2**, …)

  Select a numeric value if you want all transmit clocks to be driven by the receive clock of the digital interface card that is installed in that slot number.

- **INTERNAL:** To allow the NetPerformer to use its internal reference clock.

  Select **INTERNAL** (the default value) if you want all transmit clocks to be driven by the internal reference clock (up to 1.544 Mbps for T1, up to 2.048 Mbps for E1, or up to 128 Kbps for ISDN-BRI S/T).

NOTE: **The value you select for this parameter will be applied globally to all digital ports on the same NetPerformer unit.**

Values: INTERNAL, 1, 2, 3, 4 (numeric values are product-dependent)

Default: INTERNAL

Signaling mode

Console	SNMP	Text-based Config
Signaling mode	ifwanSignaling	[ifwan#] Signaling

Specifies the type of signaling that is in effect on this digital port. The setting you select determines what other parameters must be defined, as detailed in following sections.

For data transport on a T1 interface card the *Signaling mode* can be set to:

- **NONE:** data-only connection
- **NONE (30 TS):** data-only connection

NOTE: When the E1 framer ports are configured with NONE (30 TS), the behavior is similar to “NONE” signaling mode, except that the timeslot 16 is not used for a configuration with many consecutive timeslots.

- **ROB BIT:** T1 Robbed-bit signaling
- **TRSP-ORIG** (originate) or **TRSP-ANSW** (answer): Transparent HDLC-based or PCM64K-based transport

NOTE: **TRSP-ORIG** and **TRSP-ANSW** signaling modes are not available when the NetPerformer SIP VoIP licensed software option has been installed. They are used to establish a permanent point-to-point transparent signaling connection between digital interfaces (T1 or E1).

- **NTT** or **KDD:** ISDN access in Japan
- **4ESS**, **5ESS**, **DMS100** or **NI2:** ISDN access in other countries (mainly North America)
- **QSIG:** Standard PBX access with supplementary services support.

Values: NONE, ROB BIT, TRSP-ORIG, TRSP-ANSW, NTT, KDD, 4ESS, 5ESS, DMS100, NI2, QSIG

Default: NONE

Interface Mode

Console	SNMP	Text-based Config
Interface Mode	ifwanInterfaceMode	[ifwan#] InterfaceMode

For an E1/T1 interface only (Dual Framer)

Sets the type of interface mode used on this physical port. **The two ends of the digital connection must have opposite NT/TE values.**

- **TE:** The interface uses the receive clock from the connected device to transmit data
- **NT:** The interface provides the clock signal to the connected device.

NOTE: On the T1, legacy E1-120 and ISDN-BRI S/T interface cards, the interface mode is set through hardware strapping . For strapping instructions, refer to the *Hardware Installation Guide* for your NetPerformer product.

Values: TE, NT

Default: TE

6.1.2 NONE Signaling Mode (on T1)

NOTE: Some of the parameters detailed in this section also apply to other signaling modes on the T1 LINK.

Pcm encoding law

Console	SNMP	Text-based Config
Pcm encoding law	ifwanEncodingLaw	[ifwan#] EncodingLaw

Specifies the PCM coding law in effect on this interface.

Values: A-LAW, MU-LAW

Default: MU-LAW

Hunt Group Sorting

Console	SNMP	Text-based Config
Hunt Group Sorting	ifwanHuntRules	[ifwan#] HuntRules

Specifies the Hunt Group sorting rules in effect on this interface when it is used for placing a voice call:

- **LSA:** Linear Selection Ascending. When the NetPerformer tries to place a call, **timeslot selection always starts with the lowest numbered timeslot**, and cycles upward to higher numbered timeslots until a free timeslot is found.
- **LSD:** Linear Selection Descending. When the NetPerformer tries to place a call, **timeslot selection always starts with the highest numbered timeslot**, and cycles downward to lower numbered timeslots until a free timeslot is found.
- **RRA:** Round Robin Ascending. When the NetPerformer tries to place a call, **timeslot selection starts with the timeslot that is one number higher than the last used timeslot on that interface**, and cycles upward to higher numbered timeslots until a free timeslot is found. If the highest numbered timeslot is reached, the cycle continues with timeslot 1.
- **RRD:** Round Robin Descending. When the NetPerformer tries to place a call, **timeslot selection starts with the timeslot that is one number lower than the last used timeslot on that interface**, and cycles downward to lower numbered timeslots until a free timeslot is found. If timeslot 1 is reached, the cycle continues with the highest numbered timeslot on the interface.

For further details on Hunt Group sorting rules, refer to the *Advanced Voice Features* module of this document series.

Values: LSA, LSD, RRA, RRD

Default: RRA

Idle code

Console	SNMP	Text-based Config
Idle code	ifwanIdleCode	[ifwan#] IdleCode

Specifies the byte that will be transmitted when there is no real data to transmit over this port.

Values: OO - FF

Default: 7F

Zero suppression mode

Console	SNMP	Text-based Config
Zero suppression mode	ifwanLineCoding	[ifwan#] LineCoding

Specifies the type of 1's density control or zero suppression on this link.

Values: B8ZS, AMI, B7ZS

Default: B8ZS

Long Haul

Console	SNMP	Text-based Config
Long Haul	ifwanLongHaul	[ifwan#] LongHaul

For an E1/T1 interface only (Dual Framer)
Specifies whether the connection is made over a long haul (**YES**) or short haul (**NO**). This determines the gain to be applied on the received DS1 signal.

Values: NO, YES

Default: NO

Gain limit

Console	SNMP	Text-based Config
Gain limit	ifwanGainLimit	[ifwan#] GainLimit

For a T1 or E1 interface only.
Specifies the gain to be applied on the received DS1 signal.

Values: -30DB, -36DB

Default: -30DB

Framing mode

Console	SNMP	Text-based Config
Framing mode	ifwanFraming	[ifwan#] Framing

Determines the framing format (**ESF** or **D4**).

Values: ESF, D4

Default: ESF

Line Build Out

Console	SNMP	Text-based Config
Line Build Out	ifwanLineBuild	[ifwan#] LineBuild

Specifies the build-out to be applied on the transmitted DS1 signal. The *Line Build Out* adjusts waveshaping on the T1 driver to match lines of different lengths.

Values: 0-133FT, 133-266FT, 266-399FT, 399-533FT, 533-655FT,
-7.5DB, -15DB, -22.5DB

Default: 0-133FT

Custom Waveform

Console	SNMP	Text-based Config
Custom Waveform	ifwanCustomWaveform	[ifwan#] CustomWave-form

For an E1/T1 interface only (Dual Framer)
Determines whether custom waveshaping will be applied to this interface. When this parameter is set to **ENABLE**, the custom waveform overrides the waveshaping determined from the *Line Build Out* parameter.

When you enable *Custom Waveform*, you can customize the waveform parameters using the **E1-T1 WAVEFORM** submenu of the **SE/CUSTOM** console command. Start by loading a standard waveform template, either **E1 75 OHMS** or **E1 120 OHMS**, and adjust the parameters that are displayed.

NOTE: Custom waveshaping is not configurable using SNMP or text-based config-ureation.

Values: DISABLE, ENABLE

Default: DISABLE

Loopback

Console	SNMP	Text-based Config
Loopback	ifwanT1E1LoopBack	[ifwan#] T1E1LoopBack

Determines whether a loopback condition is activated on this physical port. Set *Loopback* to **ENABLE** for troubleshooting purposes only. For details, refer to the *Monitoring and Statistics* module of this document series.

Values: DISABLE, ENABLE

Default: DISABLE

PVCR Link management

Console	SNMP	Text-based Config
PVCR Link management	ifwanPvcrLinkManage-ment	[ifwan#] PvcrLinkMan-agement

For an E1/T1 interface only (Dual Framer)
Determines whether the PVCR Link Management feature will be enabled on this interface.

Values: DISABLE, ENABLE

Default: DISABLE

When *PVCR Link management* is set to **ENABLE**, the following three parameters are displayed at the NetPerformer console.

Link management alarm type

Console	SNMP	Text-based Config
Link management alarm type	ifwanPvcrManagement-AlarmType	[ifwan#] PvcrManage-mentAlarm-Type

For an E1/T1 interface only (Dual Framer) when PVCR Link Managment is enabled

Determines what type of alarm will be generated on this interface for the PVCR Link Management feature:

- **RAI/YELLOW:** For an RAI alarm on an E1 interface or a YELLOW alarm on a T1 interface.

  The **ALM** LED for the interface will turn **yellow** when this type of alarm is detected.

- **AIS:** For an AIS alarm on either an E1 or T1 interface.

  The **ALM** LED on the interface will turn **red** when this type of alarm is detected.

Values: RAI/YELLOW, AIS

Default: RAI/YELLOW

Remote unit

Console	SNMP	Text-based Config
Remote unit	ifwanLinkRemoteUnit	[ifwan#] LinkRemoteUnit

For an E1/T1 interface only (Dual Framer) when PVCR Link Managment is enabled

Determines which remote NetPerformer will be monitored for an alarm signal. Enter the *Unit name* that has been configured on the remote unit. The *Unit name* is defined on the remote unit using the **SETUP/GLOBAL** submenu. Refer to the appendix *SE/GLOBAL Configuration Parameters* in the *Quick Configuration* module of this document series.

NOTE: The decision to send or remove an alarm on the E1/T1 interface is based on the loss or recovery of the PVCR link to the unit specified by the *Remote unit* parameter.

Values: Maximum 32-character alphanumeric string

Default: no value

Remote port number

Console	SNMP	Text-based Config
Remote port number	ifwanLinkRemotePort	[ifwan#] LinkRemotePort

For an E1/T1 interface only (Dual Framer) when PVCR Link Managment is enabled

Determines which remote digital LINK will receive the alarm indication. Enter the port number that corresponds to the LINK configuration on the remote side (**100**, **150**, **200**, **250**, **300**, **350**, **400** or **450**). By default, the same number as the local LINK number is selected.

Values: product-dependent LINK port numbers **(100, 150, 200, 250, 300, 350, 400, 450)**

Default: the local LINK port number

6.1.3 ROB BIT Signaling Mode

All parameters for **ROB BIT** signaling mode are common to a T1 port with no signaling. Refer to "NONE Signaling Mode (on T1)" on page 6-5.

6.1.4 TRSP-ORIG and TRSP-ANSW Signaling Modes (on T1)

All parameters for **TRSP-ORIG** and **TRSP-ANSW** signaling modes are common to a T1 port with no signaling. Refer to "NONE Signaling Mode (on T1)" on page 6-5.

6.1.5 CCS Signaling Modes

On a T1 card, the CCS signaling modes that can be used for digital data transport are: **NTT**, **KDD**, **4ESS**, **5ESS**, **DMS100**, **NI2** and **QSIG**.

NOTE: CCS parameters that are common to a T1 port with no signaling are listed under "NONE Signaling Mode (on T1)" on page 6-5.

CCS side

Console	SNMP	Text-based Config
CCS side	ifwanCcsSide	[ifwan#] CcsSide

Reflects which side of the CCS application the NetPerformer unit is facing via this channel:

- **USER:** Select this value when the unit is facing the network side of the application
- **NETWORK:** Select this value when the unit is facing the user side of the application.

Caution: The two sides of a CCS connection must have opposite values for this parameter.

Values: USER, NETWORK

Default: USER

QSIG master

Console	SNMP	Text-based Config
QSIG master	ifwanQsigMaster	[ifwan#] QsigMaster

For QSIG Signaling mode only:
Reflects the type of QSIG equipment the NetPerformer unit is facing via this port:

- **YES:** Select this value when the unit connects to equipment that is defined as slave
- **NO:** Select this value when the unit connects to equipment that is defined as master.

Caution: The two sides of a QSIG connection must have opposite values for this parameter.

Values: YES, NO

Default: NO

Channel selection mode

Console	SNMP	Text-based Config
Channel selection mode	ifwanChannelSelection-Mode	[ifwan#] ChannelSelectionMode

When the NetPerformer sends an ISDN Setup message to the PBX or CO, it suggests a channel number for the connection. The *Channel selection mode* parameter determines the way in which the NetPerformer selection will be interpreted during call setup.

NOTE: **This parameter is not required for a data connection, and can be left at its default value, PREFERRED.** It is applied only to outgoing ISDN/QSIG calls that are initiated by the NetPerformer unit.

Values: PREFERRED, EXCLUSIVE

Default: PREFERRED

Local number

Console	SNMP	Text-based Config
Local number	ifwanLocalNum	[ifwan#] LocalNum

Defines the local ISDN number for the digital port. This number is used when multiple ISDN devices are connected on the same bus.

The *Local number* is optional. When it is not defined, the NetPerformer will process every ISDN Setup message received on this port.

When the *Local number* is defined, the NetPerformer ignores or processes a Connection Request received on this port based on the value of this parameter. This value is validated against the Called Party Number included in the ISDN Setup message that initiates the connection.

Values: maximum 20 digits (**0** to **9**)

Default: no value

Local subaddress

Console	SNMP	Text-based Config
Local subaddress	ifwanLocalSubAddr	[ifwan#] LocalSubAddr

Defines the local ISDN subaddress for the digital port. This address is used when multiple ISDN devices with the same subscriber number are connected on the same bus. It uniquely identifies the NetPerformer unit in an application where more than one device can be reached using a single ISDN number.

The *Local subaddress* is optional. When it is not defined, the NetPerformer will process every ISDN Setup message received on this port (provided that the Setup message includes a Called Party Number which matches the value of the *Local number*).

When the *Local subaddress* is defined, the NetPerformer ignores or processes a Connection Request received at this port based on the value of this parameter. This value is validated against the Called Party Subaddress included in the ISDN Setup message that initiates the connection.

Values: maximum 20 digits (**0** to **9**)

Default: no value

For an application that uses ISDN signaling you can define the *Numbering Plan* that must be used when placing a call. Five parameters control the numbering plan definition:

- *Calling number type of network*
- *Calling number numbering plan*

- *Called number type of network*
- *Called number numbering plan*
- Override remote types and plans

The values of the first four of these parameters are based on definitions in the ISDN standard specifications. **Configure each parameter according to the current configuration of the switch/PBX.** The last parameter is used to override the numbering plan configured on the remote unit when ISDN signaling is the same on both sides of the connection.

Calling number type of network

Console	SNMP	Text-based Config
Calling number type of network	ifwanCallingTypeNumber	[ifwan#] CallingTypeN-umber

Defines the network at the calling number location, using a 3-digit binary code:

- **000:** Unknown. This is the default value.
- **001:** International
- **010:** National
- **011:** Network specific
- **100:** Subscriber number
- **110:** Abbreviated number
- **111:** Reserved.

Values: 000 (Unknown), 001 (International), 010 (National), 011 (Network specific), 100 (Subscriber number), 110 (Abbreviated number), 111 (Reserved)

Default: 000 (Unknown)

Calling number numbering plan

Console	SNMP	Text-based Config
Calling number number-ing plan	ifwanCallingNumbering-Plan	[ifwan#] CallingNumber-ingPlan

Defines the numbering plan used at the calling number location, using a 4-digit binary code:

- **0000:** Unknown. This is the default value.
- **0001:** Telephony numbering using E.164
- **0011:** Data-based numbering using X.121
- **0100:** Telex numbering using F.69

- **1000:** National standard
- **1001:** Private
- **1111:** Reserved.

Values: 0000 (Unknown), 0001 (Telephony numbering E.164), 0011 (Data X.121), 0100 (Telex F.69), 1000 (National standard), 1001 (Private), 1111 (Reserved)

Default: 0000 (Unknown)

Called number type of network

Console	SNMP	Text-based Config
Called number type of network	ifwanCalledTypeNumber	[ifwan#] CalledTypeNumber

Defines the network at the called number location, using a 3-digit binary code:

- **000:** Unknown. This is the default value.
- **001:** International
- **010:** National
- **011:** Network specific
- **100:** Subscriber number
- **110:** Abbreviated number
- **111:** Reserved.

Values: 000 (Unknown), 001 (International), 010 (National), 011 (Network specific), 100 (Subscriber number), 110 (Abbreviated number), 111 (Reserved)

Default: 000 (Unknown)

Called number numbering plan

Console	SNMP	Text-based Config
Called number numbering plan	ifwanCalledNumberingPlan	[ifwan#] CalledNumberingPlan

Defines the numbering plan used at the called number location, using a 4-digit binary code:

- **0000:** Unknown. This is the default value.
- **0001:** Telephony numbering using E.164
- **0011:** Data-based numbering using X.121
- **0100:** Telex numbering using F.69
- **1000:** National standard

- **1001:** Private
- **1111:** Reserved.

Values: 0000 (Unknown), 0001 (Telephony numbering E.164), 0011 (Data X.121), 0100 (Telex F.69), 1000 (National standard), 1001 (Private), 1111 (Reserved)

Default: 0000 (Unknown)

Override remote types and plans

Console	SNMP	Text-based Config
Override remote types and plans	ifwanOverrideRemote-TypesPlans	[ifwan#] OverrideRemote-TypesPlans

Determines whether the local settings for the numbering plan parameters will override the settings configured on the remote side when ISDN signaling is the same on both sides of the connection. These parameters include:

- *Calling number type of network*
- *Calling number numbering plan*
- *Called number type of network*
- *Called number numbering plan*

Set *Override remote types and plans* to **YES** to force the remote port to the local settings.

Values: NO, YES

Default: NO

Generate ring back locally

Console	SNMP	Text-based Config
Generate ring back locally	ifwanRingBackLocally	[ifwan#] RingBackLocally

The *Generate ring back locally* parameter is used in voice applications that require **Ring Back** at the local unit. For data applications, leave it at its default setting, **DISABLE**.

Values: DISABLE, ENABLE

Default: DISABLE

6.2 E1 Port

NOTE: E1 parameters that are identical to those on a T1 port are listed under "T1 Port" on page 6-2. For E1 parameters that have different values than the same parameter on a T1 port, the differences are noted in this section. E1 parameters that are unique to an E1 port are fully detailed in this section.

6.2.1 Signaling mode

NOTE: Refer to "Signaling mode" on page 6-4 for the SNMP and text-based configuration equivalents of this parameter.

For data transport on an E1 interface card the *Signaling mode* can be set to:

- **NONE:** data-only connection
- **TRSP-ORIG** (originate) or **TRSP-ANSW** (answer): Transparent HDLC-based or PCM64K-based transport

NOTE: **TRSP-ORIG** and **TRSP-ANSW** signaling modes are not available when the NetPerformer SIP VoIP licensed software option has been installed. They are used to establish a permanent point-to-point transparent signaling connection between digital interfaces (T1 or E1).

- **CAS:** data connection using Channel Associative Signaling (CAS)
- **EURO-ISDN:** data connection using ISDN-PRI (ETSI) in Europe
- **QSIG:** Standard PBX access with supplementary services support.

Values: NONE, TRSP-ORIG, TRSP-ANSW, CAS, EURO-ISDN, QSIG

Default: NONE

6.2.2 NONE Signaling Mode (on E1)

Pcm encoding law

NOTE: Refer to "Pcm encoding law" on page 6-5 for the SNMP and text-based configuration equivalents of this parameter.

The default value on an E1 port is **A-LAW** rather than **MU-LAW** (on T1 port).

Values: A-LAW, MU-LAW

Default: A-LAW

Idle code

NOTE: Refer to "Idle code" on page 6-6 for the SNMP and text-based configuration equivalents of this parameter.

The default value on an E1 port is **7E** rather than **7F** (on T1 port).

Values: OO - FF

Default: 7E

Zero suppression mode

NOTE: Refer to "Zero suppression mode" on page 6-6 for the SNMP and text-based configuration equivalents of this parameter.

The values for *Zero suppression mode* on an E1 port are different from those for the same parameter on a T1 port.

Values: HDB3, AMI

Default: HDB3

Gain limit

NOTE: Refer to "Gain limit" on page 6-7 for the SNMP and text-based configuration equivalents of this parameter.

The values for *Gain limit* on an E1 port are different from those for the same parameter on a T1 port.

Values: -12DB, -43DB

Default: -12DB

Impedance and Line Build Out

Console	SNMP	Text-based Config
Impedance and Line Build Out		[ifwan#]

For an E1/T1 interface set to E1 only.
Specifies the impedance on the interface, which determines the build-out to be applied on the transmitted DS1 signal.

Values: 120 OHMS, 75 OHMS

Default: 120 OHMS

CRC4 mode

Console	SNMP	Text-based Config
CRC4 mode	ifwanCrc4	[ifwan#] Crc4

Determines whether the CRC-4 procedure is enabled or disabled on the E1 port.

Values: DISABLE, ENABLE

Default: ENABLE

International bit

Console	SNMP	Text-based Config
International bit	ifwanT1E1InterBit	[ifwan#] ifwanT1E1InterBit

Determines whether the International Bit (I-bit) is set (**ENABLE**) or not set (**DISABLE**) on the E1 port.

Values: DISABLE, ENABLE

Default: ENABLE

ETS 300 011 mode

Console	SNMP	Text-based Config
ETS 300 011 mode	ifwanE1Ets300011	[ifwan#] E1Ets300011

For a legacy E1 interface only.
Determines whether *ETS 300 011 Mode* is enabled or disabled on the E1 port.

Values: DISABLE, ENABLE

Default: DISABLE

6.2.3 TRSP-ORIG and TRSP-ANSW Signaling Modes (on E1)

All parameters for **TRSP-ORIG** and **TRSP-ANSW** signaling modes are common to an E1 port with no signaling. Refer to "NONE Signaling Mode (on E1)" on page 6-16.

6.2.4 CAS Signaling Mode

All parameters for the **CAS** signaling mode are common to an E1 port with no signaling. Refer to "NONE Signaling Mode (on E1)" on page 6-16.

6.2.5 EURO-ISDN Signaling Mode

The following parameters for **EURO-ISDN** signaling mode on an E1 card are identical to the same parameters on a T1 port configured with a CCS *Signaling mode*, as detailed in "CCS Signaling Modes" on page 6-10:

- *CCS side*: see "CCS side" on page 6-10
- *Channel selection mode*: see "Channel selection mode" on page 6-11
- *Local number*: see "Local number" on page 6-12
- *Local subaddress*: see "Local subaddress" on page 6-12
- *Calling number type of network*: see "Calling number type of network" on page 6-13
- *Calling number numbering plan*: see "Calling number numbering plan" on page 6-13
- *Called number type of network*: see "Called number type of network" on page 6-14
- *Called number numbering plan*: see "Called number numbering plan" on page 6-14
- *Override remote types and plans:* see "Override remote types and plans" on page 6-15
- *Generate ring back locally*: see "Generate ring back locally" on page 6-15.

The other parameters for the **EURO-ISDN** signaling mode are common to an E1 port with no signaling. Refer to "NONE Signaling Mode (on E1)" on page 6-16.

6.3 ISDN-BRI S/T Port

NOTE: ISDN-BRI S/T parameters that are identical to those on an E1 port are listed under "E1 Port" on page 6-16. ISDN-BRI S/T parameters that are unique to an ISDN-BRI S/T port are fully detailed in this section.

6.3.1 Signaling mode

NOTE: Refer to "Signaling mode" on page 6-4 for the SNMP and text-based configuration equivalents of this parameter.

For data transport on an ISDN-BRI S/T interface card the *Signaling mode* can be set to:

- **NONE:** data-only connection
- **EURO-ISDN:** data connection using ISDN-BRI S/T in Europe
- **INS-NET** or **KDD:** ISDN-BRI S/T access in Japan
- **NI1**, **NI2**, **5ESS** or **DMS100:** ISDN-BRI S/T access in other countries
- **QSIG:** Standard PBX access with supplementary services support.

Values: NONE, EURO-ISDN, INS-NET, KDD, NI1, NI2, 5ESS, DMS100, QSIG

Default: NONE

6.3.2 NONE Signaling Mode (on ISDN-BRI S/T)

Most of the parameters for an ISDN-BRI S/T port with no signaling are identical to parameters configured on an E1 port configured with EURO-ISDN signaling (refer to "EURO-ISDN Signaling Mode" on page 6-19). The following parameters are unique to ISDN-BRI S/T ports.

Terminal Endpoint Identifier (TEI)

Console	SNMP	Text-based Config
Terminal Endpoint Identifier (TEI)	ifwanTeiModeOrValue	[ifwan#] TeiModeOrValue

Determines the mode or specific value of the Terminal Endpoint Identifier:

- **AUTOMATIC:** The TEI is automatically negotiated
- **0:** The TEI is ignored or disabled

- **1 to 63:** The TEI has a predefined value equal to the selected value.

Values: AUTOMATIC, 0 (disabled), 1 - 63

Default: AUTOMATIC

Power Mode

Console	SNMP	Text-based Config
Power Mode	ifwanPwrMode	[ifwan#] PwrMode

The NetPerformer can supply power through an NT port to external TE equipment. The *Power Mode* parameter determines the power feed mode that is used:

- **OFF:** No power is provided to the external TE device.
- **PHANTOM:** Fixed power is provided to the connected device using a standard I.430 Phantom supply.

NOTE: The *Power Mode* parameter is listed **only when the ISDN-BRI S/T port is set to NT termination mode**. If the ISDN-BRI S/T port is set to TE termination mode, the power source is automatically disabled by the hardware.

Values: OFF, PHANTOM

Default: OFF

6.3.3 CCS Signaling Modes

On an ISDN-BRI S/T card, the CCS signaling modes that can be used for digital data transport are: **EURO-ISDN**, **INS-NET**, **KDD**, **NI1**, **NI2**, **5ESS**, **DMS100** and **QSIG**.

All parameters for **EURO-ISDN** signaling are as for **EURO-ISDN** on an E1 card (see “EURO-ISDN Signaling Mode” on page 6-19) or no signaling on an ISDN-BRI S/T card (see “NONE Signaling Mode (on ISDN-BRI S/T)” on page 6-20).

Two additional parameters are required for configuration of an ISDN-BRI S/T port in **NI1**, **NI2**, **5ESS**, **DMS100** or QSIG signaling mode:

Local SPID 1

Console	SNMP	Text-based Config
Local SPID 1	ifwanLocalIsdnSpid1	[ifwan#] LocalIsdnSpid1

Defines the first Service Profile Identifier (SPID) that is included with the ISDN-BRI call setup message. It is required for a port with **NI1** signaling, and optional for a port with **NI2**, **5ESS** or **DMS100** signaling.

Values: Maximum 20 digits (0 to 9)

Default: no value

Local SPID 2

Console	SNMP	Text-based Config
Local SPID 2	ifwanLocalIsdnSpid2	[ifwan#] LocalIsdnSpid2

Defines the second SPID that is included with the ISDN-BRI call setup message. It is required for a port with **NI1** signaling, and optional for a port with **NI2**, **5ESS** or **DMS100** signaling.

Values: Maximum 20 digits (0 to 9)

Default: no value

7

SE/SLOT/#/CHANNEL Configuration Parameters

7.1 Common Parameters

The following parameters are required for **CHANNEL** configuration on all analog interface cards.

NOTE: Parameters that are required for **CHANNEL** configuration on a **digital interface card only** are detailed in the appendix *SE/SLOT/#/CHANNEL Configuration Parameters* of the *Digital Voice* fascicle of this document series.

7.1.1 Protocol

Console	SNMP	Text-based Config
Protocol	ifvceProtocol	[ifvce#] Protocol

Determines the operating protocol for this voice channel. The port protocol must be set to the same value on the local and remote NetPerformer voice ports.

The protocols that are available depend on the NetPerformer model, and whether you are configuring analog voice on an analog or digital interface card. To view which protocols are available on your unit, enter a question mark (**?**) after the command prompt for the *Protocol* parameter.

- **OFF:** Select this value when the port is not used.

NOTE: If you leave the Protocol parameter set to **OFF**, no other configuration parameters will be displayed for this voice port.

- **ACELP-CN:** ACELP Comfort Noise voice compression at 8 Kbps/6 Kbps with bad/lost packet interpolation

NOTE: This protocol provides a slightly higher quality of voice than ACELP8K, takes less bandwidth during silence, and has a packet pace that permits double and triple buffering to reduce the number of cells processed per second.

- **G.723:** A standards-based voice codec (G.723.1) designed for video conferencing and telephony over standard phone lines, with realtime encoding and decoding
- **G726 16K:** Adaptive Differential Pulse Code Modulation at 16 Kbps. Conforms to ITU-T Recommendation G.726
- **G726 24K:** G.726 at 24 Kbps

- **G726 32K:** G.726 at 32 Kbps
- **G726 40K:** G.726 at 40 Kbps
- **G.729:** CS-ACELP voice compression at 8 Kbps according to ITU-T Recommendation G.729. (Optional: Available only with SIP license activated on the unit.)
- **LDCD:** Low Delay Codec at 16 Kbps
- **PCM64K:** Pulse Code Modulation with non-linear compression at 64 Kbps

NOTE: If you select the PCM64K protocol, make sure that the bandwidth allocated to the link port is greater than 64 Kbps. Lower bandwidth levels will produce choppy voice quality.

Values: Analog interface cards: OFF, ACELP-CN, PCM64K

Digital interface cards: OFF, ACELP-CN, G.723, G726 16K, G726 24K, G726 32K, G726 40K, G.729, PCM64K

Default: :OFF

7.1.2 ACELP-CN Parameters

The following parameters are required when the *Protocol* is set to **ACELP-CN**.

DSP packets per frame
8K packetization selection (Y/N)

Console	SNMP	Text-based Config
DSP packets per frame 8K packetization selection (Y/N)	ifvceRate8kx1 ifvceRate8kx2 ifvceRate8kx3 ifvceRate8kx4	[ifvce#] Rate8kx1 Rate8kx2 Rate8kx3 Rate8kx4

For ACELP-CN only:
Sets the buffering scheme for 8K packetization, which determines how the bit rate is reduced when congestion occurs (fallback).

To set the buffering scheme for 8K packetization, enter **N** (no) or **Y** (yes) beneath the numbers **1** to **4** indicated for the *DSP packets per frame*.

Buffering adjusts the fallback options for this voice channel, for example:

- **YNNN:** Enables fallback to 8 Kbps with single buffering, or 1 packet per frame at 8 Kbps
- **NYNN:** Enables fallback to 8 Kbps with double buffering (2 packets per frame)
- **NNYN:** Enables fallback to 8 Kbps with triple buffering (3 packets per frame)

- **NNNY:** Enables fallback to 8 Kbps with quadruple buffering (4 packets per frame).

The buffering scheme can be set to any available combination, for example:

- **YYYN:** Enables fallback to 8 Kbps with single or double buffering.
  - When congestion is first detected the transmit rate changes from 8K single-buffered to 8K double-buffered
  - If congestion persists, the transmit rate will change to 8K triple-buffered
  - In other words, for each level of congestion that occurs the rate is reduced to the next lower fallback rate that is enabled (packetization selection set to **Y**)
  - As congestion levels are cleared, the rate is automatically increased to the next higher fallback rate that is enabled.

Values: 4-character string with N or Y in each position

Default: YNNN (single buffering)

DSP packets per frame 6K packetization selection (Y/N)

Console	SNMP	Text-based Config
DSP packets per frame 6K packetization selection (Y/N)	ifvceRate6kx1 ifvceRate6kx2 ifvceRate6kx3 ifvceRate6kx4 ifvceRate6kx5	[ifvce#] Rate6kx1 Rate6kx2 Rate6kx3 Rate6kx4 Rate6kx5

For ACELP-CN only:
Sets the buffering scheme for 6K packetization, which determines how the bit rate is reduced during transmission of multi-frequency tones, signaling tones and background noise, and how the port will operate when congestion occurs.

To set the buffering scheme for 6K packetization, enter **N** (no) or **Y** (yes) beneath the numbers **1** to **5** indicated for the *DSP packets per frame*.

Buffering adjusts the fallback options for this voice channel, for example:

- **YNNNN:** Enables fallback to 6 Kbps with single buffering, or 1 packet per frame at 6 Kbps
- **NYNNN:** Enables fallback to 6 Kbps with double buffering (2 packets per frame)
- **NNYNN:** Enables fallback to 6 Kbps with triple buffering (3 packets per frame)
- **NNNYN:** Enables fallback to 6 Kbps with quadruple buffering (4 packets per frame)
- **NNNNY:** Enables fallback to 6 Kbps with quintuple buffering (5 packets per frame).

The 6K buffering scheme is used in conjunction with the 8K buffering scheme, for example:

- 8K buffering: **YYNN**
- 6K buffering: **NYNNN**
  - When congestion is first detected the transmit rate changes from 8K single-buffered to 8K double-buffered
  - If congestion persists, the transmit rate will change to 6K double-buffered
  - In other words, for each level of congestion that occurs the rate is reduced to the next lower fallback rate that is enabled (packetization selection set to **Y**)
  - As congestion levels are cleared, the rate is automatically increased to the next higher fallback rate that is enabled.

Values: 5-character string with N or Y in each position

Default: NNNNN (no buffering)

Comfort noise level

Console	SNMP	Text-based Config
Comfort noise level	ifvceComfortNoiseLevel	[ifvce#] ComfortNoise-Level

For ACELP-CN only:
Determines the level of background noise that is generated for a voice call on this channel. During silent periods, the comfort noise ensures the listener that the line is not dead.

Values: 0 - 10

Default: 0

7.1.3 PCM/ADPCM/G729 Parameters

The following parameter is required when the *Protocol* is set to **PCM64K**, **G726 16K**, **G726 24K**, **G726 32K**, **G726 40K**, **G729** or **G729A**.

Silence suppression level

Console	SNMP	Text-based Config
Silence suppression level	ifvceSilenceSuppress	[ifvce#] SilenceSuppress

For PCM, ADPCM and G729/A only:
Specifies the degree to which periods of silence are suppressed and reduced during transmissions. Its value determines the sensitivity at which silence is detected. A higher Silence Suppression level increases the attenuation of the line, lowers background noise and reduces overall bandwidth use. The lowest value, **1**, disables silence suppression.

Values: 1 - 5

Default: 1

7.1.4 Other Parameters Common to All Protocols

The following parameters are required for voice channels on all analog interface cards when the *Protocol* is set to **ACELP-CN**, **PCM64K**, **G723**, **G726 16K**, **G726 24K**, **G726 32K**, **G726 40K**, **G729** or **G729A**.

Local inbound voice level (db)

Console	SNMP	Text-based Config
Local inbound voice level (db)	ifvceLocalInbound	[ifvce#] LocalInbound

Specifies the local voice level going into the port, measured in 1 dB increments. The value of this parameter determines how sensitive the local voice channel will be to the signal from the attached device. The lower the value, the more sensitive the voice channel is to the input, and the louder the voice output at the remote end will sound.

Ideally, this parameter should match the level of the input signal. A more negative setting produces a higher input gain, the highest input gain being delivered when this parameter is set to **-22**.

Values: -22 - 8

Default: 0

Local outbound voice level (db)

Console	SNMP	Text-based Config
Local outbound voice level (db)	ifvceLocalOutbound	[ifvce#] LocalOutbound

Specifies the local voice level going out of the port, measured in 1 dB increments. The higher the value, the louder the volume will be.

Values: -22 - 8

Default: -3

Priority Level

Console	SNMP	Text-based Config
Priority Level	ifvcePriorityLevel	[ifvce#] PriorityLevel

Specifies the priority level of this voice channel. This can be used to ensure that the highest priority voice calls are established. If the maximum number of voice calls allowed on the WAN (PVCR) link has been reached, lower priority calls will be disconnected to permit connection of more high-priority calls.

The *Priority Level* parameter can be set from **0** to **10**, where **0** represents the lowest priority and **10** the highest priority. Voice channels with a higher priority assignment take precedence over channels with a lower priority assignment.

- Voice channels can be limited only when a priority greater than **0** is assigned to several voice channels on the same unit
- By default, all voice channels have a *Priority Level* of **0**, which means voice connections are not established according to priority.

NOTE: The maximum number of voice channels that can be established over a link is defined using the PVCR link parameter *Maximum number of voice channels*. To make good use of the voice channel priority feature, the number of voice channels set with high priority should not exceed the value of this parameter.

Examples:

- The NetPerformer receives a high priority call when the *Maximum number of voice channels* has already been reached. The lowest priority active voice channel is dropped to permit connection of the higher priority voice channel.
- The NetPerformer receives a call when the *Maximum number of voice channels* has already been reached. The requested voice connection has a priority level lower than or equal to that of the currently active voice channels. In this case the incoming call is refused and a busy signal is generated.

Values: 0 - 10

Default: 0

Echo canceler

Console	SNMP	Text-based Config
Echo canceler	ifvceEchoCanceler	[ifvce#] EchoCanceler

Determines whether echo cancellation is used on this voice channel to prevent double talk.

Values: DISABLE, ENABLE

Default: ENABLE

Double talk threshold (db)

Console	SNMP	Text-based Config
Double talk threshold (db)	ifvceDTalkThreshold	[ifvce#] DTalkThreshold

Specifies the echo cancellation threshold, measured in 1 dB increments.

Values: -12 - 12

Default: 6

Pulse frequency (pps)

Console	SNMP	Text-based Config
Pulse frequency (pps)	ifvceDialPulseFrequency	[ifvce#] DialPulseFrequency

Specifies the pulse frequency in pulses per second (pps). This frequency is used for detection purposes.

Values: 10, 20

Default: 10

Activation type

Console	SNMP	Text-based Config
Activation type	ifvceActivationType	[ifvce#] ActivationType

Determines how the voice channel is activated:

- **PREDEFINED:** The destination unit and port number are preconfigured, using the *Remote unit* and *Remote port number* parameters (described on "PREDEFINED Activation Type Parameters" on page 7-9). As soon as the device connected to the voice channel goes off-hook, the NetPerformer begins a calling procedure with the device at the other site.

  This creates a dedicated connection. The two voice channels linked through predefined line activation cannot be accessed by any other voice channel in the network.

- **SWITCHED:** The NetPerformer selects the remote location according to a configurable *Speed dial number* that the user enters into the telephone set.

  All speed dial numbers are defined in the Voice Mapping Table along with the associated destination unit, extension number and optional dialing sequence to be forwarded to the attached voice equipment.

  No predetermined connection is set up between any two ports.

- **AUTODIAL:** Autodial line activation behaves like a switch that always dials to the same remote unit or set of remote units. The NetPerformer reaches the remote location using a predefined number. This number is permanently configured for the voice port, and does not need to be manually entered.

  NetPerformer begins the calling procedure with the remote site as soon as the device connected to the voice channel goes off-hook.

  Unlike predefined line activation, inward dialing is allowed on a voice channel configured for **AUTODIAL** line activation. An **AUTODIAL** channel is accessible from any other **SWITCHED** or **AUTODIAL** analog voice channel in the network.

- **BROADCAST:** Permits sending a single voice message to multiple destinations using a Frame Relay one-way multicast service. For broadcast activation, the local NetPerformer (or root) transmits the broadcast frames via a special Broadcast PVC (the Mdlci) to a multicast server. The multicast server then distributes the frames via PVCR PVCs to each remote NetPerformer (or leaf). PVCR PVCs must also be defined for each direct path between root and leaf.

The **PREDEFINED** and **BROADCAST** line activation types are not available on a NetPerformer installed with the SIP VoIP licensed software option.

Values: PREDEFINED, SWITCHED, AUTODIAL, BROADCAST

Default: PREDEFINED

7.1.5 PREDEFINED Activation Type Parameters

The following parameters are required when the *Activation type* is set to **PREDEFINED**.

Remote unit

Console	SNMP	Text-based Config
Remote unit	ifvceRemoteUnit	[ifvce#] RemoteUnit

For PREDEFINED Activation only
Specifies the NetPerformer at the remote site to which voice calls from this voice channel will be directed. Enter the *Unit name* of the remote NetPerformer unit.

NOTE: The *Unit name* is defined on the remote unit using the **SETUP/GLOBAL** sub-menu. Refer to the appendix *SE/GLOBAL Configuration Parameters* in the *Quick Configuration* fascicle of this document series.

Values: Maximum 32-character alphanumeric string

Default: NONE

Remote port number

Console	SNMP	Text-based Config
Remote port number	ifvceRemotePort	[ifvce#] RemotePort

For PREDEFINED Activation only
Specifies the voice channel on the remote NetPerformer to which voice calls from this channel will be directed. Enter the number of the voice channel that is connected to the device you want to reach.

Values: 1 - 65534

Default: the local voice channel number

7.1.6 SWITCHED Activation Type Parameters

The following parameters are required when the *Activation type* is set to **SWITCHED**.

Hunt Group active

Console	SNMP	Text-based Config
Hunt Group active	ifvceHuntGroup	[ifvce#] HuntGroup

For SWITCHED or AUTODIAL Activation only
The NetPerformer can hunt more than one voice channel to place an incoming call, using the *Hunt Forwarding* feature. The *Hunt Group active* parameter determines to which Hunt Group this voice channel belongs.

When a switched or autodial call comes in for a particular Hunt Group, the NetPerformer will attempt to connect the call to a voice channel with that *Hunt Group active*, starting with the oldest unused port. To allow this, a Hunt Group must be targeted in the Voice Mapping Table entry associated with the call, using the *Destination extension source* and *Hunt group* parameters.

For details, consult the *Hunt Forwarding* chapter in the *Advanced Voice Features* fascicle of this document series.

Values: **A**, **B**, **C**, **D**, **E**, **F**, **NONE**

Default: NONE

Delete digits

Console	SNMP	Text-based Config
Delete digits	ifvceDelDigits	[ifvce#] DelDigits

For SWITCHED Activation only
Specifies the number of leading dial digits, if any, that will be deleted from a dial string before it is forwarded to the attached voice equipment. This parameter serves to delete the leading dial digits that may be inserted by an attached PBX.

For example, a PBX may insert a **9** prefix in any dial string. If you dial **1234**, the result would be **91234**. To forward the correct dial string to the remote voice equipment, the local NetPerformer must be able to delete the first digit of the string. In this case, the *Delete digits* parameter on the local voice channel should be set to **1**.

NOTE: When *Delete digits* is set to **0**, no dial digits are deleted.

Values: 0 - 4

Default: 0

Port extension number

Console	SNMP	Text-based Config
Port extension number	ifvceExtNumber	[ifvce#] ExtNumber

For SWITCHED Activation only
Specifies the extension number for the end device attached to this voice channel. If the *Hunt Group active* parameter is set to **NONE** for this port, a connection will be attempted on this port only.

All extension numbers in the network must contain the same number of digits, to ensure correct parsing of the dial digit sequence.

- The length of the *Port extension number* is determined by the *Extension number (no. of digits)* parameter of the **SETUP/GLOBAL** menu
- The default length is set to 3 digits. The default value is the local voice channel number
- You must enter the correct number of digits, as specified by the *Extension number (no. of digits)* parameter
- If you change the value of the *Port extension number*, at least one digit must be non-zero. That is, values such as **00**, **000** and **0000** are not permitted.

NOTE: Wildcard characters cannot be used when configuring or dialing a port extension number. To have the remote NetPerformer attempt more than one port when placing a call, configure *Hunt Group active* at the remote site, and set the *Hunt group* (in the **SETUP/MAP** menu) at the local site.

Values: 0 - 9 for each digit; number of digits determined by the Global *Extension number (no. of digits)* parameter

Default: the local voice channel number

Fwd digits

Console	SNMP	Text-based Config
Fwd digits	ifvceFwdDigits	[ifvce#] FwdDigits

For SWITCHED Activation only
Specifies which dial digits (if any) should be forwarded from the remote unit to the destination device.

- **NONE:** No dial digits are forwarded to the destination device when a call is initiated on this channel.
- **ALL:** The speed dial number and its associated extended digits are forwarded to the attached equipment. The extended digits may be specified in the Voice Mapping Table or manually dialed by the user.

- **EXT:** Only the extended digits string is forwarded to the destination device.

  The **EXT** value is not available when the NetPerformer is installed with the SIP VoIP licensed software option.

Use the *Fwd digits* parameter when the NetPerformer voice channel connects to the trunk side of a PBX or a CO. If an automated answering system responds to the call, the PBX or CO can dial the telephone number to connect to a device on the station side.

NOTE: This parameter does not influence the digits that are sent from the local NetPerformer to the remote NetPerformer. Both the speed dial number and any associated extended digits are sent to the remote site at all times.

Values: NONE, ALL, EXT

Default: NONE

If the Fwd digits parameter is set to **ALL** or **EXT**, configuration of the following two parameters is also required.

Fwd type

Console	SNMP	Text-based Config
Fwd type	ifvceFwdType	[ifvce#] FwdType

For SWITCHED Activation only

Determines how the dial digits are sent to the remote unit for forwarding to the attached device, such as a PBX. They can be sent using pulse dial or Tone. Select **PULSE** or **TONE** according to the requirements of the destination user equipment.

NOTE: If you set the Forward Type parameter to **PULSE**, ensure that the dial digit string does not contain an asterisk (*) or pound sign (#). These characters cannot be generated as pulse digits.

Values: TONE, PULSE

Default: TONE

Fwd delay (ms)

Console	SNMP	Text-based Config
Fwd delay (ms)	ifvceFwdDelay-ms	[ifvce#] FwdDelay-ms

For SWITCHED Activation only

Specifies the length, in milliseconds, of a pause that precedes the forwarded dial digit string.

- Set this parameter to a non-zero value if the remote PBX requires a delay before forwarding the telephone number to the station side.
- When set to **0**, no pause is made.
- If you enter the delay with a `set` command, the value is rounded down to the nearest multiple of 250 ms.

The *Fwd delay (ms)* parameter also determines the length of a pause that is inserted in the extended digits string using the pause character (,). When this character is encountered, the forwarding NetPerformer will pause for the length of time specified by *Fwd delay (ms)* before forwarding additional extended digits.

Values: **0 - 10000**, in increments of 250

Default: 0

7.1.7 AUTODIAL Activation Type Parameters

The following parameters are required when the *Activation type* is set to **AUTODIAL**.

Speed dial number

Console	SNMP	Text-based Config
Speed dial number	ifvceSpeedDialNum	[ifvce#] SpeedDialNum

For AUTODIAL Activation only

Specifies which speed dial number will be dialed when a off-hook condition occurs on this voice channel. Select a valid speed dial number from the Voice Mapping Table.

To view a list of speed dial numbers, execute the Display Map File (**DMF**) command from the NetPerformer console command line.

Values: 0 - 9, * for each digit, determined by Voice Mapping Table entries

Default: NONE

NOTE: The *Hunt Group active* parameter is also requested when the *Activation type* is set to **AUTODIAL**. Refer to “Hunt Group active” on page 7-10.

7.1.8 BROADCAST Activation Type Parameters

The following parameters are required when the *Activation type* is set to **BROADCAST**.

Broadcast direction

Console	SNMP	Text-based Config
Broadcast direction	ifvceBroadcastDir	[ifvce#] BroadcastDir

For BROADCAST Activation only
Specifies whether this voice channel will transmit (**TX**) or receive (**RX**) broadcast messages. Set the voice channel on the root NetPerformer to **TX**, and those on the leaf NetPerformer to **RX**.

Values: RX, TX

Default: RX

PVC number

Console	SNMP	Text-based Config
PVC number	ifvceBroadcastPvc	[ifvce#] BroadcastPvc

For BROADCAST Activation only
Specifies which PVC will be used for transmitting (on the root) or receiving (on the leaves) the broadcast frames.

- On a root NetPerformer, enter the number of the PVC that is defined in BROAD-CAST mode.
- On a leaf NetPerformer, enter the number of the PVCR PVC that is included in the multicast group. This is the PVC that the multicast server will use to send the broadcast frames to this NetPerformer.

Values: 1 - 300

Default: 1

7.1.9 Other Parameters Common to All Activation Types

The following parameters are required for all analog interface cards and for all *Activation types*.

Link down busy

Console	SNMP	Text-based Config
Link down busy	ifvceLinkDwnBusy	[ifvce#] LinkDwnBusy

Enables (**YES**) or disables (**NO**) an automatic *busy out* condition on this voice port when no link is available. A broadcast link down setting (**BROADCAST**) is also available.

The overall effect of *Link down busy* is that when no link is available, the NetPerformer:

- Sends a **Link Down** indication to all voice channels
- Seizes the voice channels (busy out condition), and

- Generates a fast busy tone.

If an alternate route can be found or the link comes back up, the NetPerformer stops the busy out condition, and the voice channel becomes available for calls. To stop the fast busy tone you have to do an **ON HOOK/OFF HOOK** sequence.

NOTE: Link Down Busy must be set to **NO** on all voice ports that may use SVCs to reach their destination.

If a PBX is connected to the NetPerformer using CAS signaling on a T1 or E1 line, and if the T1/E1 line goes down, the NetPerformer cannot complete calls to any destination. When set to **YES**, Link Down Busy will take effect if no other link to another NetPerformer is available. The NetPerformer will raise the T1/E1 line and busy out all timeslots at once, including inactive voice ports. The PBX learns that all voice ports are busy, so it can immediately reroute the call on an alternate path. CAS signaling is addressed in the *Digital Voice* fascicle of this document series.

Effects of Link Down Busy on Call Progress

If *Link down busy* is set to **YES** and the link goes down during an active call:

- A fast busy tone is generated when the link goes down
- If all the links are down, all voice ports are advised with a **Link Down** indication, and fall into Link Down Busy state. If not all links are down, only **PREDEFINED** or **AUTODIAL** ports that use this link will be advised with a **Link Down** indication, and fall into Link Down Busy state
- After 5 seconds, the NetPerformer verifies if another link is available
- If another link is available, the NetPerformer sends a **Link Up** indication to all DSPs in Busy handler state that could use that link to reach the destination. (Predefined or autodial ports that do not use the alternate link will not receive a **Link Up** indication.)
- The user must hang up to stop the fast busy tone. A dial tone is restored after an **ON HOOK/OFF HOOK** sequence
- The user can then redial
- If no other link is available, no **Link Up** indication is sent to the DSP, and the fast busy tone continues even after an **ON HOOK/OFF HOOK** sequence.

If *Link down busy* is set to **YES** and a new call is attempted while the link is down:

- A dial tone is generated when the user picks up the phone
- The user dials the number, and the NetPerformer sends a Connect Request to try to reach the destination
- Since the link is down, the NetPerformer receives a **No Destination** indication

- The NetPerformer generates a fast busy tone
- The user must hang up
- After 5 seconds, the NetPerformer verifies if another link is available (a port falls into Link Down Busy state for at least 5 seconds)
- If another link is available, the NetPerformer sends a **Link Up** indication to the DSP
- Once a link becomes available, the dial tone will be restored after an **ON HOOK/ OFF HOOK** sequence
- The user can then redial.

NOTE: If the link is connected to a PBX that can route calls to an alternate source based on an *all busy* condition, the PBX will be able to recognize this condition while the link is down and reroute any calls (when *Link down busy* is set to **YES**).

If *Link down busy* is set to **NO** and the link goes down during an active call:

- No tone is generated. There is silence on the line for as long as the link is down
- If the link comes back up within 10 seconds (before the DSP timeout) the call is reinstated, as long as the user did not hang up
- If the link is down for more than 10 seconds, the NetPerformer generates a normal busy tone
- The user must hang up to stop the busy tone. A dial tone is restored after an **ON HOOK/OFF HOOK** sequence
- The user can then redial.

If *Link down busy* is set to **NO** and a new call is attempted while the link is down:

- A dial tone is generated when the user picks up the phone
- The user dials the number, and the NetPerformer sends a Connect Request to try to reach the destination
- Since the link is down, the NetPerformer receives a **No Destination** indication
- The NetPerformer generates a normal busy tone
- The user must hang up. A dial tone is restored after an **ON HOOK/OFF HOOK** sequence
- The user can then redial.

NOTE: When Link down busy is deactivated (set to **NO**) there is no fast busy tone, the link is never seized, and you can always dial out.

On analog voice/fax ports there is no busy tone when the Country Code is equal to 100. Code 100, used in Korea, silences the busy tone on an FXS interface. The #6 DTMF tone is generated in the busy state every 30 seconds, regardless of the configuration.

Link Down Busy and NetPerformer Boot-up:

When *Link down busy* is set to **YES** and the NetPerformer is booted up, the response of the voice port depends on its activation type:

- **PREDEFINED:** When the NetPerformer boots up, predefined ports will initially be unable to reach the remote side, since no link is available for the first few seconds. A higher level task determines whether the destination is known. Once the destination is recognized, the port is no longer busied out, or seized. A link is now available, and the normal calling procedure can be carried out to the destination.
- **SWITCHED:** For a switched port, the NetPerformer searches the Voice Mapping Table for the unit names of all valid destinations. As soon as one of these destinations is recognized and becomes available, the port is no longer seized. Thus the NetPerformer can attempt a call as soon as one accessible and valid remote site is found. Note that a remote site may be made accessible via a data connection.
- **AUTODIAL:** For an autodial port, the NetPerformer searches the Voice Mapping Table for the name of the destination unit. If this unit is recognized, the port is no longer seized and a calling procedure can begin. If the destination is inaccessible, the port is busied out. When *Link down busy* is activated on an autodial port a Voice Mapping Table entry with the corresponding Speed Dial Number must exist. If no entry has been configured, or if the entry is invalid, the voice port will remain indefinitely in a Link Down Busy state.
- **BROADCAST**: When a broadcast voice port is first opened, the value of the *Link down busy* parameter is ignored, and the port goes immediately into **IDLE** state. The subsequent behavior of a broadcast voice port when a link goes down depends on the value of the Country Code.

Viewing the Link Down Busy State

To see whether a fast busy tone is due to a Link Down Busy situation, execute the Display Call States (**DCS**) command from the console command line. The status message **NONE NO LINK** is displayed for a voice port that is not accessible due to Link Down Busy. This means that there is no possible link available to reach the destination, the port is busied out and the attached equipment cannot process a call.

Values: NO, YES, BROADCAST

Default: NO

TONE type

Console	SNMP	Text-based Config
TONE type	ifvceToneType	[ifvce#] ToneType

Specifies the kind of multi-frequency tones that will be detected on this voice channel, including those used for call setup. Choose between Dual Tone Multi-Frequency (**DTMF**) and Multi-Frequency (**MF**) tones.

Values: DTMF, MF

Default: DTMF

TONE regeneration

Console	SNMP	Text-based Config
TONE regeneration	ifvceToneDetectRegen-s	[ifvce#] ToneDetectRe-gen-s

Specifies the number of seconds during which the tones generated by the remote site user equipment will be detected and regenerated at the local site.

Tones that have been compressed and decompressed by a voice compression algorithm may become distorted. Distorted tones may not be recognized by voice switching equipment (such as a PBX), depending on the error margin of the equipment. The *TONE regeneration* parameter allows tones passed through a voice channel to be regenerated locally, thus avoiding the possibility of distortion.

- When the *TONE regeneration* parameter is set to **0**, the tone regenerator is always disabled and tones coming from the remote side through the voice channel are decompressed as voice
- When set to **1**, the tone regenerator is always on
- Values from **2** to **255** define a specific period, in seconds, during which the tone regenerator is enabled locally.

NOTE: The human voice can produce frequencies which may be interpreted as tones and then regenerated at the remote site if *TONE regeneration* is always on (set to **1**). To avoid tone regeneration during a conversation, it is recommended that this parameter be enabled for a limited amount of time (set to **2** to **255**).

Values: 0 - 255

Default: 1

TONE ON (ms)

Console	SNMP	Text-based Config
TONE ON (ms)	ifvceToneOn-ms	[ifvce#] ToneOn-ms

Specifies the duration, in milliseconds, of the multi-frequency tone the NetPerformer uses to generate a single digit.

Values: 30 - 1000 in increments of 10

Default: 100

TONE OFF (ms)

Console	SNMP	Text-based Config
TONE OFF (ms)	ifvceToneOff-ms	[ifvce#] ToneOff-ms

Specifies the duration, in milliseconds, of the silence the NetPerformer adds between multi-frequency tones when generating a dial digit string.

The actual duration of the multi-frequency tone depends to a great extent on the user's actions, especially for telephones that sound a tone for as long as a number key is pressed. People typically execute a tone duration of 100 - 750 ms, with silence periods of 300 - 1500 ms. Much shorter periods result from automated dialing: typically 60 - 120 ms for tones, with 50 - 150 ms of silence between tones.

Values: 30 - 1000 in increments of 10

Default: 100

Pulse make/break ratio

Console	SNMP	Text-based Config
Pulse make/break ratio	ifvcePulseMakeBreak-ms	[ifvce#] PulseMakeBreak-ms

Specifies the duration, in milliseconds, of the contact (or *make*) for each digit that is dialed using rotary type dialing. The NetPerformer calculates the break value by subtracting the make value from 100.

For example, the value **34** defines the make/break ratio as **34/66**. This value is used in North America, Belgium, Denmark, U.K., France, Portugal and other countries, and is the default *Pulse make/break ratio*.

When you rotate the dial to a number and release it, a break/contact sequence is repeated once for each digit the dial passes as it returns to the starting position. If, for instance, you dial the number **3**, what follows is a 66 ms break, a 34 ms make, a 66 ms break, a 34 ms make, a 66 ms break, then a continuous make (closed loop).

Telephone equipment manufacturers in other countries have implemented other make/break ratios. To ensure correct configuration of the Pulse Make/Break Ratio parameter for your site, contact your local telephone company for the make/break ratio that is currently in use.

Values: 30 - 50 in increments of 4

Default: 34

Fax relay

Console	SNMP	Text-based Config
Fax relay	ifvceFaxRelay	[ifvce#] FaxRelay

Enables or disables handling of fax calls on this voice channel.

- **NONE:** Fax connections are not detected. Calls are treated as voice only.

NOTE: If a voice call experiences excessive noise, the NetPerformer could interpret it as a fax call. To ensure that all calls will be treated as voice regardless of noise conditions, set this parameter to **NONE**.

- **FAX:** Both voice and fax calls are allowed. The NetPerformer will pass fax signals to the remote unit. The switch from voice to fax mode takes place automatically when a fax tone is detected.

NOTE: The *Fax relay* values are different when the NetPerformer is installed with the SIP VoIP licensed software option:

- **NONE:** Fax connections are not detected. Calls are treated as voice only.
- **T.38:** T.38 negotiation is carried out. A SIP **Invite** message will be sent in T.38 fax mode as well as any voice codecs configured for negotiation (typically G.711 for fax). The unit will accept a T.38 fax call or a SIP **Re-invite** with one of the codecs configured for negotiation.

  For example, if you set the **SIP/CODEC NEGO** parameters *G711 alaw* and *G711 ulaw* to **YES**, and set the *Fax relay* parameter on the voice channel to **T.38**, a SIP **Re-invite** message will be sent in T.38 fax mode as well as G711 alaw and G711 µlaw. Refer to the chapter *Codec Negotiation* in the *Voice over IP (VoIP) Option* fascicle of this document series.
- **T.38_SG3:** For support of Super G3 faxes. When the NetPerformer detects a Super G3 answering tone it forces a fallback to the G3 standard (at 14.4 Kbps). This permits establishing the connection using T.38 Fax Relay, which requires less bandwidth than Modem Passthru using G.711.

Values: NetPerformer base product: NONE, FAX

NetPerformer with SIP VoIP option: NONE, T.38, T.38_SG3

Default: NetPerformer base product: FAX

NetPerformer with SIP VoIP option: T.38

Maximum fax rate

Console	SNMP	Text-based Config
Maximum fax rate	ifvceMaxFaxRate	[ifvce#] MaxFaxRate

Determines the maximum speed of the fax connection in bits per second. Fax connections can be made at standard speeds from 2400 bps to 14.4 Kbps.

Values: 2400, 4800, 7200, 9600, 12000, 14400

Default: 14400

ECM mode

Console	SNMP	Text-based Config
ECM mode	ifvceFaxEcmMode	[ifvce#] FaxEcmMode

Determines whether Error Correction Mode (ECM) will be used for fax connections on this voice channel. Set to **ENABLE** for ECM mode.

Values: DISABLE, ENABLE

Default: DISABLE

Modem relay

Console	SNMP	Text-based Config
Modem relay	ifvceModemRelay	[ifvce#] ModemRelay

Enables or disables handling of modem calls on this voice channel, and permits the Modem Passthru function.

- **NONE:** Modem connections are not detected. Calls are treated as voice only.
- **MODEM:** Both voice and modem calls are allowed. The NetPerformer will pass modem signals to the remote unit. The switch from voice to modem mode takes place automatically when a modem tone is detected.
- **PASSTHRU:** For Modem Passthru, which allows a modem connection to be established without using compression, echo cancelling or any other DSP processing of the traffic stream. The modem signal is sampled using the PCM64K codec algorithm. Modem Passthru simplifies the traffic on a PCM64K connection, permitting higher modem connection speeds.

Values: NONE, MODEM, PASSTHRU

Default: NONE

If the *Modem relay* parameter is set to **MODEM**, the following parameter is also required.

Maximum modem rate

Console	SNMP	Text-based Config
Maximum modem rate	ifvceMaxModemRate	[ifvce#] MaxModemRate

Determines the maximum speed of the modem connection in bits per second. Modem connections can be made at standard speeds from 4800 bps to 14.4 Kbps.

Values: 4800, 7200, 9600, 12000,14400

Default: 14400

V.22 Modem relay

Console	SNMP	Text-based Config
Slot channel x / V22 Modem relay	ifvceV22ModemRelay	[ifvce #] V22ModemRelay

Description: V22 Modem relay mode to support Point Of Sale (POS) at 1200 and 2400 bps.

Values: NONE, V22 RELAY, V22+POS RELAY, POS RELAY; default value: NONE

V22 Maximum modem rate

Console	SNMP	Text-based Config
Slot channel x / V22 Maximum modem rate	ifvceV22MaxModemRate	[ifvce #] V22MaxModemRate

Description: V22 Maximum modem rate. This parameter is available only when the "V22 Modem relay" parameter is not set to a value different than NONE.

Values: 1200, 2400; default value: 1200

Enable DTMF Detection ON-TIME

Console	SNMP	Text-based Config
Enable DTMF Detection ON-TIME	ifvceEnableDtmfOnTime	[ifvce#] EnableDtmfOnTime

The NetPerformer provides transport of DTMF signals. The *Enable DTMF Detection ON-TIME* parameter determines whether the duration of DTMF ON can be used to filter unwanted DTMF tones during call progress. Set this parameter to **YES** for DTMF tone filtering.

Values: NO, YES

Default: NO

If the *Enable DTMF Detection ON-TIME* parameter is set to **YES**, the following parameter is also required.

DTMF ON-TIME duration (ms)

Console	SNMP	Text-based Config
DTMF ON-TIME duration (ms)	ifvceDtmfOnTime	[ifvce#] DtmfOnTime

This parameter specifies the DTMF ON duration, in milliseconds, for filtering unwanted DTMF tones during call progress. To be properly detected by the NetPerformer, DTMF signals must be **ON** for a duration longer than the value of this parameter.

Values: 20 - 50

Default: 35

Redundant channel

Console	SNMP	Text-based Config
Redundant channel	ifvceRedundantChannel	[ifvce#] RedundantChan-nel

Determines whether this channel will be used as a redundant link on a backup system. For details, consult the *Redundancy Option* fascicle of this document series.

NOTE: This parameter does not appear when the NetPerformer is installed with the SIP VoIP licensed software option, since PowerCell voice is not supported in SIP mode.

Values: NO, YES

Default: NO

Egress ANI operation mode

Console	SNMP	Text-based Config
Egress ANI operation mode	ifvceEgressANIMode	[ifvce#] EgressANIMode

During call setup, the NetPerformer can send locally defined egress ANI digits, or regenerate the ANI digits received from the calling unit. The *Egress ANI operation mode* parameter determines from which source the ANI digits are taken.

- **NONE:** The NetPerformer sends **only** those ANI digits that have been received from the calling unit
- **INSERT:** The NetPerformer inserts the egress ANI digits that are defined locally on the voice channel, but **only if** the remote unit did not send any ANI digits
- **ALWAYS:** The NetPerformer **always** sends the egress ANI digits that are defined locally on the voice channel, and ignores any ANI digits received from the calling unit.

Values: NONE, INSERT, ALWAYS

Default: NONE

Egress CHANNEL ANI digits

Console	SNMP	Text-based Config
Egress CHANNEL ANI digits	ifvceEgressChannelANI-Digits	[ifvce#] EgressChannelA-NIDigits

Specifies the ANI digits that will be sent during call setup when the *Egress ANI operation mode* requires locally defined egress ANI digits (**INSERT** or **ALWAYS** setting).

Values: Maximum 20-character alphanumeric string: 0-9, A-D, *, #

Default: no value

Ingress ANI operation mode

Console	SNMP	Text-based Config
Ingress ANI operation mode	ifvceIngressANIMode	[ifvce#] IngressANIMode

Ingress ANI digits can be defined at both ends of a NetPerformer connection. Typically, ingress ANI is defined at the remote location and is transported to the central site. However, it can operate in the other direction, depending on the direction of the call.

During call setup, the ingress ANI digits are transported from the site where the call originates to the site that receives the call, over the voice channels involved in the connection. The *Ingress ANI operation mode* parameter determines the source of the ANI digits that are sent with the call:

- **NONE:** The NetPerformer sends **only** those ANI digits that have been received from the Telco equipment
- **INSERT:** The NetPerformer inserts the ingress ANI digits that are defined locally on the voice channel, but **only if** the Telco equipment did not send any ANI digits
- **ALWAYS:** The NetPerformer **always** sends the ingress ANI digits that are defined locally on the voice channel, and ignores any ANI digits received from the Telco equipment.

NOTE: On a NetPerformer installed with the SIP VoIP licensed software option, another mode is available: **GATEWAY ID**, which inserts the gateway number in the **INVITE** message during call setup to define the source of the call.

Values: NONE, INSERT, ALWAYS

Default: NONE

Ingress CHANNEL ANI digits

Console	SNMP	Text-based Config
Ingress CHANNEL ANI digits	ifvceIngressChannelANI-Digits	[ifvce#] IngressChan-nelANIDigits

Specifies the ANI digits that will be sent during call setup when the *Ingress ANI operation mode* requires locally defined ingress ANI digits (**INSERT** or **ALWAYS** setting).

Values: Maximum 20-character alphanumeric string: 0-9, A-D, *, #

Default: no value

7.2 FXS Channel Parameters

In addition to the common parameters described earlier, the following parameters are required for configuration of an FXS channel.

Country settings

Console	SNMP	Text-based Config
Country settings	ifvceRingTypePhoenix	[ifvce#] RingTypePhoenix

For an FXS or FXO channel

Determines the ring cadence that will be generated from this voice channel, according to the standards used in various countries. Select the country where the attached telephony device is located.

On an FXS channel you can also set this parameter to **CUSTOM**, which allows you to customize the ring. If you select this value you will be prompted for all customizable settings.

NOTE: On legacy NetPerformer products this parameter is referred to as *Ring type* (SNMP: *ifvceRingType*), and has a shorter list of countries. At the console, enter a question mark (**?**) after the parameter to view the available choices.

Values: ARGENTINA, AUSTRALIA, AUSTRIA, BAHRAIN, BELGIUM, BRAZIL, BULGARIA, CANADA, CHILE, CHINA, COLOMBIA, CROATIA, CYPRUS, CZECH REPUBLIC, DENMARK, ESCUADOR, EGYPT, EL SALVADOR, FINLAND, FRANCE, GERMANY, GREECE, HONG KONG, HUNGARY, ICELAND, INDIA, INDONESIA, IRELAND, ISRAEL, ITALY, JAPAN, JORDAN, KAZAKHSTAN, KUWAIT, LATVIA, LEBANON, LUXEMBOURG, MACAO, MALAYSIA, MALTA, MEXICO, MOROCCO, NETHERLANDS, NEW ZEALAND, NIGERIA, NORWAY, OMAN, PAKISTAN, PERU, PHILIPPINES, POLAND, PORTUGAL, ROMANIA, RUSSIA, SAUDI ARABIA, SINGAPORE, SLOVAKIA, SLOVENIA, SOUTH AFRICA, SOUTH KOREA, SPAIN, SWEDEN, SWITZERLAND, SYRIA, TAIWAN, THAILAND, UNITED ARAB EMIRATES, UK, USA, YEMEN, CUSTOM

Default: USA

Caller ID (ANI) transmission protocol

Console	SNMP	Text-based Config
Caller ID (ANI) transmission protocol	ifvceAnalogCallerID	[ifvce#] AnalogCallerID

For an FXS channel only

Determines the protocol used for retransmitting the Caller ID (ANI) received from a remote unit. This feature is available on the **SDM-9220 and SDM-9230 only**.

- **Bell 202:** The voice channel uses *Bell 202* tone modulation at 1200 baud to send the Caller ID. This is the best choice for a unit located in North America, Australia, China, Hong Kong or Singapore.
- **V23:** Uses *CCITT V23* modem tones to send the Caller ID. This is the best choice for a unit located in Europe.
- **OFF:** No caller ID is transported over the FXS channel.

NOTE: **OFF** is the default value. **You must change this value to allow the FXS channel to retransmit the caller ID.**

Values: OFF, BELL 202, V23

Default: OFF

Billing signals

Console	SNMP	Text-based Config
Billing signals	ifvceAnalogBillingTones	[ifvce#] AnalogBilling-Tones

For an FXS channel only

Determines the conditions for generating billing signals on this FXS channel.

- **EGRESS:** Billing signals are generated if this channel **places a call** to the phone that is plugged into it
- **INGRESS:** Billing signals are generated if this channel **receives a call** from the phone that is plugged into it
- **BOTH ENDS:** Billing signals are generated if this channel **either places or receives a call** over the phone that is plugged into it.
- **DISABLE:** No billing signals are generated on this channel.

Values: DISABLE, INGRESS, EGRESS, BOTH ENDS

Default: INGRESS

First billing signal time (s)

Console	SNMP	Text-based Config
First billing signal time (s)	ifvceAnalogFirstBilling-ToneTime	[ifvce#] AnalogFirstBilling-ToneTime

For an FXS channel only

Sets the **delay**, in seconds, before the first billing signal is transmitted on this channel.

For example, if the first 3 minutes of a call are free of charge, set this parameter to **180** seconds.

Values: 0 - 600

Default: 1

Billing signal duration (ms)

Console	SNMP	Text-based Config
Billing signal duration (ms)	ifvceAnalogBillingTone-Duration	[ifvce#] AnalogBillingTone-Duration

For an FXS channel only
Sets the **duration**, in milliseconds, of each billing signal that is generated on this channel.

Values: 20 - 1000 in increments of 20

Default: 20

Billing signal intervals (s)

Console	SNMP	Text-based Config
Billing signal intervals (s)	ifvceAnalogBillingTone-Intervals	[ifvce#] AnalogBillingTo-neIntervals

For an FXS channel only
Sets the **wait time**, in seconds, between the billing signals that are generated on this channel.

Values: 0 - 600

Default: 1

7.3 FXO Channel Parameters

In addition to the common parameters described earlier, the following parameters are required for an FXO channel.

FXO seizure delay

Console	SNMP	Text-based Config
FXO seizure delay	ifvceFxoSeizureDelay	[ifvce#] FxoSeizureDelay

For an FXO channel only
This parameter determines whether a delay will be added during the call connection process to permit extended digit forwarding before the line is seized. Set *FXO seizure delay* to **ENABLE** if extended digit forwarding will be used.

Values: DISABLE, ENABLE

Default: ENABLE

FXO timeout (s)

Console	SNMP	Text-based Config
FXO timeout (s)	ifvceFxoTimeout-s	[ifvce#] FxoTimeout-s

For an FXO channel only
This parameter determines the maximum amount of time, in seconds, that the local voice channel will wait for a fax transmission to begin. If no fax tone occurs by the time the *FXO timeout* expires, the connection is terminated.

Values: 6 - 99

Default: 30

Impedance

Console	SNMP	Text-based Config
Impedance	ifvceImpedancePhoenix	[ifvce#] ImpedancePhoe-nix

For an FXO channel only
Determines the frequency and cadence of the ring that will be used on this voice channel. From the list below, select the value that is appropriate to your network. The default value, **COUNTRY SPECS COMPATIBLE**, means the impedance setting is determined from the current *Country settings* value (see "Country settings" on page 7-26).

NOTE: On legacy NetPerformer products, this parameter can be set to **DEFAULT** or any multiple of 25 from **500** to **1000**. The SNMP equivalent is *ifvceImpedance*.

Values: COUNTRY SPECS COMPATIBLE,
600 OHMS,
900 OHMS,
270 OHMS + (750 OHMS || 150 NF),
220 OHMS + (820 OHMS || 120 NF),
370 OHMS + (620 OHMS || 310 NF),
320 OHMS + (1050 OHMS || 230 NF),
370 OHMS + (820 OHMS || 110 NF),
275 OHMS + (780 OHMS || 150 NF),
120 OHMS + (820 OHMS || 110 NF),
350 OHMS + (1000 OHMS || 210 NF),
0 OHM + (900 OHMS || 30 NF),
600 OHMS + 2.16 UF,
900 OHMS + 1 UF,
900 OHMS + 2.16 UF,
600 OHMS + 1 UF,
GLOBAL COMPLEX IMPEDANCE

Default: COUNTRY SPECS COMPATIBLE

Caller ID (ANI) detection protocol

Console	SNMP	Text-based Config
Caller ID (ANI) detection protocol	ifvceAnalogCallerID	[ifvce#] AnalogCallerID

For an FXO channel only
Determines the protocol used for detecting the Caller ID (ANI) received from an FXS voice channel on a remote unit. Caller ID (ANI) can be detected on an FXO interface on the **SDM-9220 and SDM-9230 only**.

Select the same protocol as that configured on the FXS voice channel with the *Caller ID*

(ANI) transmission protocol (see “Caller ID (ANI) transmission protocol” on page 7-26).

- **Bell 202:** The voice channel uses *Bell 202* tone modulation at 1200 baud to detect the Caller ID. This is the best choice for a unit located in North America, Australia, China, Hong Kong or Singapore.
- **V23:** Uses *CCITT V23* modem tones to detect the Caller ID. This is the best choice for a unit located in Europe.
- **OFF:** The caller ID cannot be detected on the FXO channel.

NOTE: **OFF** is the default value. **You must change this value to allow the FXO channel to detect the caller ID.**

Values: OFF, BELL 202, V23

Default: OFF

NOTE: The *Country settings* (on the SDM-9220 or SDM-9230) or *Ring type* (on legacy products) parameter is also required for an FXO channel (see “Country settings” on page 7-26). However, the **CUSTOM** setting of this parameter is not available for FXO.

7.4 E&M Channel Parameters

The following parameters are required for configuration of an E&M channel only.

E&M signaling type

Console	SNMP	Text-based Config
E&M signaling type	ifvceSignaling	[ifvce#] Signaling

For an E&M channel only
Determines the type of signaling that is used on this voice channel.

- **IMMEDIATE START:** Transmission takes place immediately. This is the industry standard for E&M operation.
- **WINK START:** The NetPerformer waits for the attached PBX to request a dial register (the PBX raises its M-lead). When this occurs, the NetPerformer sends a dial register to the PBX. It then toggles the E-lead when the PBX indicates it is ready for dial digits (no dial tone is transmitted to the PBX). When the E-lead returns to its original state the PBX will transmit dial digits.
- **CUSTOM:** Manually-configured custom signaling used for communicating with non-standard equipment. If you select this value you will be prompted for all customizable settings.

Values: IMMEDIATE START, WINK START, CUSTOM

Default: IMMEDIATE START

Analog E&M type

Console	SNMP	Text-based Config
Analog E&M type	ifvceAnalogEmType	[ifvce#] AnalogEmType

For an E&M channel only
Selects a 2-wire (unbalanced) or 4-wire (balanced) E&M connection.

Values: 2 WIRE, 4 WIRE

Default: 4 WIRE

TE timer (s)

Console	SNMP	Text-based Config
TE timer (s)	ifvceTeTimer-s	[ifvce#] TeTimer-s

For an E&M channel only
Specifies the delay, in seconds, at which the E-lead follows the M-lead for Timed-E signaling.

Values: 0 - 255

Default: 0

Hoot & Holler application

Console	SNMP	Text-based Config
Hoot & Holler application	ifvceHootHoller	[ifvce#] HootHoller

For an E&M channel only

A *Hoot and Holler* connection is a permanent voice connection, that is, one that is always off-hook. The call is always considered up, no matter what signaling information is carried from the user equipment.

- To configure a Hoot and Holler connection on this voice channel, set the *Hoot & Holler application* parameter to **YES**. Predefined line activation will be used on both sides of the connection.
- When this parameter is set to **NO**, the line activation type is determined from the *Activation type* parameter on this voice channel (see "Activation type" on page 7-8).

Values: NO, YES

Default: NO

Push to Talk application

Console	SNMP	Text-based Config
Push to Talk application	ifvcePushToTalk	[ifvce#] PushToTalk

For an E&M channel only

Enables a *Push To Talk* (PTT) application on the E&M interface. PTT is used to key a radio from a remote mobile location to either a control station or another remote location.

In an application involving a fixed control station:

- For the E&M channel to the control station, set the *Push to Talk application* parameter to **PTT CONTROL**
- For the E&M channel to the remote location, set *Push to Talk application* to **PTT ANSWER**
- The M lead is permanently connected to Signal Ground at the **PTT ANSWER** side, so that the E&M channel is up at all times.

In an application involving two remote mobile locations:

- Set the *Push to Talk application* parameter for the E&M channels on **both sides** of the connection to **PTT CONTROL**.

Tip: As a general rule of thumb in any PTT application, if the PTT device has a control button, you can set the corresponding E&M channel to **PTT CONTROL**.

Set this parameter to **DISABLE** when Push To Talk is not required on this voice channel.

NOTE: Push To Talk is not available on the voice channel if the *Hoot & Holler application* parameter is set to **YES**.

Values: DISABLE, PTT CONTROL, PTT ANSWER

Default: DISABLE

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