Analog Voice
NetPerformer® System Reference

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NetPerformer Support of Analog Voice

1.1 Analog Equipment Connections Supported

A NetPerformer can be connected to any of the following using an analog interface:

- **PSTN**: Using an FXO interface, the NetPerformer offers compatibility with a Public Switched Telephone Network (PSTN)
  - The NetPerformer interfaces with the Central Office (CO) like a standard telephone set
  - It is able to detect a ring and generate off-hook and on-hook signals.
- **POTS line**: Using an FXS interface, the NetPerformer provides CO connections to standard telephone sets or facsimile machines.
- **KTS**: A Key Telephone System (KTS) unit is usually connected to multiple ports on the NetPerformer, using FXS interfaces
  - The NetPerformer behaves like a CO
  - It is able to select an extension and generate a ring.
- **PBX**: The NetPerformer provides the 2/4-wire E&M interfaces required for tie trunks to analog PBXs.
  - E&M channels perform routing functions which offload PBX processing requirements.
  - E&M Types I, II and V are supported
  - For station-side connection to a PBX an FXO interface is used, since the NetPerformer acts like a standard telephone

NOTE: NetPerformer connection to a digital PBX is accomplished using ISDN signaling. For further information, consult the *Digital Voice* fascicle of this document series.

1.2 Signaling Engine Technology and Analog Voice Protocols

The NetPerformer Signaling Engine expansion board is equipped with:

- An MPC860 processor, responsible for handling the signaling used to establish the voice connection
- Communication interface cards, either digital or analog, which provide the physical connection to external devices
- State-of-the-art Digital Signal Processors (DSPs), which process the voice traffic using the following voice codecs:
    - **ACELP-CN:** ACELP Comfort Noise (CN) at both 8 Kbps and 6 Kbps (refer to the next section). Available on both legacy NetPerformer products and the new NetPerformer product line, and can be used for interworking between the two.
    - **PCM64K:** Pulse Code Modulation (PCM) using the G.711 codec, available on both legacy and current NetPerformer products. Refer to "PCM" on page 1-4.
    - **G.723:** Designed for videoconferencing and telephony support using IP and POTS standards
    - **G.726:** Uses ADPCM transcoding to convert a PCM stream from a 16, 24, 32 or 40 Kbps channel
    - **G.729:** For coding of speech signals at 8 kbit/s Conjugate-Structure Algebraic Code-Excited Linear Prediction (CS-ACELP). (Optional: Available only when SIP license activated on the unit.)

NOTE: A detailed description of the Signaling Engine hardware, as well as installation procedures for the DSPs and interface cards, are provided in the *Hardware Installation Guide* for the specific NetPerformer product.

1.2.1 ACELP-CN

The NetPerformer uses the ACELP (Algebraic Code Excited Linear Prediction) Comfort Noise (ACELP-CN) voice compression algorithm, or codec for superior throughput and voice quality.

ACELP-CN is a toll quality dual-rate codec that maintains high-quality sound with a compression rate of 8 Kbps or 6 Kbps.

It is ideal for multiplexing applications, can handle DTMF (Dual Tone Multi-Frequency) codes and provides a low-cost solution to maintaining voice quality in high-traffic networks.

It also offers bad/lost packet interpolation, reduced bandwidth during silence, a packet pace that permits double and triple buffering and improved quality for high-pitched voices.

1.2.2 PCM

PCM (Pulse Code Modulation) technology is a compression technique based on scalar quantification of the voice stream. The analog voice signal is directly coded in binary format. Quantification may be uniform or non-uniform, depending on the application.

The PCM method was first defined in CCITT/ITU standard G.711. It is based on the modulation of coded pulses, and provides a throughput of 64 Kbps. After non-linear compression is applied, the amplitude of samples is quantified over 8 bits.

This technology was very popular in the past, due to its simplicity and the fact that it does not require highly powerful processors. On a NetPerformer voice channel G.711 is configured with the **PCM64K** protocol.

1.2.3 Modem Relay

NetPerformer voice/fax channels can also be used to connect a modem and pass the formatted data to another modem in the network. The modem relay function is included in the ACELP algorithm, and requires V.32bis modems. Enabling modem relay also enables the fax relay feature. When enabled the switch from voice to modem/fax is carried out automatically. Both fax and modem connections can be made at standard speed intervals from 4800 bps to 14.4 Kbps.

1.3 Digital Signal Processor (DSP) Functions

Analog-to-digital (A/D) conversion must be performed before an analog voice signal can be carried over a digital line. A DSP is a microprocessor designed to digitize and process voice signals. The NetPerformer DSP carries out digitization and compression algorithms while consuming very little bandwidth. It is also used to handle other features of digitized voice processing, such as variable bit rates and echo canceling.

Low-cost implementations of CELP-type compression algorithms in single-chip form became practical with the advent of the most recent generation of high-performance DSPs. The NetPerformer DSP and ACELP codec represent the latest advances in voice compression technology, and together provide very efficient voice compression.

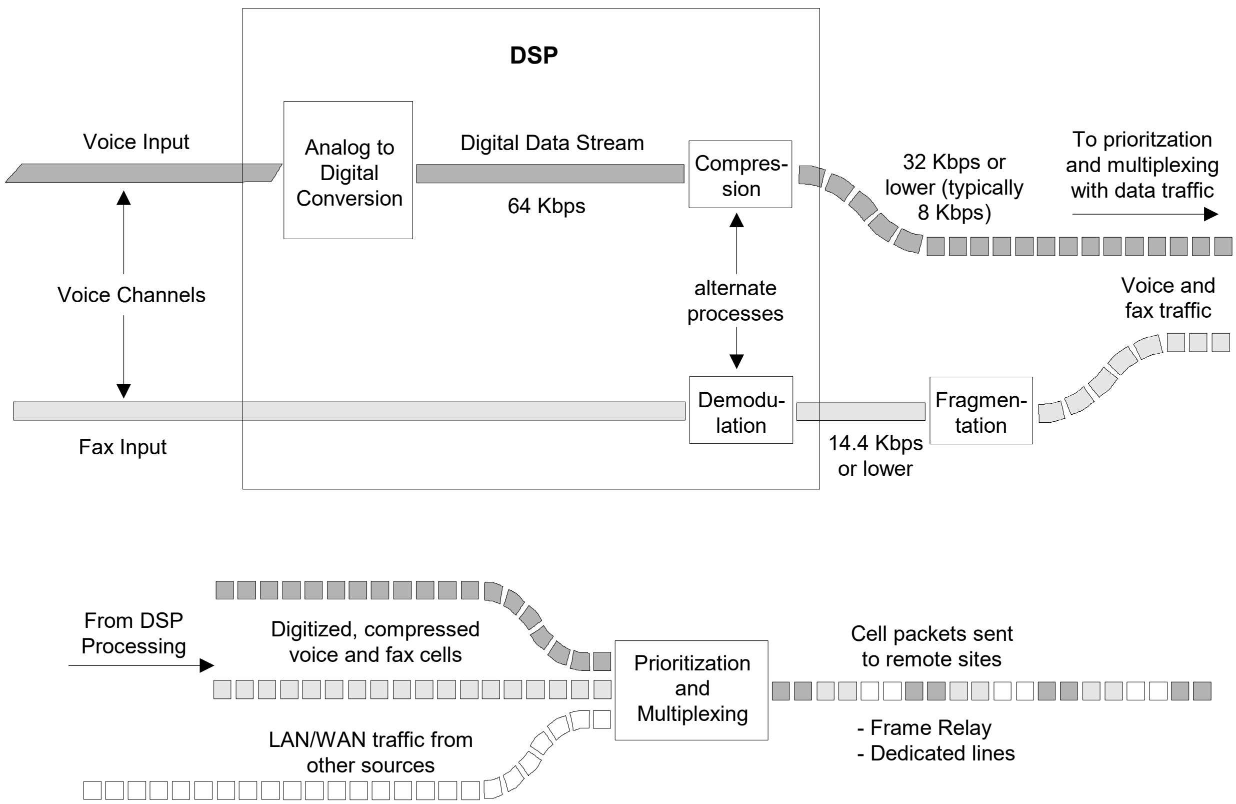


Figure 1-1: Processing of Voice/Fax Traffic

1.3.1 ACELP Compression/Decompression Procedure

- If the voice input is in analog format., the NetPerformer takes this analog source and converts it to a 64 Kbps digital stream in PCM (Pulse Code Modulation) format.
- Using ACELP-CN, for example, the DSP cuts the data into 20 ms cells.
- The DSP then analyzes the voice spectrum and compresses the digital stream to 8 Kbps using the ACELP-CN algorithm. This provides a compression ratio of 8:1.

- The NetPerformer then combines the compressed voice cells with data from different sources according to assigned priorities. By default, voice traffic is defined as high priority, since it is extremely delay-sensitive.
- The mixed traffic is then transmitted over the wide area network using NetPerformer Cell Relay technology.
- At the remote end these processes are reversed. The remote channel's DSP receives the compressed voice traffic and decompresses it to a 64 Kbps digital PCM stream.
- If the output is analog, the remote NetPerformer reconverts the PCM stream to analog format and sends it to the attached voice equipment.

1.3.2 Fax Demodulation

When a NetPerformer DSP detects a fax tone it stops compressing the voice stream and starts demodulating the fax stream.

- The NetPerformer demodulates the fax signals into HDLC (High-level Data Link Control) data at speeds of 14.4 Kbps or lower.
- The HDLC data is then fragmented into cells.
- The NetPerformer then combines the HDLC/fax cells with data from different sources according to assigned priorities. Fax, which is extremely delay-sensitive, is given high-priority status by default.
- The mixed traffic is then transmitted to remote sites using Cell Relay technology.
- At the remote end these processes are reversed. The remote channel receives the HDLC/fax cells and converts them to digital fax signals.
- The remote NetPerformer then reconverts the digital stream to analog format and sends it to the attached fax equipment.

Since audio transmission signals at 64 Kbps are converted to a digitized stream at 14.4 Kbps or lower, fax demodulation reduces the required bandwidth. Combined with the advantages of fragmentation and cell relay, the result is more efficient transport of fax signals, with reduced delays.

1.3.3 Variable Bit Rate

The NetPerformer's ACELP-CN compression algorithm produces 8 Kbps or 6 Kbps voice output. However, the NetPerformer can lower the bit rate even further depending on the nature of the voice stream. The speed is automatically reduced to a lower bit rate when:

- Signaling and DTMF tones are transmitted, **or**
- Silent periods occur. The reduced bandwidth is used to maintain background noise on the line (without some background noise, users perceive the line to be dead).

Voice communication is intrinsically half-duplex by nature: when one person speaks at one end of the line, the person at the other end listens. Pauses may also occur during speech, for example, between sentences, when the speaker leaves the phone, or when the speaker puts the listener on hold. The NetPerformer detects these silence periods. Its variable bit rate takes advantage of the fact that noise and DTMF signaling tones require less bandwidth than the voice traffic itself. It can then allocate the bandwidth saved from a silent or signaling voice channel to channels that are processing voice traffic. The result of a variable bit rate is optimized bandwidth utilization, improved system performance and a reduction of overall network costs.

1.3.4 Echo Canceling

Echo is caused by impedance mismatches on the telephone circuit. It is a distorted and delayed replica of the incoming speech from the remote end.

The NetPerformer has an echo canceler built into its DSP. Should echo occur on a speech or non-speech signal (such as voice-band data or fax) the echo cancelled automatically reduces this echo to tolerable levels. It adapts automatically to changes in the echo that may occur in successive connections along the virtual path. It also minimizes background noise and prevents the negative effects of double talk. Echo cancellation can be enabled or disabled using the Echo Canceler parameter. By default, it is disabled.

1.3.5 Custom Signaling

NetPerformers equipped with a Signaling Engine can be configured for custom signaling and custom ring for analog and digital voice connections. These custom parameters define signaling characteristics used on non-standard equipment. Refer to the chapter *Custom Signaling* in the *Advanced Voice Features* fascicle of this document series.

NOTE: If no custom signaling parameters are defined, the NetPerformer will use the standard values associated with the line signaling protocol, determined from the Signaling Type parameter for the voice channel.

1.4 Analog Voice Connections

A NetPerformer analog port connects to an analog PBX or directly to an analog telephone.

- Analog PBX trunking connections require the use of an E&M interface card
- An FXS interface card is used when connecting to an analog telephone or KTS unit
- An FXO interface card is used when connecting to a CO or a PBX extension.

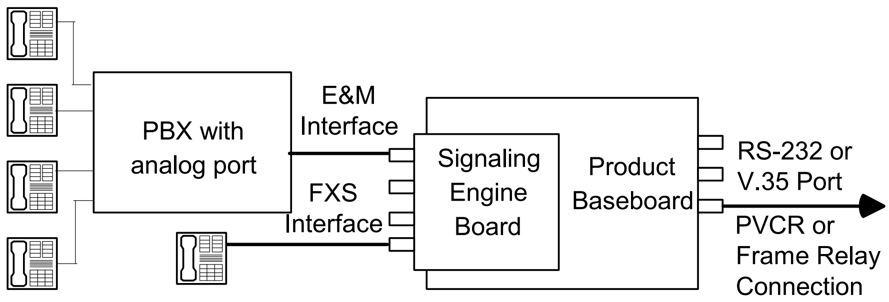


Figure 1-2: Analog Voice Support on the NetPerformer

NOTE: **Analog interface cards can be used for voice, fax and modem support only.** The voice channel *Interface type* is configured under software control from the console or through SNMP access. Further information on analog interface cards can be found in the *Hardware Installation Guide* for your NetPerformer product.

Card	Signaling	Ports per Card	Channels per Card
FXS	FXS Loop Start	2 or 4	2 analog voice
FXO	FXO Loop Start	2 or 4	2 analog voice
E&M	IMMEDIATE START, WINK START, CUSTOM	4	4 analog voice

Table 1-1: Analog voice support on analog interface cards

NOTE: 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). For details, refer to the *Digital Voice* fascicle of this document series.

1.4.1 E&M Interface Card

- The E&M interface card is used to attach an analog PBX to the NetPerformer using the **E&M** voice protocol
- The E&M interface can use a two- or four-wire circuit. Type I, II and V are supported on the NetPerformer
- The number of E&M lines supported varies from 1 to 16, depending on the NetPerformer model, the number of interface cards installed and number of DSPs available
- E&M uses conventional analog station interfaces which adhere to EIA/TIA Voiceband and Loop Signaling application standards.

1.4.2 Signaling Variations Supported

The E&M interface card supports the following signaling variations:

- **Immediate Start:** An E&M type where transmission takes place immediately. This is the industry standard for E&M operation, and the default setting on the card
- **Wink Start:** An E&M type where the unit toggles the A/B-lead before the PBX will transmit dial digits
- **Custom:** Custom signaling settings that you can use to fine-tune your application. For details, refer to the chapter *Custom Signaling* in the *Advanced Voice Features* fascicle of this document series.

1.4.3 PBX Trunk-Side Connection

The PBX trunk, or *tie line* is a communication channel between switches. It may be a direct line coming from the PBX itself, or an external device or board that supports a trunk interface.

The PBX tie line connects directly to a NetPerformer E&M voice port.

NOTE: For trunk-side connection to a NetPerformer voice port, the PBX must be able to support an E&M trunk.

Figure 1-3 demonstrates PBX trunk-side connection to the NetPerformer.

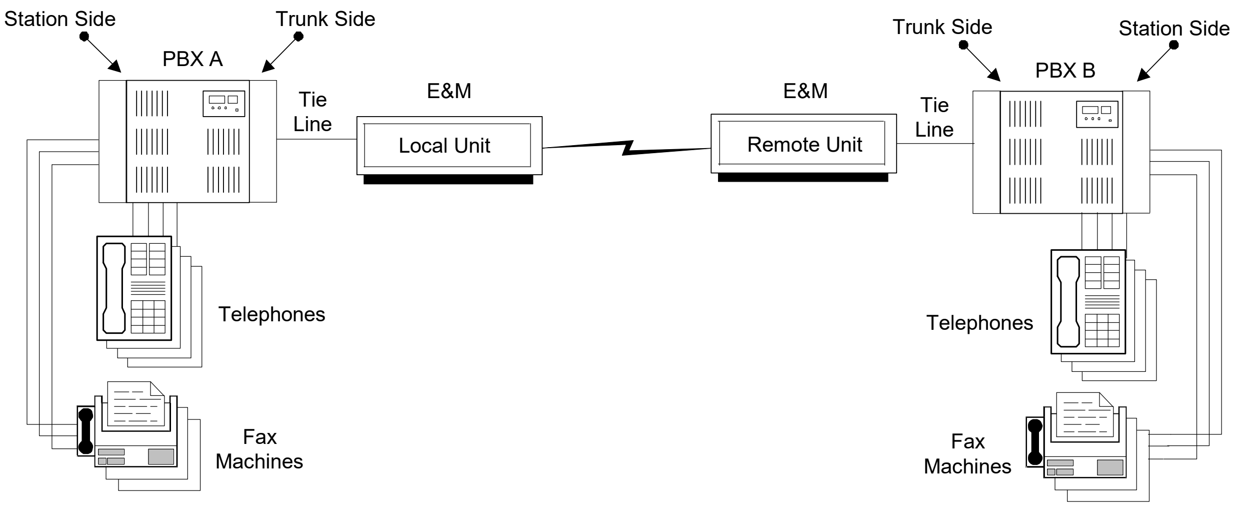


Figure 1-3: PBX Trunk-Side Connection

1.4.4 FXS Interface Card

- An FXS interface card connects a telephone or KTS unit directly to the NetPerformer unit, using the **FXS** voice protocol
- The NetPerformer presents a Telco/PTT interface that acts like a Central Office and can interface to a conventional two-wire telephone (pulse-dial or touch-tone)
- The numbers of FXS connections supported can vary from 1 to 8, depending on the NetPerformer model, the number of interface cards installed and number of DSPs available
- FXS uses the Loop Start Signaling method to seize and sense a line
    - Loop Start Signaling uses 2 wires, *Tip* and *Ring*, to perform signaling and carry Voice Frequency (VF) signals.
    - A relay opens or closes the loop between a particular subscriber and the NetPerformer FXS port. This generates current flow into the loop, which is detected by the switching equipment.
- A NetPerformer FXS port provides loop current and ring voltage, and detects the off-hook and on-hook states.

Figure 1-4 shows an application that uses both FXS and FXO interfaces (see next section) on the NetPerformer.

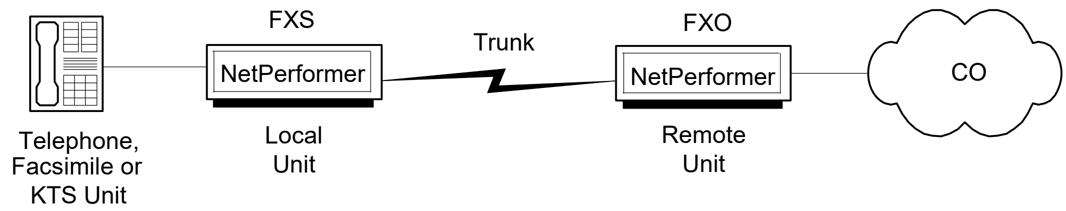


Figure 1-4: NetPerformer FXS and FXO Interfaces

1.4.5 FXO Interface Card

- An FXO interface card connects the NetPerformer to a Central Office (CO) or the station side of a PBX, using the **FXO** voice protocol
- The NetPerformer presents a Telco/PTT interface that acts like a standard telephone set
- The numbers of FXO lines supported can vary from 1 to 8, depending on the NetPerformer model, the number of interface cards installed and number of DSPs available
- FXO uses the Loop Start Signaling method to seize and sense a line

  The NetPerformer FXO connection simulates a two-wire telephone in a loop-start circuit.
- A NetPerformer FXO port detects ring voltage, closes the loop during off-hook and opens the loop in an on-hook condition.

PBX Station-Side Connection

A PBX can connect to a NetPerformer voice port from the PBX station side. In this case, the PBX is accessed in the same way as a CO.

When making a station-side connection, configure the NetPerformer voice port with an FXO interface, since the port must act like a standard telephone set (generate on-hook and off-hook, detect a ring, etc.).

Figure 1-5 demonstrates PBX station-side connection to the NetPerformer.

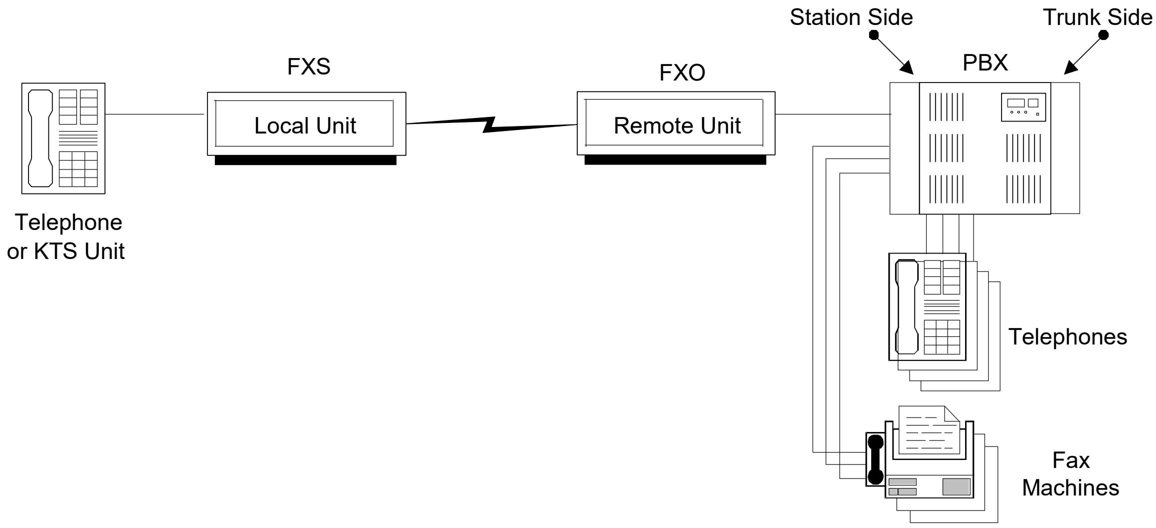


Figure 1-5: PBX Station-Side Connection

1.5 Tones Generated by the NetPerformer

The following tones can be generated by the NetPerformer for analog voice calls:

- **Audio tones:** Dial tone, ringback tone and busy signals. These are generated according to North American standards. Thus if you make a call from London, you will hear the North American signaling tones, not those for the United Kingdom. Busy signals include:
  - Slow busy, generated when the destination is busy
  - Fast busy, generated when the link goes down
  - Incompatibility tone, generated when some fatal problem with the voice connection occurs, for example, the voice algorithms are mismatched.
- **Physical tone:** Remote (equipment) ring on the telephone set, generated from the electrical signal originating from an attached CO. The frequency at which the NetPerformer generates a ring is governed by the global *Ring frequency* parameter, and may be 17, 20, 25 or 50 Hz. Its voltage is governed by the global *Ring voltage* parameter, and may be 60 or 80 Volts RMS. Refer to the chapter *Global Functions* in the *Quick Configuration* fascicle of this document series.
- **Multi-frequency tones:** These include the DTMF, MF and R2 tones. Their signals are passed transparently when a conversation is in progress. They have no effect at any time for predefined line activation (see next section). For switched line activation, they are intercepted in the early stages of the calling procedure to determine the destination unit and port from the Voice Mapping Table, and are then passed transparently once the call is placed. They can be used for interactive *touch tone* procedures during a call.
- **Billing tones:** These tones are used for billing purposes. See "Supplementary Services on an Analog Interface" on page 1-23.

NOTE: The NetPerformer does not generate a *Reorder* tone or *Flash Hook* signal.

1.6 Line Activation Types

For full network flexibility, voice/fax line activation can be configured as switched, predefined, autodial or broadcast. Use the *Activation type* parameter on the voice channel, which can be set as **SWITCHED**, **PREDEFINED**, **AUTODIAL** or **BROADCAST**. Refer to the chapter "Configuring Analog Voice Connections" on page 2-1.

1.6.1 Predefined Line Activation

For predefined activation, the destination unit and port number are preconfigured by the user. As soon as the device connected to the local port goes off-hook, the local NetPerformer begins the calling procedure with the destination device. In other words, all you have to do is lift the telephone receiver, and the remote telephone will ring immediately. **This configuration can be used only when a dedicated telephone is available at both the source and destination sites**. Often, a more popular alternative is the autodial connection, discussed later in "Autodial Line Activation" on page 1-17.

To configure a port for predefined line activation, set the voice channel *Activation type* parameter to **PREDEFINED**. You will be requested to define the *Remote unit* and *Port number*. For testing purposes, you can set up predefined line activation between two ports at the same site. This is a convenient way to test channels locally either before or after the network is in operation.

NOTE: The two voice ports linked through predefined line activation cannot be accessed by any other voice port in the network.

Example of Predefined Line Activation

Two large PBXs at different sites can be linked together through predefined line activation. The PBXs control the switching functions and the NetPerformer controls bandwidth usage and the communications line. Predefined line activation can also be used to set up a hot line, for example, between the presidents of two affiliated companies. In Figure 1-6, a hot-line connection has been set up between New York and Tokyo in a network servicing many extensions.

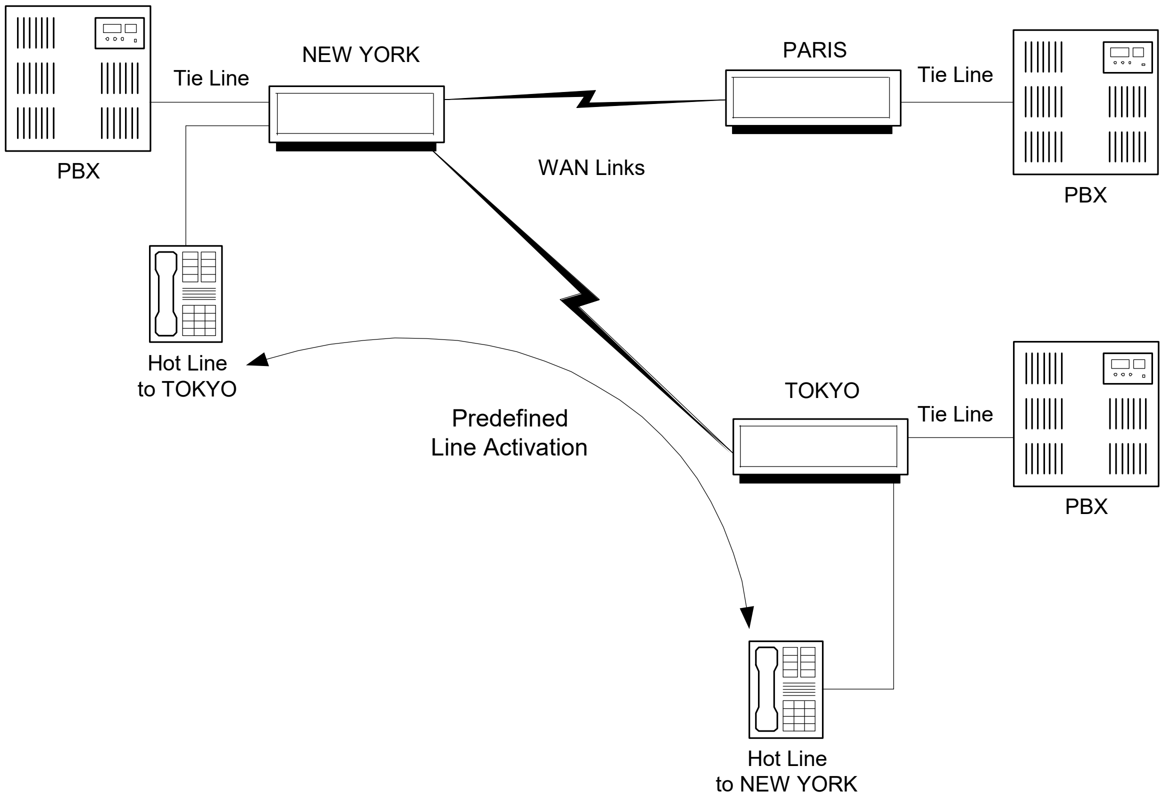


Figure 1-6: Predefined Line Activation

1.6.2 Switched Line Activation

Under switched activation (voice channel *Activation type* parameter set to **SWITCHED**), the NetPerformer selects the remote location according to a configurable *Speed Dial Number* that the user enters into the telephone set.

- No predetermined connection is set up between any two voice channels
- All speed dial numbers are kept in a Voice Mapping Table along with the associated destination unit, optional extension number, extension number length, and an optional dialing sequence (extended digits) that can be forwarded to the attached voice equipment. Refer to "Configuring the Voice Mapping Table" on page 3-1.
- Speed dial numbers in the Voice Mapping Table are variable in size (from 1 to 30 digits). The NetPerformer determines that a dialing sequence is completed when the global (inter-digits) *Dial timer* expires or when the user terminates dialing with the pound sign (#).

The advantages of switched line activation are:

- You can have an unequal number of channels at different sites. This provides a more practical and cost-effective approach to voice networks.

- Three or more offices can be interconnected without requiring a central PBX or multiple compression/decompression cycles. This method of voice/fax switching reduces delays and requires less bandwidth than for a central PBX setup.

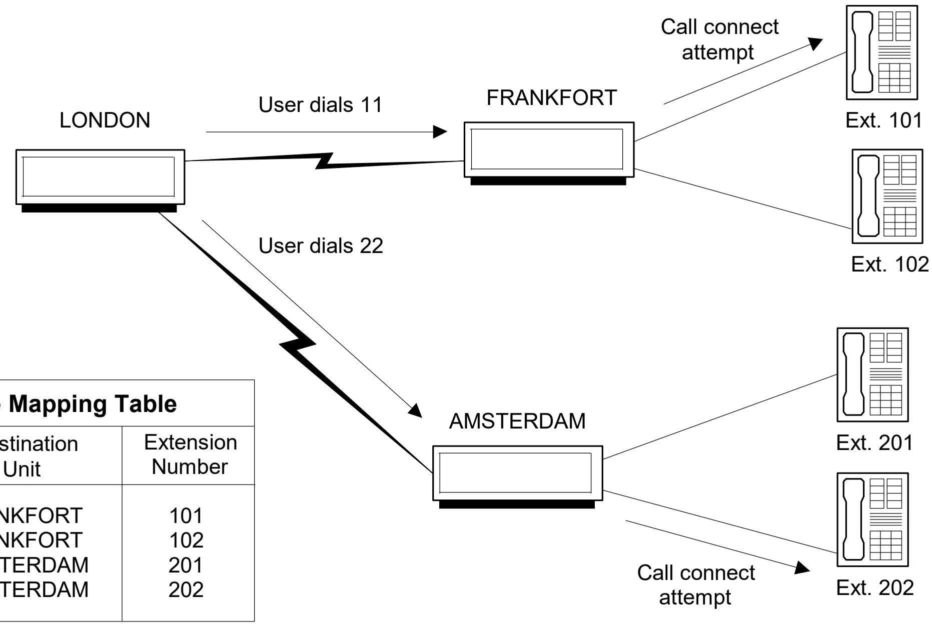
With switched line activation, the NetPerformer base product also supports:

- **Local voice switching:** Permits a switched voice call from one voice channel to another on the same unit
- **Hunt Groups:** Permits attempting more than one voice channel when trying to place an ingress call. Details on this feature are provided in the *Advanced Voice Features* fascicle of this document series.

Example of Switched Line Activation

Figure 1-7 shows how a switched connection can be made from London to either Frankfort or Amsterdam.

- The Frankfort site is defined in London's Voice Mapping Table as having speed dial numbers **11** and **12**, and Amsterdam as speed dial numbers **21** and **22**.
- The Remote Extension Number Source is set to **MAP**, which means that the extension numbers will be taken from the Voice Mapping Table and a call connect attempt will be made to one port only.
- If a user at the London site dials **11**:
  - The NetPerformer looks in its Voice Mapping Table and determines that this speed dial number is associated with destination unit Frankfort and extension number **101**.
  - It attempts to make a connection with Frankfort on the voice channel that has been defined with extension number **101**.
  - The call is placed if this port is available; otherwise a busy tone is returned to the London unit.
- The same user may dial **22** from London to access Amsterdam, extension **202**. As before, the Voice Mapping Table holds all information required for the NetPerformer to place the call.



London Voice Mapping Table

Speed Dial Number	Destination Unit	Extension Number
11	FRANKFORT	101
12	FRANKFORT	102
21	AMSTERDAM	201
22	AMSTERDAM	202

Figure 1-7: Switched Line Activation

1.6.3 Autodial Line Activation

Under autodial line activation (voice channel *Activation type* parameter set to **AUTODIAL**), the NetPerformer behaves like a switch that always dials to the same place. Autodial activation has a blend of switched and predefined features:

- Like switched activation:
  - The NetPerformer reaches the remote location using a *Speed Dial Number*. **This number is permanently configured for the voice port, and does not need to be manually entered.**
  - An extended digits sequence can be either user-dialed (configured with the voice channel definition) or defined in the Voice Mapping Table.
- Like predefined activation:
  - The 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 port configured for autodial line activation. In other words, an autodial voice channel is accessible from any other NetPerformer voice channel in the network.

1.6.4 Broadcast Line Activation

Broadcast line activation of a voice port is most commonly used in applications requiring voice broadcasting over multiple speakers, such as a PA system spanning several locations.

Under broadcast activation (voice channel *Activation type* parameter set to **BROADCAST**), the NetPerformer participates in a Frame Relay one-way multicast service. In this setup:

- Voice traffic is transmitted from one end user, called a *root*, via the multicast server (the Frame Relay switch) to all other users in the multicast group, called *leaves*.
- Frames are never sent in the opposite direction, that is, from the leaves to the root.
- The multicast service allows a user to send a single message to multiple destinations.
- Broadcast transmission is a connection-oriented service, in that a PVC connection must be established to the multicast server before the multicast data can be transmitted.
- In addition, the root must have a direct PVC to all participating leaves.

How It Works

- When sending voice traffic to the leaves, the root sends multicast frames over the multicast connection using the multicast DLCI (Mdlci).
- The multicast server accepts these frames and sends them to all of the leaves in the multicast group.
- The frames arrive at the leaves as though they were sent over the direct PVCs configured between the root and the leaves.
  - Each leaf receives the multicast frames on the PVC (DLCI) configured for direct connection to the root.
  - The Mdlci value is not included in the received frame.

NOTE: Frames sent in a multicast service are delivered to each active member of the group. If a leaf is temporarily unavailable, the frames are not kept for later delivery. The unavailable leaf will not receive frames until it returns to normal operation.

Example

In the example in Figure 1-8:

- The NetPerformer unit BOSTON (the root) uses Frame Relay for voice transmissions, and has a one-way multicast service for broadcasting voice messages to the remote NetPerformers (the leaves).

- The multicast group is logically comprised of the leaf PVCs 1, 3 and 5, which are defined for direct connection to the leaves.
- The one-way multicast service accepts broadcast frames on the Mdlci from BOSTON and transmits them to each remote unit.
- As these frames travel across the network they are treated like any other frames, and therefore arrive at their destinations as though they had actually been transmitted over the root PVCs 1 to 3.
- Thus the unit NEW_YORK receives frames on its PVC 3, CHICAGO on its PVC 5 and LOS_ANGELES on its PVC 1.

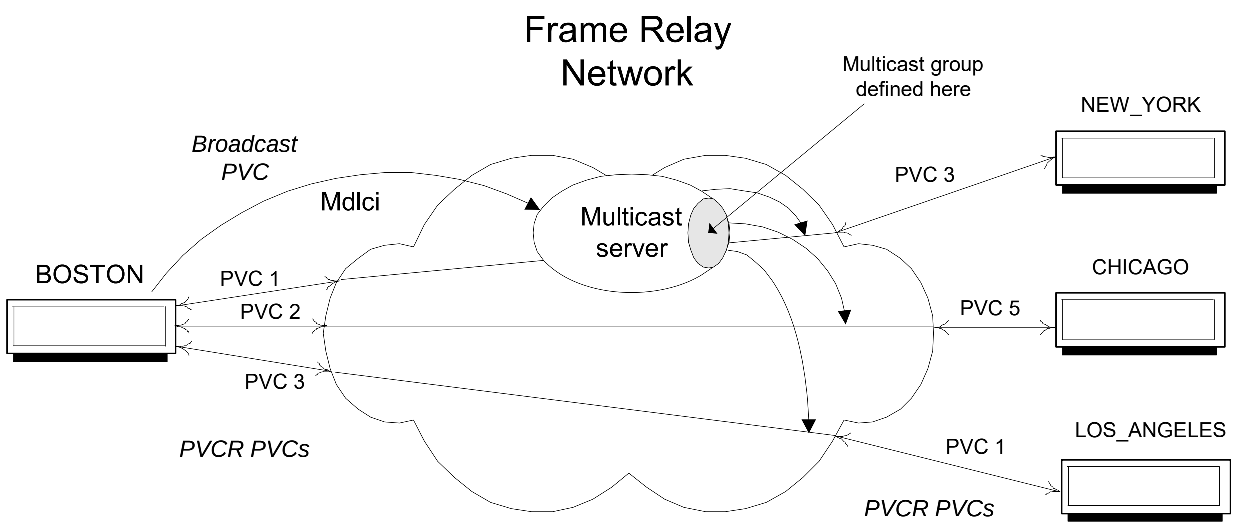


Figure 1-8: One-way Multicast Voice Broadcasting

1.6.5 Installation Requirements

To set up a Voice Broadcasting network you must:

- Install a special client interface in the PBX (or other user device) connected to the root.

  This interface generates an off-hook condition on the line and/or generates a code which the root NetPerformer sends to the Frame Relay switch through a locally initiated connection.
- Install another client interface at each device connected to the leaves.

  This interface controls the operation of the speakers at the remote sites.
- On the root NetPerformer, configure a PVCR PVC (PVC mode set to PVCR) for each connection to a leaf.
- In the example in Figure 1-8, PVCs 1, 2 and 3 at the BOSTON site are defined as PVCR PVCs (refer to the *WAN/Frame Relay* fascicle of this document series).

Since the root is on the transmitter side, the PVC *Broadcast group* parameter must be set to **NO** for all of these PVCs.

- On the root NetPerformer, configure a single Broadcast PVC (PVC mode set to **BROADCAST**) for the connection to the multicast server.

  The DLCI you define for this PVC is the Mdlci.

- On the root NetPerformer, configure an analog voice channel with Broadcast line activation (*Activation type* parameter set to **BROADCAST**) and set its *Broadcast direction* parameter to **TX**.

  This voice channel handles the actual data transmission for the PVCs. The PVC number you define for this voice port is the number of the Broadcast PVC.

NOTE: For one-way multicast broadcasting you can define only one transmitter per transmission channel.

- On each leaf NetPerformer, configure a PVCR PVC for connection to the Frame Relay network.
  - These are the PVCs that the multicast server will use to send the broadcast frames to the leaves
  - Since a leaf is on the receiver side, the PVC *Broadcast group* parameter must be set to **YES**, which defines the leaf as a member of the broadcast group.

  In the example in Figure 1-8, the PVCs at the NEW_YORK, CHICAGO and LOS_ANGELES sites are defined as PVCR PVCs in a Broadcast Group.

- On each leaf NetPerformer, configure an analog voice channel with Broadcast line activation to handle the actual data transmission between the Frame Relay network and the leaf.
  - Since the leaves receive broadcast frames, you must set the Broadcast Direction parameter to **RX** on each of these voice ports.
  - Define the *PVC number* as the number of the PVCR PVC that is included in the multicast group.

1.6.6 Operation

Voice broadcasting from the root to the leaves is typically activated in the following manner:

- The client interface at the root initiates the transmission by going off-hook
- The root starts to transmit broadcast frames, which activates the leaves.
- A connection is established between the local and remote voice channels
- The client interface at the root may generate a code digit for one second, which passes through the Frame Relay switch to all remote sites

- The client interfaces at the leaves recognize the code as a signal to activate their speakers
- Voice messages can now be sent from the root NetPerformer via the broadcast voice port and the Mdlci to the multicast server, which distributes the messages to the leaf NetPerformers, as explained earlier.

1.6.7 Deactivation

To deactivate voice broadcasting is deactivated as follows:

- The client interface at the root sends a code digit, which deactivates all leaves.
- The root hangs up and data transmission stops. Broadcast frames are no longer received by the leaf NetPerformers.

1.6.8 Specialized Line Activation

The NetPerformer supports two analog voice applications designed for special circumstances:

- *Push To Talk*, used for ground-to-air radio communications such as air traffic control, or communications between two remote locations. In both scenarios the audio path is always up.
- *Hoot and Holler*, used for creating a voice connection that is permanently off-hook.

For details on these applications turn to “Specialized Analog Voice Applications” on page 2-15.

1.7 Supplementary Services on an Analog Interface

On the **SDM-9220 and SDM-9230 only**, the following supplementary services are available on an analog interface:

- **Generation of billing signals on an FXS channel**, to keep track of billing information on voice calls. The unit must be specially configured to handle the billing information, which is generated locally (and not received from the CO). For configuration details, turn to "Configuring Billing Signals on an FXS Channel" on page 2-8.
- **Retransmission of Caller ID (ANI) over an FXS interface.** The Caller ID is sent over an FXS interface, and can be detected on the remote unit by:
  - A digital interface such as ISDN or R2, *or*
  - An analog FXO interface that has been configured for caller ID detection.

  For configuration details, turn to "Configuring Retransmission of Caller ID over an FXS Interface" on page 2-11.
- **Detection of Caller ID (ANI) on an FXO interface.** This permits the second scenario above for retransmission of caller ID over an FXS interface. Configuration details are provided on "Detection of Caller ID on an FXO Interface" on page 2-12.

2

Configuring Analog Voice Connections

2.1 Configuration Overview

To configure the NetPerformer for an analog voice application:

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

   The **LINK** parameters listed at the consildole differ according to the type of interface card used. Refer to:

   - "Configuring an FXS Physical Port (LINK)" on page 2-3
   - "Configuring an FXO Physical Port (LINK)" on page 2-3
   - "Configuring an E&M Physical Port (LINK)" on page 2-4

   If you want to configure *Supplementary Services*, also refer to "LINK Configuration Example" on page 2-8.

2. Set up all required voice channels using the **LINK/CHANNEL** option of the **SETUP** command. See "Configuring the Analog Voice Channels (CHANNEL)" on page 2-5

   If you want to configure *Supplementary Services*, also refer to "CHANNEL Configuration Example" on page 2-9.

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 "Configuring the Voice Mapping Table" on page 3-1.

   Neither the Phone profiles (using the **SETUP/PHONE** submenu) nor Caller IDs (using the **SETUP/CALLER ID** submenu) are required for an analog voice application.

Two specialized analog voice applications are described at the end of this chapter:

- "Hoot and Holler" on page 2-15
- "Push To Talk" on page 2-15.

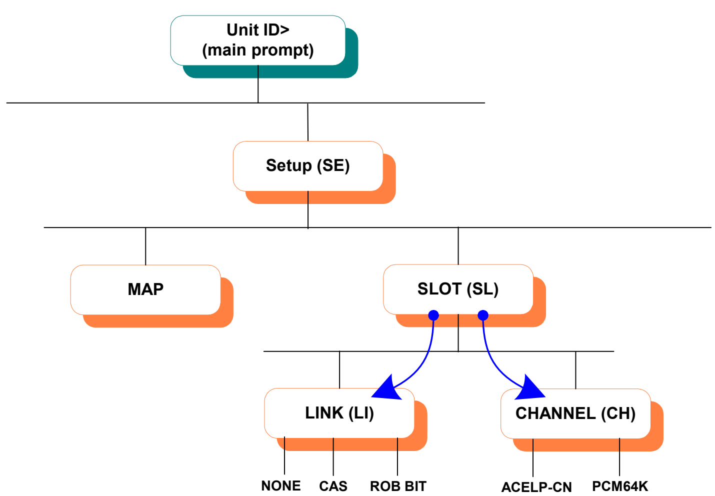


Figure 2-1: SETUP Command Paths in the CLI Tree for Analog Voice Support

2.2 Configuring the Physical Port (LINK)

2.2.1 Configuring an FXS Physical Port (LINK)

To define the physical port on an FXS 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

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

SE/SLOT/#/ LINK example: on FXS interface card

```
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:3) ?
Item (LINK/CHANNEL,def:LINK) ?
PORT 300> Status (def:DISABLE) ? ENABLE
PORT 300> Pcm encoding law (def:MU-LAW) ?
PORT 300> Billing signal type (def:12 KHZ SMOOTH) ?
```

Detailed descriptions of these parameters are provided in "SE/SLOT/#/LINK Configuration Parameters" on page 6-1. The generation of billing signals is treated in depth in the section "Configuring Billing Signals on an FXS Channel" on page 2-8.

2.2.2 Configuring an FXO Physical Port (LINK)

To define the physical port on an FXO 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

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

SE/SLOT/#/ LINK example: on FXO interface card

```
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) ?
PORT 100> Status (def:DISABLE) ? ENABLE
PORT 100> Pcm encoding law (def:MU-LAW) ?
```

These parameters are the same as those for an FXS port. Refer to “Common Parameters” on page 6-2.

2.2.3 Configuring an E&M Physical Port (LINK)

To define the physical port on an E&M 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

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

SE/SLOT/#/ LINK example: on E&M interface card

```
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) ? 3
Item (LINK/CHANNEL,def:LINK) ? LINK
PORT 300> Status (def:DISABLE) ? ENABLE
PORT 300> Pcm encoding law (def:MU-LAW) ?
PORT 300> E&M type (def:1) ?
```

- *Status* and *Pcm encoding law* behave in the same way as for an FXS port. Refer to “Common Parameters” on page 6-2.
- The *E&M type* can be set to **1**, **2** or **5**. Refer to “E&M Interface Card” on page 6-4.

2.3 Configuring the Analog Voice Channels (CHANNEL)

Refer to "Analog Voice Connections" on page 1-8 for an explanation of analog voice connections available on the analog interface cards.

To define an analog voice channel:

NOTE: Refer to "SETUP Command Paths in the CLI Tree for Analog Voice Support" on page 2-2.

1. Enter the menu sequence: **SE** ↵ **SLOT**
2. Select the *Slot number*
3. Enter **CHANNEL**
4. Select the *Port Number*: **1** or **2** on a dual-port interface card, or **1** to **4** on a quad-port interface card
5. Set the *Protocol* to a voice protocol: **ACELP-CN**, **PCM64K**, **G723**, **G726 16K**, **G726 24K**, **G726 32K**, **G726 40K**, **G729** or **G729A**
6. On an E&M interface card, select the *E&M signaling type*: **IMMEDIATE START**, **WINK START**, **CUSTOM**. Refer to "E&M Channel Parameters" on page 7-32
7. Change the other analog channel parameters from their default values, if desired.

NOTE: Analog CHANNEL configuration parameters are detailed in "SE/SLOT/#/ CHANNEL Configuration Parameters" on page 7-1.

SE/SLOT/#/ CHANNEL example: on FXS interface card with ACELP-CN Protocol

```
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) ? 3
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Port number (1-4,def:4) ? 1
VOICE 301> Protocol (def:OFF) ? ACELP-CN
VOICE 301> DSP packets per frame             1234
VOICE 301> 8K packetization selection (Y/N) ? YNNN
VOICE 301> DSP packets per frame             12345
VOICE 301> 6K packetization selection (Y/N) ? NNNNN
VOICE 301> Comfort noise level (def:0) ?
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 canceler (def:ENABLE) ?
VOICE 301> Double talk threshold (db) (def:6) ?
```

```
VOICE 301> Country settings (def:USA) ?
VOICE 301> Pulse frequency (pps) (def:10) ?
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) ?
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> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 301> Redundant channel (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> Caller ID (ANI) transmission protocol (def:OFF) ?
VOICE 301> Billing signals (def:DISABLE) ?
```

SE/SLOT/#/ CHANNEL example: on FXO interface card with PCM64K Protocol

```
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) ? CHANNEL
SLOT> Port number (1-4,def:1) ? 2
VOICE 102> Protocol (def:OFF) ? PCM64K
VOICE 102> FXO seizure delay (def:DISABLE) ?
VOICE 102> FXO timeout (s) (6-99,def:30) ?
VOICE 102> Silence suppression level (1-5,def:1) ?
VOICE 102> Local inbound voice level (db) (def:0) ?
VOICE 102> Local outbound voice level (db) (def:-3) ?
VOICE 102> Priority Level (0-10,def:0) ?
VOICE 102> Echo canceler (def:ENABLE) ?
VOICE 102> Double talk threshold (db) (def:6) ?
VOICE 102> Country settings (def:USA) ?
VOICE 102> Impedance (def:COUNTRY SPECS COMPATIBLE) ?
VOICE 102> Pulse frequency (pps) (def:10) ?
VOICE 102> Activation type (def:PREDEFINED) ?
VOICE 102> Link down busy (def:NO) ?
VOICE 102> TONE type: (def:DTMF) ? ?
VOICE 102> TONE regeneration: (0-255,def:1) ?
VOICE 102> TONE ON (ms) (30-1000,inc:10,def:100) ?
VOICE 102> TONE OFF (ms) (30-1000,inc:10,def:100) ?
VOICE 102> Pulse make/break ratio (30-50,inc:4,def:34) ?
VOICE 102> Fax relay (def:FAX) ?
VOICE 102> Maximum fax rate (def:14400) ?
VOICE 102> ECM mode (def:DISABLE) ?
VOICE 102> Modem relay (def:NONE) ?
```

```
VOICE 102> Hunt Group active (def:NONE) ?
VOICE 102> Delete digits (0-4,def:0) ?
VOICE 102> Port extension number (def:102) ?
VOICE 102> Fwd digits (def:NONE) ?
VOICE 102> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 102> Redundant channel (def:NO) ?
VOICE 102> Egress ANI operation mode (def:NONE) ?
VOICE 102> Egress CHANNEL ANI digits (def:) ?
VOICE 102> Ingress ANI operation mode (def:NONE) ?
VOICE 102> Ingress CHANNEL ANI digits (def:) ?
VOICE 102> Caller ID (ANI) detection protocol (def:OFF) ?
```

SE/SLOT/#/ CHANNEL example: on E&M interface card with ACELP-CN Protocol

```
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) ? 3
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Port number (1-4,def:1) ?
VOICE 301> Protocol (def:OFF) ? ACELP-CN
VOICE 301> DSP packets per frame              1234
VOICE 301> 8K packetization selection (Y/N) ? YNNN
VOICE 301> DSP packets per frame              12345
VOICE 301> 6K packetization selection (Y/N) ? NNNNN
VOICE 301> Comfort noise level (def:0) ?
VOICE 301> E&M signaling type (def:IMMEDIATE START) ?
VOICE 301> Analog E&M type (def:4 WIRE) ?
VOICE 301> TE timer (s) (0-255,def:0) ?
VOICE 301> Hoot & Holler application (def:NO) ?
VOICE 301> Push to Talk application (def:DISABLE) ?
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 canceler (def:ENABLE) ?
VOICE 301> Double talk threshold (db) (def:6) ?
VOICE 301> Pulse frequency (pps) (def:10) ? ?
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) ?
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> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 301> Redundant channel (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:) ?
```

2.4 Configuring Supplementary Services

On the **SDM-9220 and SDM-9230 only**, the following supplementary services can be configured on an analog interface:

- Generation of billing signals on an FXS channel, to keep track of billing information on voice calls (see next section)
- Retransmission of Caller ID (ANI) over an FXS interface ("Configuring Retransmission of Caller ID over an FXS Interface" on page 2-11)
- Detection of Caller ID (ANI) on an FXO interface ("Detection of Caller ID on an FXO Interface" on page 2-12).

2.4.1 Configuring Billing Signals on an FXS Channel

FXS channels on the SDM-9220 or SDM-9230 must be specially configured to handle billing information, which is generated locally (and not received from the CO).

To define and enable billing signals on an FXS interface card installed in the SDM-9220 or SDM-9230:

- Configure the **LINK** to generate billing signals with the appropriate *Billing signal type* for your network (see next section for details).

  As soon as a call is placed, the NetPerformer will generate this type of billing signal at configurable intervals over the physical interface of the FXS card.

- Enable the *Billing signals* parameter on each participating **CHANNEL** by selecting **EGRESS**, **INGRESS** or **BOTH ENDS**. Refer to "CHANNEL Configuration Example" on page 2-9 for details on these values.

  Caution: Billing signals will *not* be generated for any FXS channel that has its *Billing signals* parameter set to DISABLE. This is the default value.

LINK Configuration Example

Here is an example of billing signal configuration on the **LINK** of an FXS interface card:

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PVC/SCHEDULE/SLOT/USER/VLAN,def:SLOT) ? SLOT
SLOT> Slot number (1/2/3,def:1) ? 2
Item (LINK/CHANNEL,def:LINK) ? LINK
PORT 200> Status (def:DISABLE) ? ENABLE
PORT 200> Pcm encoding law (def:MU-LAW) ?
PORT 200> Billing signal type (def:12 KHZ SMOOTH) ? POL REV SMOOTH
```

- The *Billing signal type* parameter defines three characteristics of the billing signal (see Table 2-1):
  - **Frequency** of the tone: either **12 KHz** or **16 KHz**

- **Ramping** of the tone (or its reversal) up to its peak and down at the end: either **SMOOTH** or **ABRUPT**
- Whether *polarity reversal* is used: **POL REV**.

NOTE: The frequency of polarity reversal settings does not need to be specified, as these tones are not generated.

Billing Signal Type	Frequency	Ramping	Polarity Reversal
12 KHZ SMOOTH	12 KHz	smooth	no
12 KHZ ABRUPT	12 KHz	abrupt	no
16 KHZ SMOOTH	16 Khz	smooth	no
16 KHZ ABRUPT	16 Khz	abrupt	no
POL REV SMOOTH	N/A	smooth	yes
POL REV ABRUPT	N/A	abrupt	yes

Table 2-1: Characteristics of the Billing Signal Type Parameter

NOTE: When configuring a unit with SNMP, use the *ifwanFXSBillingToneType* variable to define the billing signal type.

CHANNEL Configuration Example

Here is an example of how billing signals are enabled and configured on an individual FXS channel. Required parameters are given in boldface type.

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PVC/SCHEDULE/SLOT/USER/VLAN,def:SLOT) ?
SLOT> Slot number (1/2/3,def:2) ?
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Port number (1-4,def:1) ?
VOICE 201> Protocol (def:OFF) ? ACELP-CN
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:0) ?
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> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 201> Redundant channel (def:NO) ?
VOICE 201> Egress ANI operation mode (def:NONE) ?
VOICE 201> Egress CHANNEL ANI digits (def:) ?
VOICE 201> Ingress ANI operation mode (def:NONE) ?
VOICE 201> Ingress CHANNEL ANI digits (def:) ?
VOICE 201> Caller ID (ANI) transmission protocol (def:OFF) ? V23
VOICE 201> Billing signals (def:DISABLE) ? BOTH ENDS
VOICE 201> First billing signal time (s) (0-600,def:1) ?
VOICE 201> Billing signal duration (ms) (20-1000,inc:20,def:20) ?
VOICE 201> Billing signal intervals (s) (0-600,def:1) ?
```

NOTE: *Billing signals* must be enabled separately on each participating FXS channel.

Parameter	Range of Values	Default	Function
Billing signals *ifvceAnalog-BillingTones*	NetPerformer versions later than V10.1.0 R03: DISABLE, EGRESS, INGRESS, BOTH ENDS	DISABLE	Determines the conditions for generating billing signals on this FXS channel: **EGRESS:** Billing signals are generated if this FXS channel **places a call** to the phone that is plugged into it. **INGRESS:** Billing signals are generated if this FXS channel **receives a call** from the phone that is plugged into it. **BOTH ENDS:** Billing signals are generated if this FXS channel **either places or receives a call** over the phone that is plugged into it. **DISABLE:** No billing signals are generated on this FXS channel.

Table 2-2: Parameters for Billing Signals on an FXS Channel

Parameter	Range of Values	Default	Function
First billing signal time (s) *ifvceAnalog-FirstBilling-ToneTime*	0 to 600	1	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.
Billing signal duration (ms) *ifvceAnalog-BillingTone-Duration*	20 to 1000, in increments of 20 ms	20	Sets the **duration**, in milliseconds, of each billing signal that is generated on this channel.
Billing signal intervals (s) *ifvceAnalog-BillingTone-Intervals*	0 to 600	1	Sets the **wait time**, in seconds, between the billing signals that are generated on this channel.

Table 2-2: Parameters for Billing Signals on an FXS Channel

2.4.2 Configuring Retransmission of Caller ID over an FXS Interface

A *caller ID* (ANI) received from a remote unit can be sent over an FXS interface on the **SDM-9220 and SDM-9230 only**. This caller ID is detected by the remote unit:

- On a digital interface such as ISDN or R2. In this case, only the caller ID can be retransmitted by the FXS interface.
- On an analog FXO interface configured for caller ID detection (see next section). In this case, the entire call setup message packet is retransmitted by the FXS interface, including all of its information such as the calling party name, date and time.

NOTE: The caller ID can also be configured on any interface type using the *Ingress ANI operation mode* and *Ingress CHANNEL ANI digits* parameters.

For each participating FXS channel you must define the *Caller ID (ANI) transmission protocol* parameter (SNMP variable: *ifvceAnalogCallerID*).

- At the console, enter **SE ↵ SLOT ↵ CHANNEL**.
- The *Caller ID (ANI) transmission protocol* parameter appears **after** the *Egress* and *Ingress ANI* parameters. Enter one of the following values:

- **Bell 202:** 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.**

Here is an example from the console:

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PVC/SCHEDULE/SLOT/USER/VLAN,def:SLOT) ?
SLOT> Slot number (1/2/3,def:2) ?
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Port number (1-4,def:1) ?
VOICE 201> Protocol (def:OFF) ? ACELP-CN
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:0) ?
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> Enable DTMF Detection ON-TIME (def:NO) ?
VOICE 201> Redundant channel (def:NO) ?
VOICE 201> Egress ANI operation mode (def:NONE) ?
VOICE 201> Egress CHANNEL ANI digits (def:) ?
VOICE 201> Ingress ANI operation mode (def:NONE) ?
VOICE 201> Ingress CHANNEL ANI digits (def:) ?
VOICE 201> Caller ID (ANI) transmission protocol (def:OFF) ? BELL
202
```

2.4.3 Detection of Caller ID on an FXO Interface

A *caller ID* (ANI) can be detected on an FXO interface on the **SDM-9220 and SDM-**

9230 only. For each participating FXO channel you must define the *Caller ID (ANI) detection protocol* parameter (SNMP variable: *ifvceAnalogCallerID*).

- At the console, enter **SE** ↵ **SLOT** ↵ **CHANNEL**.
- The *Caller ID (ANI) detection protocol* parameter appears **after** the *Egress* and *Ingress ANI* parameters. Enter one of the following values:
  - **Bell 202:** Uses *Bell 202* tone modulation at 1200 baud when receiving 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 when receiving the caller ID. This is the best choice for a unit located in Europe.
  - **OFF:** The caller ID, if present, is ignored on this FXO channel.

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

Here is an example from the console:

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/
PHONE/
PORT/PU/PVC/SCHEDULE/SLOT/USER/VLAN,def:SLOT) ?
SLOT> Slot number (1/2/3,def:1) ?
Item (LINK/CHANNEL,def:LINK) ? CHANNEL
SLOT> Port number (1-4,def:1) ?
VOICE 101> Protocol (def:OFF) ? ACELP-CN
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> FXO seizure delay (def:DISABLE) ?
VOICE 101> FXO timeout (s) (6-99,def:30) ?
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> Country settings (def:USA) ?
VOICE 101> Impedance (def:COUNTRY SPECS COMPATIBLE) ?
VOICE 101> Pulse frequency (pps) (def:10) ? ?
VOICE 101> Pulse frequency (pps) (10/20,def:10) ?
VOICE 101> Activation type (def:SWITCHED) ?
.
.
.
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) detection protocol (def:OFF) ? BELL 202
```

2.5 Specialized Analog Voice Applications

The NetPerformer supports two analog voice applications designed for special circumstances:

- *Hoot and Holler* (see next section), used for creating a voice connection that is permanently off-hook.
- *Push To Talk* (see "Push To Talk" on page 2-15), used for ground-to-air radio communications such as air traffic control, or communications between two remote locations

2.5.1 Hoot and Holler

The NetPerformer supports a *Hoot and Holler* connection on the E&M interface. 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 enable this type of connection use the *Hoot & Holler application* parameter, which is listed at the console for E&M voice channel configuration. The equivalent SNMP variable is *ifvceHootHoller.*

- When the *Hoot & Holler application* parameter is set to **YES**, predefined line activation is 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 the voice channel.

2.5.2 Push To Talk

Hardware Support

The NetPerformer supports a *Push To Talk* (PTT) application on the E&M interface card.

NOTE: A new Push To Talk 2/4-wire analog interface card is being developed for the NetPerformer SDM-9220/9230 as an alternative to the E&M card. This new interface card:

- Supports applications that cannot use the signaling types available on the E&M interface to carry the PTT signals
- Uses +24 VDC rather than -48VDC for these applications
- Eliminates the need for an expensive PTT to E&M converter.

Contact NetPerformer Technical Support for information on the availability of this new card. Be prepared to provide details on the PTT signal levels required for your application.

Operations

PTT is used to key a radio in two different scenarios:

- **Scenario 1:** From a remote mobile location to a control station, such as an air traffic control application. See Figure 2-2 on "Push-To-Talk Application: Scenario 1" on page 2-18.
- **Scenario 2:** Between two remote locations. See Figure 2-3 on "Push-To-Talk Application: Scenario 2" on page 2-19.

In both of these scenarios:

- The audio path is *always* up
- The PTT CONTROL device produces M lead transitions on the E&M connection
- The NetPerformer detects these M lead transitions, and regenerates them as E lead transitions
- The E lead transitions are transmitted and replayed at the other end of the connection *without clearing the audio path*
- As a result, more rapid communications can take place.

In an application involving a fixed control station (Scenario 1):

- 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
- Only the control station operates as a PTT CONTROL device.

In an application involving two remote locations (Scenario 2):

- The M lead is not permanently connected to Signal Ground at either end of the connection
- The E&M connection goes up and stays up once the first M lead transition occurs, in other words, when the first user at either location presses the talk button
- Users at both locations can use their radio as a PTT CONTROL device, producing transitions on the M lead which the NetPerformer detects and regenerates as E lead transitions at the opposite end of the connection.

Configuring Push To Talk

Push To Talk is controlled by the *Push to Talk application* parameter, which is listed at the console for E&M channel configuration. The equivalent SNMP variable is *ifvcePushToTalk*.

Scenario 1:

To configure the NetPerformer for *Push To Talk* in an application where a control station communicates with a remote mobile location (Scenario 1):

- Set the *Push to Talk application* parameter for the E&M channel on the button side of the connection to **PTT CONTROL**.

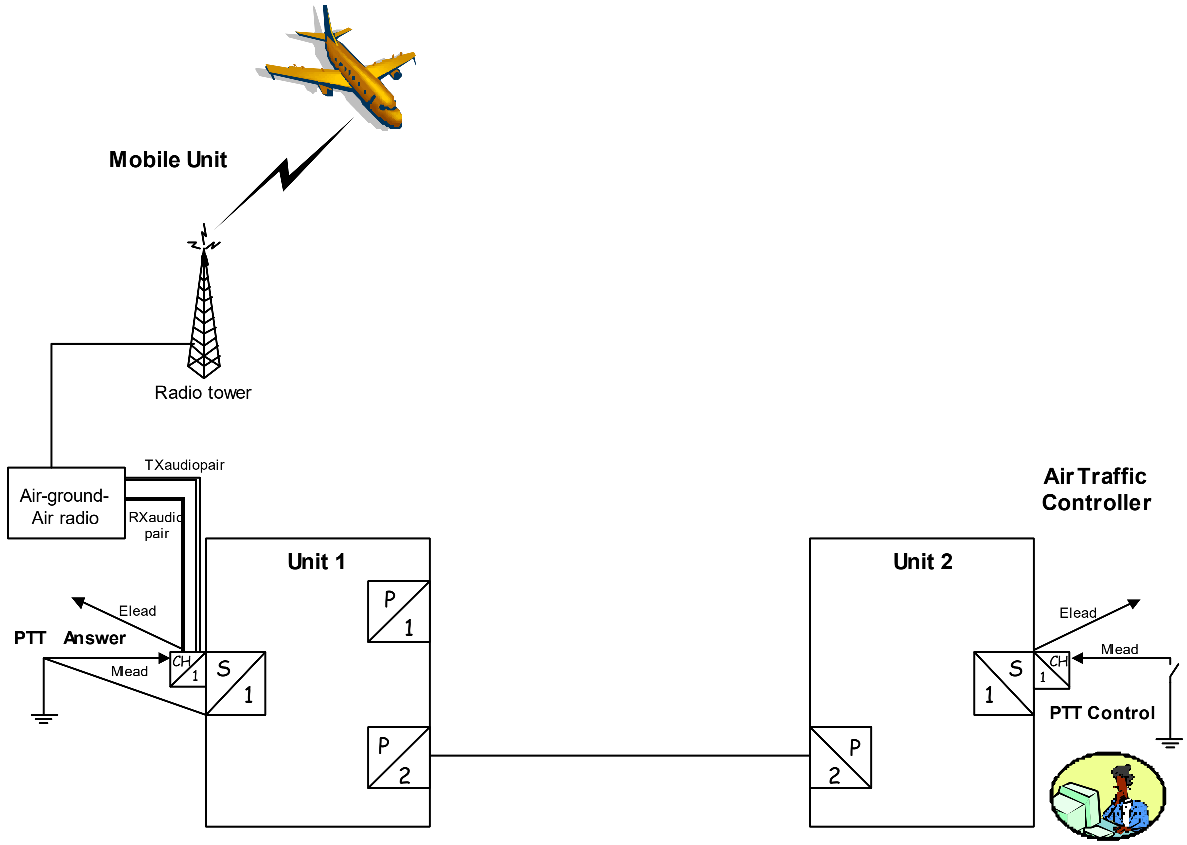
This is required for the air traffic controller (Unit 2) in Scenario 1 (see Figure 2-2).

- Set the *Push to Talk application* parameter for the E&M channel on the mobile unit side to **PTT ANSWER**.

Scenario 2: To configure the NetPerformer for *Push To Talk* in an application involving two remote locations (Scenario 2):

- 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**.



Unit 1 Configuration:

```
PORT  #100>  Status..............................ENABLE
PORT  #100>  Pcm encoding law....................A-LAW
VOICE  #101>  Protocol...........................PCM64K
VOICE #101> E&M signaling type.................IMMEDIATE START
VOICE #101> Analog E&M type....................4 WIRE
VOICE #101> TE timer (sec).....................0
VOICE #101> Hoot & Holler application..........NO
VOICE #101> Push to Talk application...........PTT ANSWER
VOICE #101> Silence suppression level..........1
VOICE #101> Local inbound voice level (db)......-3
VOICE #101> Local outbound voice level (db).....0
VOICE #101> Priority Level.....................0
VOICE #101> Echo canceler......................DISABLE
VOICE #101> Pulse frequency (pps)..............10
VOICE #101> Activation type....................PREDEFINED
VOICE #101> Link down busy.....................YES
VOICE #101> TONE type:.........................DTMF
VOICE #101> TONE regeneration:.................1
VOICE #101> TONE ON (msec).....................100
VOICE #101> TONE OFF (msec)....................100
VOICE #101> Pulse make/break ratio.............34
VOICE #101> Fax/modem relay....................NONE
VOICE #101> Remote unit.......................SDM9585#3
VOICE #101> Remote port number.................101
VOICE #101> Accept incoming ATM AAL1 calls......NO
VOICE #101> Egress ANI operation mode..........NONE
VOICE #101> Egress CHANNEL ANI digits..........
VOICE #101> Ingress ANI operation mode.........NONE
VOICE #101> Ingress CHANNEL ANI digits.........
```

Unit 2 Configuration:

```
PORT  #100>  Status..............................ENABLE
PORT  #100>  Pcm encoding law....................A-LAW
VOICE  #101>  Protocol...........................PCM64K
VOICE #101> E&M signaling type.................IMMEDIATE START
VOICE #101> Analog E&M type....................4 WIRE
VOICE #101> TE timer (sec).....................0
VOICE #101> Hoot & Holler application..........NO
VOICE #101> Push to Talk application...........PTT CONTROL
VOICE #101> Silence suppression level..........1
VOICE #101> Local inbound voice level (db)......-3
VOICE #101> Local outbound voice level (db).....0
VOICE #101> Priority Level.....................0
VOICE #101> Echo canceler......................DISABLE
VOICE #101> Pulse frequency (pps)..............10
VOICE #101> Activation type....................PREDEFINED
VOICE #101> Link down busy.....................YES
VOICE #101> TONE type:.........................DTMF
VOICE #101> TONE regeneration:.................1
VOICE #101> TONE ON (msec).....................100
VOICE #101> TONE OFF (msec)....................100
VOICE #101> Pulse make/break ratio.............34
VOICE #101> Fax/modem relay....................NONE
VOICE #101> Remote unit........................SDM9585#1
VOICE #101> Remote port number.................101
VOICE #101> Accept incoming ATM AAL1 calls......NO
VOICE #101> Egress ANI operation mode..........NONE
VOICE #101> Egress CHANNEL ANI digits..........
VOICE #101> Ingress ANI operation mode.........NONE
VOICE #101> Ingress CHANNEL ANI digits.........
```

Figure 2-2: Push-To-Talk Application: Scenario 1

Unit 1 Configuration:

```
PORT #100> Status..............................ENABLE
PORT #100> Pcm encoding law....................A-LAW
VOICE #101> Protocol...........................PCM64K
VOICE #101> E&M signaling type.................IMMEDIATE START
VOICE #101> Analog E&M type....................4 WIRE
VOICE #101> TE timer (sec).....................0
VOICE #101> Hoot & Holler application..........NO
VOICE #101> Push to Talk application...........PTT CONTROL
VOICE #101> Silence suppression level..........1
VOICE #101> Local inbound voice level (db)......-3
VOICE #101> Local outbound voice level (db).....0
VOICE #101> Priority Level.....................0
VOICE #101> Echo canceler......................DISABLE
VOICE #101> Pulse frequency (pps)..............10
VOICE #101> Activation type....................PREDEFINED
VOICE #101> Link down busy.....................YES
VOICE #101> TONE type:.........................DTMF
VOICE #101> TONE regeneration:.................1
VOICE #101> TONE ON (msec).....................100
VOICE #101> TONE OFF (msec)....................100
VOICE #101> Pulse make/break ratio.............34
VOICE #101> Fax/modem relay....................NONE
VOICE #101> Remote unit.......................SDM9585#3
VOICE #101> Remote port number.................101
VOICE #101> Accept incoming ATM AAL1 calls......NO
VOICE #101> Egress ANI operation mode..........NONE
VOICE #101> Egress CHANNEL ANI digits..........
VOICE #101> Ingress ANI operation mode.........NONE
VOICE #101> Ingress CHANNEL ANI digits..........
```

Unit 2 Configuration:

```
PORT #100> Status..............................ENABLE
PORT #100> Pcm encoding law....................A-LAW
VOICE #101> Protocol...........................PCM64K
VOICE #101> E&M signaling type.................IMMEDIATE START
VOICE #101> Analog E&M type....................4 WIRE
VOICE #101> TE timer (sec).....................0
VOICE #101> Hoot & Holler application..........NO
VOICE #101> Push to Talk application...........PTT CONTROL
VOICE #101> Silence suppression level..........1
VOICE #101> Local inbound voice level (db)......-3
VOICE #101> Local outbound voice level (db).....0
VOICE #101> Priority Level.....................0
VOICE #101> Echo canceler......................DISABLE
VOICE #101> Pulse frequency (pps)..............10
VOICE #101> Activation type....................PREDEFINED
VOICE #101> Link down busy.....................YES
VOICE #101> TONE type:.........................DTMF
VOICE #101> TONE regeneration:.................1
VOICE #101> TONE ON (msec).....................100
VOICE #101> TONE OFF (msec)....................100
VOICE #101> Pulse make/break ratio.............34
VOICE #101> Fax/modem relay....................NONE
VOICE #101> Remote unit........................SDM9585#1
VOICE #101> Remote port number.................101
VOICE #101> Accept incoming ATM AAL1 calls......NO
VOICE #101> Egress ANI operation mode..........NONE
VOICE #101> Egress CHANNEL ANI digits..........
VOICE #101> Ingress ANI operation mode.........NONE
VOICE #101> Ingress CHANNEL ANI digits..........
```

Figure 2-3: Push-To-Talk Application: Scenario 2

3

Configuring the Voice Mapping Table

3.1 About the Voice Mapping Table

The Voice Mapping Table includes definitions of all dial numbers used for call setup on a voice channel. On the NetPerformer base product, it defines all *Speed Dial Numbers* that are used for voice channels configured with **SWITCHED** or **AUTODIAL** activation.

You can add, modify or delete a **MAP** entry.

NOTE: **The MAP entries cannot be defined using SNMP.**

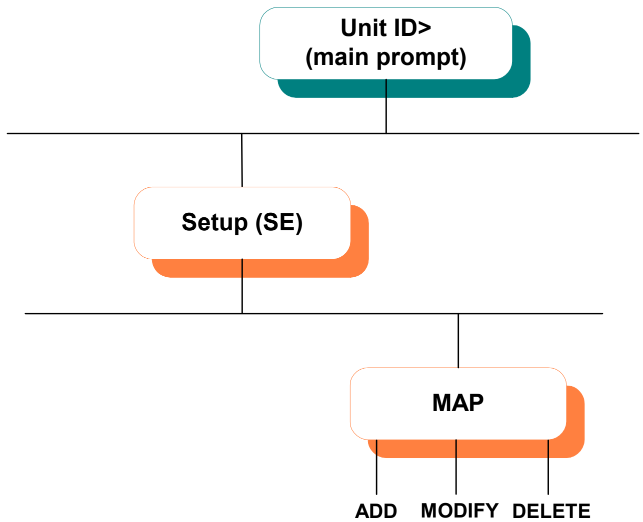


Figure 3-1: SETUP/MAP Path in the CLI Tree

A total of 1000 MAP entries can be defined in the Voice Mapping Table, allowing for a total of 1000 Speed Dial Numbers.

NOTE: If you have a large number of Voice Mapping Table entries to configure, you may prefer to upload or download the Map file using FTP. Refer to "Uploading or Downloading the MAP File" on page 3-8.

3.2 Adding a MAP Entry

To define a new **MAP** entry for a **SWITCHED** or **AUTODIAL** voice channel:

1. Enter the menu sequence: **SE** ↵ **MAP**
2. Set the *Operation* to **ADD**
3. Enter an *Entry digits* string. The character ***** can be used as a wildcard character, if desired.
   - With the Voice Traffic Routing (VTR) function, the character **!** (exclamation mark) can be used as a special wildcard character to concatenate user-dialed digits to the extended digits. During call setup the **!** is replaced by the extended digits dialed by the user. For details on VTR, consult the *Advanced Voice Features* fascicle of this document series.
   - An overloaded MAP entry can be configured if you enter the *Entry digits* string of a currently defined MAP entry. The current MAP entry is listed first, followed by the confirmation prompt:

   ```
   MAP> Do you want to overload this entry (NO/YES,def:YES) ?
   ```

   Enter **YES** to continue with overloaded MAP entry configuration. An example is provided on "SE/MAP/ADD example" on page 3-4 (**MAP 1.2**).
4. Set the *Destination name* to the *Unit name* of the remote NetPerformer. Use ***** as a wildcard character, if desired.
5. Specify the *Destination extension source*:
   - **HUNT:** The call will be connected to the first available voice channel on the destination unit that belongs to a specific Hunt Group

NOTE: Specify the desired *Hunt group* (**A** to **F**)

   - **USER:** The call will be connected to the destination extension number that the user dials (after dialing the Speed Dial Number)
   - **MAP:** The call will be connected to the extension number defined in this MAP entry

NOTE: Specify the desired *Destination extension*.

6. Specify the *Extended digits source*, if required:
   - **NONE:** No extended digits are forwarded to the remote side
   - **USER:** The user dials the extended digits that are forwarded to the remote side

NOTE: Specify the *Number of user extended digits* that can be dialed.

- **MAP:** The extended digits are taken from this MAP entry

NOTE: Specify the desired *Extended digits to forward*.

7. Set *Use SVC connection* to **YES** to allow an SVC to be defined on this voice channel
8. If using an SVC connection, change the other SVC parameters from their default values, if desired
9. Enter **YES** at the *Add another map entry* prompt if you would like to create another new MAP entry.

SE/MAP/ADD example

```
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) ? MAP
MAP> Operation (ADD/MODIFY/DELETE,def:ADD) ?
```

MAP 1: Calls any extension in a Hunt Group

```
MAP > Entry digits (def:) ? 459
MAP 1> Destination name (def:) ? CHICAGO-9230
MAP 1> Destination extension source (def:HUNT) ?
MAP 1> Hunt group (def:A) ?
MAP 1> Extended digits source (def:NONE) ?
MAP 1> Use SVC connection (def:NO) ?
MAP> Add another map entry (NO/YES,def:NO) ? YES
```

MAP 2: Calls a specific extension number, dialed by the user

```
MAP > Entry digits (def:) ? 123
MAP 2> Destination name (def:) ? CHICAGO-9220
MAP 2> Destination extension source (def:HUNT) ? USER
MAP 2> Extended digits source (def:NONE) ? USER
MAP 2> Number of user extended digits (0-27,def:0) ? 3
MAP 2> Use SVC connection (def:NO) ? YES
MAP 2> SVC address type (def:E.164) ?
MAP 2> SVC network address (def:) ? 45600731
MAP> Add another map entry (NO/YES,def:NO) ? YES
```

MAP 3: Calls a specific extension number, defined in the MAP entry

```
MAP > Entry digits (def:) ? 12345
MAP 3> Destination name (def:) ? MONTREAL-9230
MAP 3> Destination extension source (def:HUNT) ? MAP
MAP 3> Destination extension (def:) ? 123
MAP 3> Extended digits source (def:NONE) ? MAP
MAP 3> Extended digits to forward (def:) ? 1220
MAP 3> Use SVC connection (def:NO) ?
MAP> Add another map entry (NO/YES,def:NO) ? YES
```

MAP 1 renamed MAP 1.1 when a new MAP entry uses the same Entry digits

```
MAP > Entry digits (def:) ? 459

MAP 1.1> Map type...............................NAME
MAP 1.1> Entry digits...........................1
MAP 1.1> Destination name.......................CHICAGO-9230
MAP 1.1> Destination extension source...........HUNT
MAP 1.1> Hunt group.............................A
MAP 1.1> Extended digits source.................NONE
MAP 1.1> Use SVC connection.....................NO

MAP> Do you want to overload this entry (NO/YES,def:YES) ? YES
```

MAP 1.2: Defines an overloaded MAP entry

```
MAP> Position of the map to add (1-2,def:2) ? 2
MAP 1.2> Destination name (def:) ? BOSTON-8400
MAP 1.2> Destination extension source (def:HUNT) ?
MAP 1.2> Hunt group (def:A) ?
MAP 1.2> Extended digits source (def:NONE) ? USER
MAP 1.2> Number of user extended digits (0-30,def:0) ? 23
MAP 1.2> Use SVC connection (def:NO) ? YES
MAP 1.2> SVC address type (def:E.164) ?
MAP 1.2> SVC network address (def:) ? 0
MAP> Add another map entry (NO/YES,def:NO) ?
```

These parameters are detailed in “SE/MAP Configuration Parameters” on page 8-1.

3.3 Modifying a MAP Entry

To modify a MAP entry that has already been defined:

1. Enter the menu sequence: **SE** ↵ **MAP**
2. Set the *Operation* to **MODIFY**
3. Enter the *Entry digits* of the MAP entry you want to modify
4. The NetPerformer lists all parameters for this **MAP** entry with their current values, and then prompts you for the new values. Change the parameters to the new values desired, or press **<Enter>** to skip to the next parameter.

> **NOTE:** On the NetPerformer base unit, all MAP entries are identified as having the **NAME** MAP type. This cannot be changed.

5. Enter **YES** at the *Modify another map entry* prompt if you would like to modify another existing **MAP** entry.

SE/MAP/ MODIFY example

```
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) ? MAP
MAP> Operation (ADD/MODIFY/DELETE,def:ADD) ? MODIFY
MAP> Entry digits (def:) ? 123
MAP 3> Map type................................NAME
MAP 3> Entry digits............................123
MAP 3> Destination name........................CHICAGO-9220
MAP 3> Destination extension source............USER
MAP 3> Extended digits source..................USER
MAP 3> Number of user extended digits..........3
MAP 3> Use SVC connection......................YES
MAP 3> SVC address type........................E.164
MAP 3> SVC network address.....................45600731

MAP 3> Entry digits (def:123) ?
MAP 3> Destination name (def:CHICAGO-9220) ?
MAP 3> Destination extension source (def:USER) ? MAP
MAP 3> Destination extension (def:) ? 456
MAP 3> Extended digits source (def:USER) ? ? NONE
MAP 3> Use SVC connection (def:YES) ? NO
MAP> Modify another map entry (NO/YES,def:NO) ?
```

3.4 Deleting a MAP Entry

To delete an entry from the Voice Mapping Table:

1. Enter the menu sequence: **SE** ↵ **MAP**
2. Set the *Operation* to **DELETE**
3. Enter the *Entry digits* of the MAP entry you want to delete

   Caution: The MAP entry will be deleted immediately, with no confirmation requested. **Enter the *Entry digits* with care.**

4. Enter **YES** at *Delete another map entry* if you would like to delete another currently defined MAP entry.

NOTE: You can delete all entries in the Voice Mapping Table simultaneously with the Erase Map File (**EMF**) command. Enter **EMF** at the console command prompt.

SE/MAP/ DELETE example

```
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) ? MAP
MAP> Operation (ADD/MODIFY/DELETE,def:MODIFY) ? DELETE
MAP> Entry digits (def:) ? 123
MAP> Delete another map entry (NO/YES,def:NO) ?
```

3.5 Uploading or Downloading the MAP File

If you have a large number of Voice Mapping Table entries to configure, you can upload or download the MAP file from another NetPerformer unit using FTP.

Caution: Before transferring a MAP file, ensure that the MAP file versions on the two units are compatible. On each unit, view the current setting of the global Extended Parameter **MAPVERSION**, which defines the format of the Voice Mapping Table used on the unit. **MAP files are not backward compatible.** Do not replace a MAP file with one that has a lower numbered version. For assistance, contact NetPerformer Technical Support.

To download a MAP file from another NetPerformer unit:

1. Access the unit using FTP
2. Execute the *get* command, specifying the MAP filename (**MAP.TXT**)
3. After transfer is complete, change the filename to reflect the name of your NetPerformer product, using the **RENAME** command from the console.

To upload a MAP file to another NetPerformer unit:

1. Access the unit using FTP
2. Execute the *put* command, specifying the appropriate local MAP filename
3. After transfer is complete, change the filename to reflect the product name of the destination unit.

4

Monitoring Analog Voice Connections

4.1 About the NetPerformer console command

The following areas of the NetPerformer console command set provide information on how analog voice connections are operating in your network and the current configuration of parameters involved:

- To view the current status of a voice channel, 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
- To view the number and type of errors that have occurred on an interface card and its channels, use the **SLOT** option of the Display Errors (**DE**) command
- To view the current values of all analog voice parameters, use the **SLOT** option of the Display Parameters (**DP**) command.

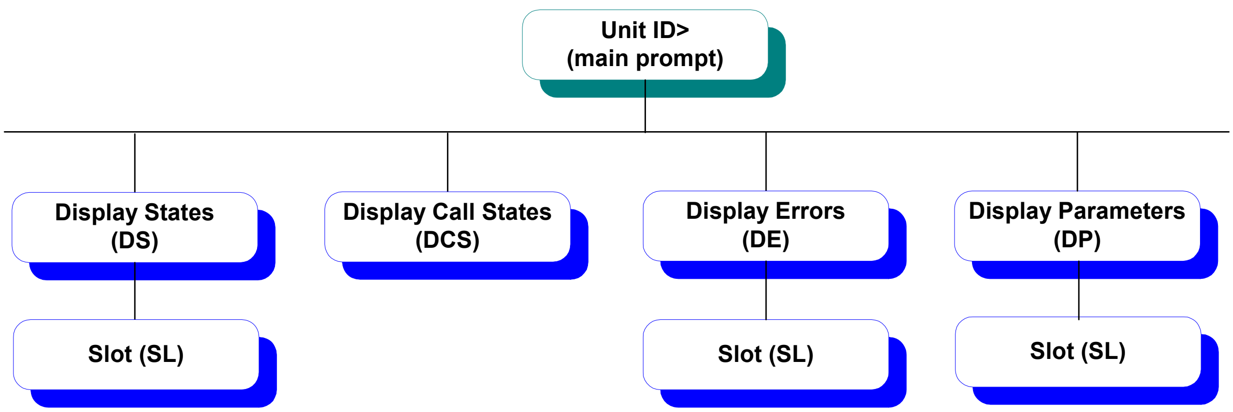


Figure 4-1: Statistics Commands in the CLI Tree for Analog Interface Cards

4.1.1 Display States (DS)

To display the current status of an analog interface card:

1. At the console command line, enter the menu sequence: **DS** ↵ **SLOT**
2. Select the *Slot number* of the analog interface card.

DS/SLOT example: on an FXS interface card

```
SDM-9230>DS
DISPLAY STATES
Item (GLOBAL/PORT/PU/PVC/SLOT/SVC/VLAN,def:GLOBAL) ? SLOT
SLOT> Slot number (1/2/3/ALL,def:1) ? 3
SLOT 3>
PORT 300> State................................ENABLE

VOICE 301> State...............................IDLE
VOICE 301> Protocol............................ACELP-CN
VOICE 301> Last error..........................NONE
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..........................NONE
VOICE 302> DSP relay rate......................NO DSP
VOICE 302> DSP relay mode......................NO DSP

VOICE 303> State...............................IDLE
VOICE 303> Protocol............................ACELP-CN
VOICE 303> Last error..........................NEW PARMS
VOICE 303> DSP relay rate......................NO DSP
VOICE 303> DSP relay mode......................NO DSP

VOICE 304> State...............................IDLE
VOICE 304> Protocol............................PCM64K
VOICE 304> Last error..........................NONE
VOICE 304> DSP relay rate......................NO DSP
VOICE 304> DSP relay mode......................NO DSP

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

4.1.2 Display Channel States (DCS)

The Display Channel States command (**DCS**) shows the status of all analog and digital channels in real time. To execute this command:

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

DCS example: on an FXS interface card

```
CHICAGO>DCS
DISPLAY CHANNEL STATES

----------------------------------- SLOT 3 : FXS -----------------------------
| #  Status Remote Unit Name  #  Rate  | #  Status Remote Unit Name  #   Rate |
|301 ONLINE 9350             1   8.0Kx1|                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
|                                      |                                      |
-------------------------------------------------------------------------------

Use LEFT and RIGHT arrow keys to change slot.  Press any other key to exit.
```

4.1.3 Display Errors (DE)

To display the number and type of errors that have occurred on an analog interface card:

1. At the console command line, enter the menu sequence: **DE** ↵ **SLOT**
2. Select the *Slot number* of the analog interface card.

DE/SLOT example: on an FXS interface card

```
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>
VOICE 301> Number of overruns..................0
VOICE 301> Number of underruns.................0
VOICE 301> No DSP available....................0

VOICE 302> Number of overruns..................0
VOICE 302> Number of underruns.................0
VOICE 302> No DSP available....................0

VOICE 303> Number of overruns..................0
VOICE 303> Number of underruns.................0
VOICE 303> No DSP available....................0

VOICE 304> Number of overruns..................0
VOICE 304> Number of underruns.................0
VOICE 304> No DSP available....................0

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

4.1.4 Display Configuration Parameters (DP)

The **DP** command provides a complete list of current values for all configuration parameters. To display the analog voice configuration parameters and their values:

1. At the console command line, enter the menu sequence: **DP** ↵ **SLOT**
2. Select the *Slot number* of the analog interface card.

DP/SLOT example: on an FXO interface card

```
SDM-9230>DP
DISPLAY PARAMETERS
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) ?
PORT 100> Status...............................ENABLE
PORT 100> Pcm encoding law.....................MU-LAW
VOICE 101> Protocol............................ACELP-CN
```

```
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..................0
VOICE 101> FXO seizure delay....................ENABLE
VOICE 101> FXO timeout (s)....................30
VOICE 101> Local inbound voice level (db).......0
VOICE 101> Local outbound voice level (db)......-3
VOICE 101> Priority Level......................0
VOICE 101> Echo canceler.......................ENABLE
VOICE 101> Double talk threshold (db)...........6
VOICE 101> Country settings....................CANADA
VOICE 101> Impedance...........................GLOBAL COMPLEX
IMPEDANCE
VOICE 101> Pulse frequency (pps)................10
VOICE 101> Activation type.....................PREDEFINED
VOICE 101> Link down busy......................NO
VOICE 101> TONE type:..........................DTMF
VOICE 101> TONE regeneration:..................1
VOICE 101> TONE ON (ms).......................100
VOICE 101> TONE OFF (ms)......................100
VOICE 101> Pulse make/break ratio...............34
VOICE 101> Fax relay...........................FAX
VOICE 101> Maximum fax rate....................14400
VOICE 101> ECM mode............................DISABLE
VOICE 101> Modem relay.........................NONE
VOICE 101> Remote unit.........................NONE
VOICE 101> Remote port number..................101
VOICE 101> Enable DTMF Detection ON-TIME........NO
VOICE 101> Redundant channel...................NO
VOICE 101> Egress ANI operation mode...........NONE
VOICE 101> Egress CHANNEL ANI digits...........
VOICE 101> Ingress ANI operation mode..........NONE
VOICE 101> Ingress CHANNEL ANI digits..........
VOICE 101> Caller ID (ANI) detection protocol...OFF
...
```

5

Calling Procedures

5.1 Initiating a Call

NOTE: This section describes the predefined and switched line activation types only. Autodial is initiated in the same manner as predefined line activation. Broadcast initiation is described in "Broadcast Line Activation" on page 1-18.

5.1.1 Predefined and Autodial Line Activation

To initiate a call when the local and remote voice ports of the NetPerformer are configured with predefined or Autodial line activation:

- Pick up the telephone receiver (at either the local or remote site).
- The call is placed automatically, and the telephone at the other end will ring.

5.1.2 Switched Line Activation

When the local and remote voice ports of the NetPerformer are configured with voice switching, the call must be initiated manually using a speed dial number that has been configured for the NetPerformer using the Setup Map menu:

Interfaces (local - remote)	Connected Equipment (local - remote)	To Initiate a Call (from local site)
FXS - FXS	Telephone - Telephone	a) Dial the speed dial number, b) Dial the remote extension number if it is not sent automatically.
FXS - FXS	Telephone - FXO equipment	a) Dial the speed dial number, b) Dial the remote extension number if it is not sent automatically, c) Ask key telephone system attendant for the desired line.
FXS - FXS	FXO equipment - Telephone	a) Select a free line on the key telephone system, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically.
FXS - FXS	FXO equipment - FXO equipment	a) Select a free line on the key telephone system, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) Ask key telephone system attendant for the desired line.

Table 5-1: Initiation Sequence for Switched Line Activation

Interfaces (local - remote)	Connected Equipment (local - remote)	To Initiate a Call (from local site)
FXS - FXO	Telephone - CO	a) Dial the speed dial number, b) Dial the remote extension number if it is not sent automatically, c) Dial the required telephone number for the CO to place the call.
FXS - FXO	FXO equipment - CO	a) Select a free line on the key telephone system, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) Dial the required telephone number for the CO to place the call.
FXS - FXO	Telephone - PBX station side	a) Dial the speed dial number, b) Dial the remote extension number if it is not sent automatically, c) Dial the required extended digits for the PBX to place the call, if they are not automatically forwarded.
FXS - FXO	FXO equipment - PBX station side	a) Select a free line on the key telephone system, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, c) Dial the required extended digits for the PBX to place the call, if they are not automatically forwarded.
FXO - FXS	CO - Telephone	a) Dial the required telephone number for the CO to place a call to the NetPerformer, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically.
FXO - FXS	PBX station side - Telephone	a) Dial the required digit sequence for the PBX to place a call to the NetPerformer, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically.

Table 5-1: Initiation Sequence for Switched Line Activation

Interfaces (local - remote)	Connected Equipment (local - remote)	To Initiate a Call (from local site)
FXO - FXS	CO - FXO equipment	a) Dial the required telephone number for the CO to place a call to the NetPerformer, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) Ask key telephone system attendant for the desired line.
FXO - FXS	PBX station side - FXO equipment	a) Dial the required digit sequence for the PBX to place a call to the NetPerformer, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) Ask key telephone system attendant for the desired line.
E&M - E&M	PBX trunk side - PBX trunk side	a) Select the PBX line for the NetPerformer trunk, for example, “8”, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) If remote PBX has an operator, ask for the desired line. Otherwise, dial the required extended digits if they are not sent automatically.
E&M - FXS	PBX trunk side - Telephone	a) Select the PBX line for the NetPerformer trunk, for example, “8”, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically.
E&M - FXS	PBX trunk side - FXO equipment	a) Select the PBX line for the NetPerformer trunk, for example, “8”, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) Ask key telephone system attendant for the desired line.
E&M - FXO	PBX trunk side - CO	a) Select the PBX line for the NetPerformer trunk, for example, “8”, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) Dial the required telephone number for the CO to place the call.

Table 5-1: Initiation Sequence for Switched Line Activation

Interfaces (local - remote)	Connected Equipment (local - remote)	To Initiate a Call (from local site)
E&M - FXO	PBX trunk side - PBX station side	a) Select the PBX line for the NetPerformer trunk, for example, “8”, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) Dial the required extended digits for the PBX to place the call, if they are not automatically forwarded.
FXS - E&M	Telephone - PBX trunk side	a) Dial the speed dial number, b) Dial the remote extension number if it is not sent automatically, c) If remote PBX has an operator, ask for the desired line. Otherwise, dial the required extended digits if they are not sent automatically.
FXS - E&M	FXO equipment - PBX trunk side	a) Select a free line on the key telephone system, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) If remote PBX has an operator, ask for the desired line. Otherwise, dial the required extended digits if they are not sent automatically.
FXO - E&M	CO - PBX trunk side	a) Dial the required telephone number for the CO to place a call to the NetPerformer, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) If remote PBX has an operator, ask for the desired line. Otherwise, dial the required extended digits if they are not sent automatically.
FXO - E&M	PBX station side - PBX trunk side	a) Dial the required digit sequence for the PBX to place a call to the NetPerformer, b) Dial the speed dial number, c) Dial the remote extension number if it is not sent automatically, d) If remote PBX has an operator, ask for the desired line. Otherwise, dial the required extended digits if they are not sent automatically.

Table 5-1: Initiation Sequence for Switched Line Activation

5.2 Terminating a Call

NOTE: This section describes the predefined and switched line activation types only. Autodial is terminated in the same manner as predefined line activation. Broadcast termination is described in “Broadcast Line Activation” on page 1-18.

5.2.1 Predefined and Autodial Line Activation

To terminate a call when the local and remote voice ports of the NetPerformer are configured with predefined or Autodial line activation:

- Hang up the telephone receiver (at either the local or remote site),
- The telephone at the other end will detect the on-hook condition, and terminate the connection.

5.2.2 Switched Line Activation

When the local and remote voice ports of the NetPerformer are configured with voice switching, the way the call is terminated depends on the connected voice/fax equipment. Only those devices that can detect an ON HOOK condition are able to terminate a call. Thus when one side of the connection is a CO or the station side of a PBX (an FXO interface on the NetPerformer), the other side is responsible for terminating the call.

Interfaces (local - remote)	Connected Equipment (local - remote)	To Initiate a Call (from local site)
FXS - FXS	Telephone/FXO equipment - Telephone/FXO equipment	Local or remote user hangs up.
FXS - FXO	Telephone/FXO equipment - CO/PBX station side	Local user hangs up.
FXO - FXS	CO/PBX station side - Telephone/FXO equipment	Remote user hangs up.
E&M - E&M	PBX trunk side - PBX trunk side	Local or remote user hangs up.
E&M - FXS	PBX trunk side - Telephone/FXO equipment	Local or remote user hangs up.
E&M - FXO	PBX trunk side - CO/ PBX station side	Local user hangs up.
FXS - E&M	Telephone/FXO equipment - PBX trunk side	Local or remote user hangs up.

Table 5-2: Termination Sequence for Switched Line Activation

Interfaces (local - remote)	Connected Equipment (local - remote)	To Initiate a Call (from local site)
FXO - E&M	CO/PBX station side - PBX trunk side	Remote user hangs up.

Table 5-2: Termination Sequence for Switched Line Activation

5.3 Examples of Calling Procedures

In this section we present several NetPerformer voice/fax applications, and describe the calling procedures that take place when a call is initiated and terminated.

5.3.1 FXS-to-FXS Application with Predefined Line Activation

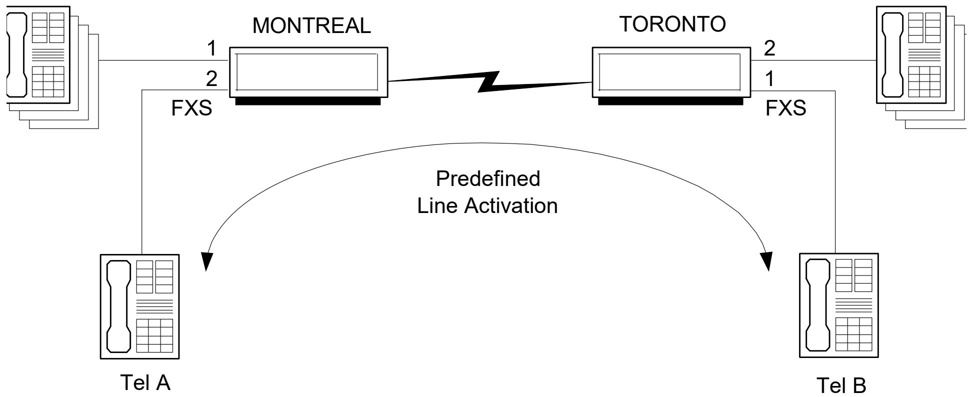


Figure 5-1: FXS-to-FXS Application with Predefined Line Activation

This application supports several switched telephones, and uses predefined line activation to send corporate information between two “hot line” telephones. The voice port configuration of the two NetPerformers specifies the connection between Tel A in Montreal and Tel B in Toronto.

Tel A is connected to voice port 2 on the local NetPerformer. Thus voice port 2 on the unit MONTREAL is configured with:

- Activation Type: **PREDEFINED**
- Remote unit: **TORONTO**
- Remote port number: **1**

Tel B is connected to voice port 1 on the remote NetPerformer. Thus voice port 1 on the unit TORONTO is configured with:

- Activation Type: **PREDEFINED**
- Remote unit: **MONTREAL**
- Remote port number: **2**

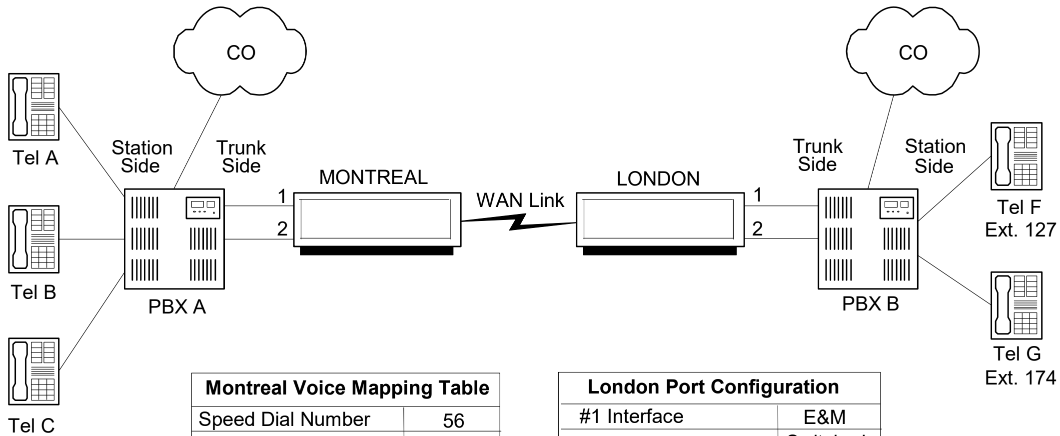
NOTE: Predefined line activation will work only if each port specifies the other as its destination. This is the first thing to check if you have problems connecting to the remote end.

Here is the calling procedure that takes place when initiating and terminating a call in this configuration:

- The user at Tel A in Montreal lifts the telephone receiver (Tel A goes off-hook). The local NetPerformer immediately requests line activation with the remote NetPerformer at the Toronto site.
- The remote NetPerformer rings Tel B by applying a physical ring signal on the port connected to Tel B. At the same time, the Toronto unit notifies the Montreal unit that it is ringing Tel B. The NetPerformer in Montreal then generates a ring-back tone at Tel A.
- The user at Tel B lifts the telephone receiver (Tel B goes off-hook). The off-hook condition is detected by the local NetPerformer, which notifies the Montreal unit to stop the ringback tone
- As soon as both ends are off-hook, a closed loop is created for transmission of voice data. Full duplex voice transmission is carried over the trunk as long as both telephones remain off hook.
- At the end of the conversation, the user at Tel B hangs up the telephone receiver (Tel B goes on-hook). The NetPerformer at the Montreal site detects the on-hook condition and deactivates the virtual connection to the NetPerformer in Toronto.

NOTE: The local NetPerformer must detect an on-hook condition before a second call can be placed. Otherwise, a busy signal will be detected, and the local unit will generate an audio busy signal at Tel A.

5.3.2 E&M-to-E&M Application with Voice Switching



Montreal Voice Mapping Table	
Speed Dial Number	56
Destination Name	London
Rem. Ext. No. Source	HUNT
Search Hunt Group	B

London Port Configuration	
#1 Interface	E&M
#1 Activation Type	Switched
#1 Hunt Group Active	B
#2 Interface	E&M
#2 Activation Type	Switched
#2 Hunt Group Active	B

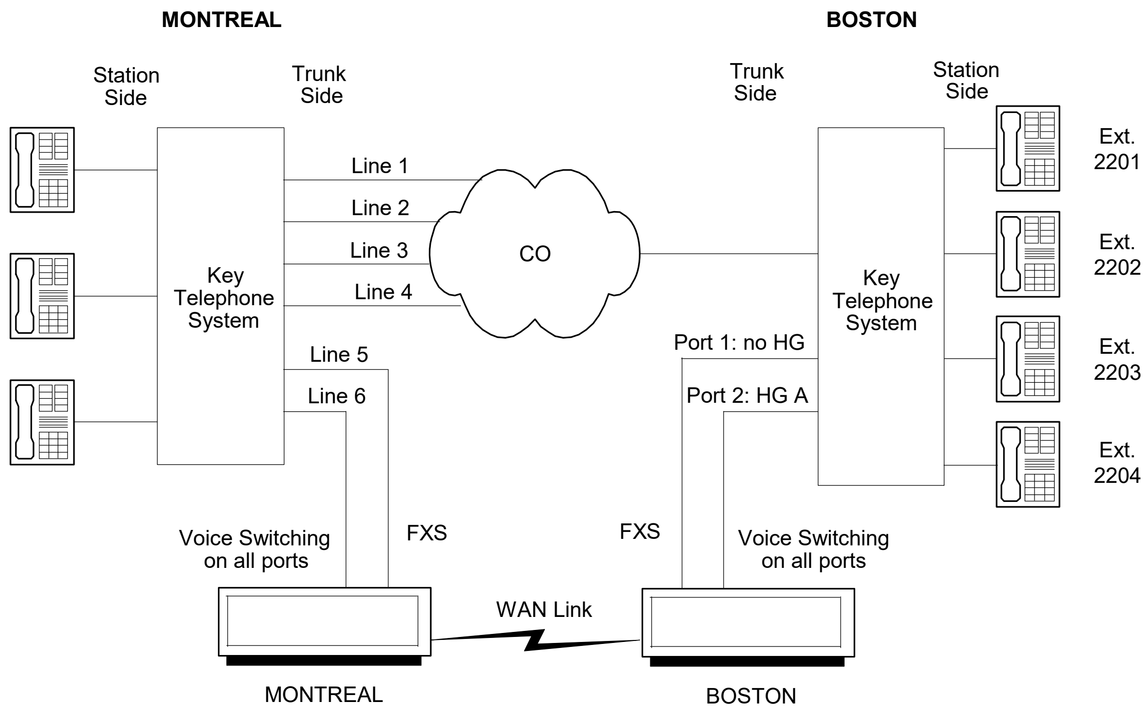
Figure 5-2: E&M-to-E&M Application with Voice Switching

In this application, Tel A in Montreal will initiate and then terminate a call to Tel G in London.

- The user at Tel A lifts the telephone receiver (Tel A goes off-hook) and hears the dial tone from PBX A.
- The user dials “8” to access an available E&M trunk from PBX A to the local NetPerformer at the Montreal site.
- The local NetPerformer detects the off-hook condition, and generates a local dial tone at Tel A.
- The user dials the speed dial number “56”.
- The local NetPerformer looks in its Voice Mapping Table, determines that speed dial number 56 is associated with the London site and that the extension number is any port configured with Hunt Group B active.
- The local NetPerformer opens a virtual connection with the remote NetPerformer in London, and sends a command to the London unit requesting connection to the first available switched port that has Hunt Group B active.
- The NetPerformer in London selects the first free port with Hunt Group B active that is available to take the call. In this example, voice ports 1 and 2 are configured as switched E&M with Hunt Group B active. The London unit tries voice port 1 first, and if it is busy tries voice port 2.
- Once connected, if PBX B is a DID (Direct Inward Dialing) system it sends a dial tone back to the NetPerformer in Montreal, which generates a local dial tone at Tel A.

- The Montreal user dials “174”, which is the extension number for Tel G. As soon as the first digit is received at London, PBX B stops the dial tone. It rings the extension, and the user at Tel G lifts the telephone receiver (Tel G goes off-hook). Note that the extended digits 174 could be transferred automatically to the London site by defining them in the Voice Mapping Table.
- If PBX B at the London site has an operator, it rings the operator station. A ring-back tone is generated at the Montreal unit. When the London operator answers, the user at Tel A in Montreal must request ext. 174 to have the call transferred manually to Tel G.
- As soon as both ends are off-hook, a closed loop is created for transmission of voice data. Full duplex voice transmission is carried over the trunk as long as both telephones remain off hook.
- At the end of the conversation, the user at Tel A hangs up the telephone receiver (Tel A goes on-hook). The NetPerformer at the London site detects the on-hook condition and deactivates the virtual connection to the NetPerformer in Montreal.

5.3.3 FXS-to-FXS Application with Voice Switching



Montreal Voice Mapping Table	
Speed Dial Number	39
Destination Name	Boston
Rem. Ext. No. Source	HUNT
Search Hunt Group	A

Boston Port Configuration	
#1 Interface	FXS
#1 Activation Type	Switched
#1 Hunt Group Active	None
#2 Interface	FXS
#2 Activation Type	Switched
#2 Hunt Group Active	A

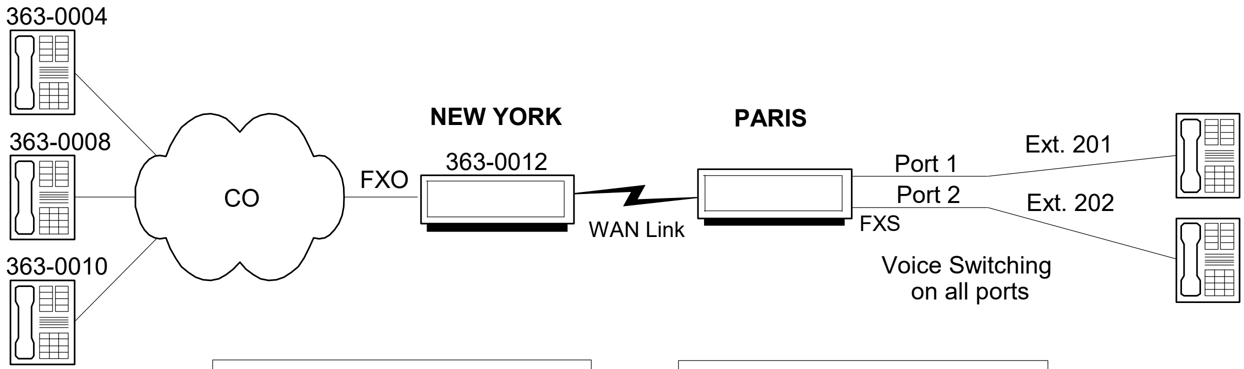
Figure 5-3: FXS-to-FXS Application with Voice Switching

In this application, the NetPerformer is used to bypass a CO. A user in Montreal initiates a call to the Boston department, and a user in Boston terminates it. One of the extensions in Boston is reserved for personal calls only, and does not take general inquiries to the Boston department.

- The user in Montreal lifts the telephone receiver and selects Line 5 (one of the lines connected to the local NetPerformer).
- The NetPerformer in Montreal detects the off-hook condition, and generates a local dial tone at the FXO extension.
- The user dials the speed dial number “39”.

- The local NetPerformer looks in its Voice Mapping Table, determines that speed dial number 39 is associated with the Boston site and that the extension number is any port configured with Hunt Group A active.
- The local NetPerformer opens a virtual connection with the remote NetPerformer in Boston, and sends a command to the Boston unit requesting connection to the first available switched port that has Hunt Group A active.
- The NetPerformer in Boston selects the first free port with Hunt Group A active that is available to take the call. In this example, voice port 1 is configured for voice switching, but it has no Hunt Group active and thus cannot take the call. Voice port 2 is configured as switched FXS with Hunt Group A active. Thus the Boston unit will attempt to complete the connection with this port.
- If voice port 2 is available, the NetPerformer at the Boston site generates a ring at the Boston KTS unit. At the same time, a ringback tone is generated at the Montreal site.
- The KTS attendant answers the call and manually forwards it to the desired extension.
- When the person at the KTS extension in Boston lifts up the telephone receiver, a closed loop is created for transmission of voice data. The local NetPerformer stops generating a ringback tone. Full duplex voice transmission is carried over the trunk as long as both telephones remain off hook.
- At the end of the conversation, the user at the Boston site hangs up the telephone receiver. The NetPerformer in Montreal detects the on-hook condition and deactivates the virtual connection to the remote NetPerformer in Boston.

5.3.4 FXO-to-FXS Application for a Specific Extension



New York Voice Mapping Table	
Speed Dial Number	14
Destination Name	Paris
Rem. Ext. No. Source	MAP
Destination Ext. No.	201

Paris Port Configuration	
#1 Interface	FXS
#1 Activation Type	Switched
#1 Hunt Group Active	A
#1 Port Extension No.	201

Figure 5-4: FXO-to-FXS Application for a Specific Extension

In this application, the WAN trunk of the NetPerformer is used as an alternative to a CO long distance connection from New York to Paris. A user connected to the CO in New York will initiate and terminate a call to a specific port extension in Paris (Ext. 201). At the Paris site, all ports are configured for Voice Switching. Here is the calling procedure:

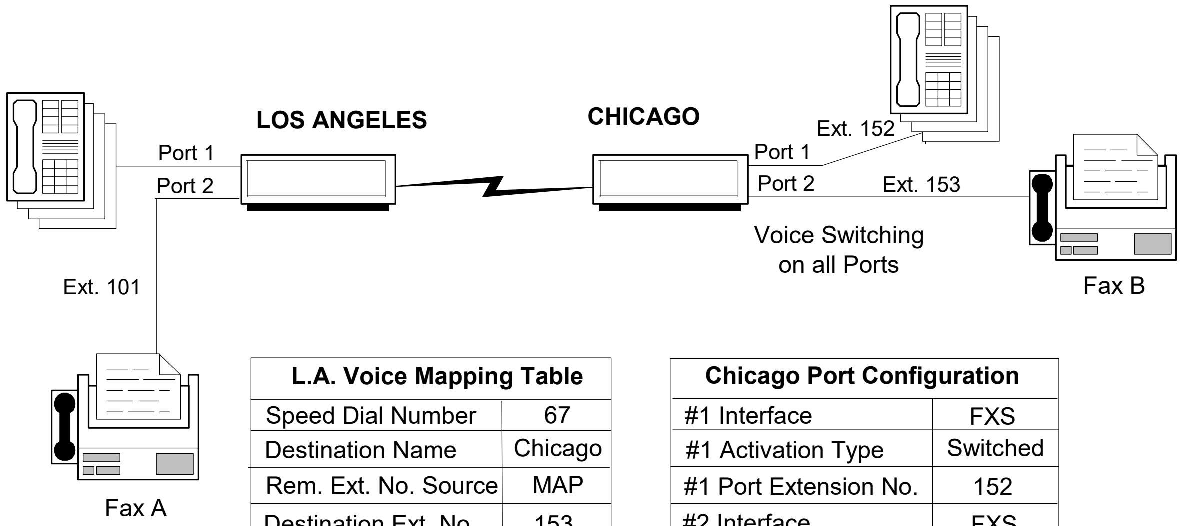
- The user connected to the CO in New York lifts the telephone receiver and hears the dial tone from the CO.
- The user dials “363-0012”, the telephone number of the local NetPerformer unit. This telephone number is within the same area code as the user’s telephone number.
- The local NetPerformer behaves like a telephone with respect to the CO. It answers the call automatically after a preconfigured number of rings. That is, it generates an off-hook condition and presents a dial tone to the New York user.
- The user dials the speed dial number “14”.
- The local NetPerformer stops the dial tone, looks in its Voice Mapping Table, determines that speed dial number 14 is associated with the Paris site, extension number 201. Since the Remote Extension Number Source is MAP (and not HUNT), the Hunt Group function will not be used when placing the call.
- The local NetPerformer opens a virtual connection with the remote NetPerformer in Boston, and sends a command to the Boston unit requesting connection to the port configured with port extension number 201.
- The Paris unit tries to connect to voice port 1, as this is the port associated with Ext. 201. Although Hunt Group A is active on voice port 1, the Hunt Group func-

tion will not be used. If voice port 1 is busy, the Paris unit sends a busy signal back to the New York site.

- If voice port 1 at the Paris site is available to take the call, it generates a ring at the telephone connected to the port. At the same time, the local NetPerformer generates a ringback tone, which is passed transparently across the CO to the user in New York.
- When the user in Paris lifts up the telephone receiver, a closed loop is created for transmission of voice data, and the local NetPerformer stops generating a ringback tone.
- At the end of the conversation, the user in New York may hang up first. However, since the local NetPerformer is configured to behave like a telephone with respect to the CO, it cannot detect the on-hook condition. Instead, the CO intercepts the on-hook indication, and simply disconnects the calling telephone.
- To terminate the call, the user in Paris at Ext. 201 **must hang up the telephone receiver to end the connection**. Since the remote NetPerformer behaves like a CO, it detects this on-hook condition, and is able to forward it to the local NetPerformer.
- In this way, the NetPerformer in New York can now detect the on-hook condition, and is able to deactivate the virtual connection to the NetPerformer in Paris.

NOTE: The local NetPerformer must detect an on-hook condition before a second call can be placed to the same extension in Paris. Otherwise, a busy signal will be generated.

5.3.5 Fax Application for a Specific Extension



L.A. Voice Mapping Table	
Speed Dial Number	67
Destination Name	Chicago
Rem. Ext. No. Source	MAP
Destination Ext. No.	153

Chicago Port Configuration	
#1 Interface	FXS
#1 Activation Type	Switched
#1 Port Extension No.	152
#2 Interface	FXS
#2 Activation Type	Switched
#2 Port Extension No.	153

Figure 5-5: Fax Application for a Specific Extension

This application supports several switched telephones, and relies on the Voice Switching configuration to send information between two fax machines without interfering with the lines reserved for voice calls.

Fax A is connected to voice port 2 on the local NetPerformer in Los Angeles. Voice port 2 on the Los Angeles unit is configured with:

- Activation Type: **SWITCHED**
- Hunt Group Active: **NONE**
- Port Extension Number: **101**

Fax B is connected to voice port 2 on the remote NetPerformer in Chicago. Voice port 2 on the Chicago unit is configured with:

- Activation Type: **SWITCHED**
- Hunt Group Active: **NONE**
- Port Extension Number: **153**

In this application, fax transmissions are sent as follows:

- A user in Los Angeles wants to send a fax to the office in Chicago. The user inserts the document into Fax A and presses “67” followed by the “Dial” button.
- Fax A generates an off-hook indication, which is detected by the local NetPerformer in Los Angeles.

- The local NetPerformer sends a dial tone to Fax A, and requests the dialing sequence.
- Fax A sends the sequence “67”. The NetPerformer in Los Angeles looks in its Voice Mapping Table, determines that speed dial number 67 is associated with the Chicago site, extension 153. Since the Remote Extension Number Source is MAP (and not HUNT), the Hunt Group function will not be used when placing the call.
- The local NetPerformer opens a virtual connection with the remote NetPerformer in Chicago, and sends a command to the Chicago unit requesting connection to the port configured with port extension number 153.
- The Chicago unit tries to connect to voice port 2, as this is the port associated with Ext. 153. If voice port 2 is busy, the Chicago unit sends a busy signal back to the Los Angeles site.
- The NetPerformer in Chicago confirms that the requested line is available, and a virtual connection is opened between voice port 2 in Los Angeles and voice port 2 in Chicago.
- The remote NetPerformer rings Fax B on Port 2 and notifies the local NetPerformer that it is ringing. The NetPerformer in Los Angeles generates a ringback tone at Fax A.
- Fax B automatically goes off-hook after a preconfigured number of rings. This off-hook condition is detected by the local NetPerformer, which stops the local ringback tone.
- Fax A generates a fax tone, which is passed transparently across the network. Fax B detects this tone, and returns another fax tone to Fax A.
- The two fax machines negotiate the speed and resolution of the transmission. Then Fax A transmits the document to Fax B over the virtual connection between the two NetPerformers.

NOTE: A fax stream requires half-duplex transmission, since one end sends a command, and then the other end responds to the command by sending data. Different modulations are used for the send (command) and receive (data) paths, for example, 300 bps for commands versus 9600 bps for data transmission. The NetPerformer monitors the fax stream and automatically adapts fax modulation to the required speed for each direction.

- Fax A notifies Fax B when document transmission is complete, and goes on-hook. The local NetPerformer detects the on-hook condition and deactivates the virtual connection to the remote NetPerformer.

6

SE/SLOT/#/LINK Configuration Parameters

NOTE: This appendix addresses LINK parameters on the **analog interface cards only**. For details on LINK parameters on the digital interface cards, refer to *SE/SLOT/#/LINK Configuration Parameters* in the *Digital Data* fascicle of this document series.

6.1 Common Parameters

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

6.1.1 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 voice channels associated with this port are disabled, but the channel configuration is not lost.

Values: DISABLE, ENABLE

Default: DISABLE

6.1.2 Pcm encoding law

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

Specifies the PCM coding law in effect on this interface.

- **A-LAW:** Commonly used in Europe
- **MU-LAW:** Commonly used in North America.

Values: A-LAW, MU-LAW

Default: MU-LAW

6.2 FXS Interface Card

The following parameter is required for an FXS interface card only:

6.2.1 Billing signal type

Console	SNMP	Text-based Config
Billing signal type	ifwanFXSBillingTone-Type	[ifwan#] FXSBillingTone-Type

Defines the nature of the billing signal that will be generated on this port, including:

- **Frequency** of the tone: either **12 KHz** or **16 KHz**
- **Ramping** of the tone (or its reversal) up to its peak and down at the end: either **SMOOTH** or **ABRUPT**
- Whether *polarity reversal* is used: **POL REV**.

NOTE: The frequency of polarity reversal settings does not need to be specified, as these tones are not generated.

Billing Signal Type	Frequency	Ramping	Polarity
12 KHZ SMOOTH	12 KHz	smooth	normal
12 KHZ ABRUPT	12 KHz	abrupt	normal
16 KHZ SMOOTH	16 Khz	smooth	normal
16 KHZ ABRUPT	16 Khz	abrupt	normal
POL REV SMOOTH	N/A	smooth	reversed
POL REV ABRUPT	N/A	abrupt	reversed

Table 6-1:

Values: 12 KHZ SMOOTH, 12 KHZ ABRUPT, 16 KHZ SMOOTH, 16 KHZ ABRUPT, POL REV SMOOTH, POL REV ABRUPT

Default: 12 KHZ SMOOTH

6.3 E&M Interface Card

The following parameter is required for an E&M interface card only:

6.3.1 E&M type

Console	SNMP	Text-based Config
E&M type	ifwanEmType1_5	[ifwan#] EmType1_5

Defines the type of E&M that is supported on this interface. The NetPerformer supports three E&M signaling standards for PBX tie line interfaces: Types I, II and V. These conventions, as defined by AT&T specifications, are described in the *Hardware and Installation Guide* for your NetPerformer product.

Values: 1, 2, 5

Default: 1

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 *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.)
- **PCM64K:** Pulse Code Modulation with non-linear compression at 64 Kbps
- **LDCD:** Low Delay Codec as 16 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#] Comfort-NoiseLevel

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

7.1.5 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

7.1.6 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

7.1.7 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

7.1.8 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

7.1.9 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

7.1.10 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. Refer to "Predefined Line Activation" on page 1-14.

- **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. Refer to "Switched Line Activation" on page 1-15.

- **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. Refer to "Autodial Line Activation" on page 1-17.

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

NOTE: 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. For details, refer to "Broadcast Line Activation" on page 1-18.

NOTE: 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

7.1.11 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** submenu. Refer to *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

7.1.12 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.

NOTE: 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

7.1.13 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.

NOTE: 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.

7.1.14 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**. Refer to “Installation Requirements” on page 1-19.

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 BROADCAST 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.

Details are provided in “Installation Requirements” on page 1-19.

Values: 1 - 300

Default: 1

7.1.15 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.

NOTE: 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.

NOTE: 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

V22 Maximum modem relay

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

Description: V22 Modem relay mode to 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#] IngressChannel-ANIDigits

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**. Refer to "Configuring Supplementary Services" on page 2-8.

- **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 (see "Configuring Supplementary Services" on page 2-8).

- **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 *ifvceImped-*

ance.

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**. Refer to “Configuring Supplementary Services” on page 2-8.

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. Refer to “Push To Talk” on page 2-15 for application examples.

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

8

SE/MAP Configuration Parameters

8.1 Operation

Console	SNMP	Text-based Map
Operation	*not available*	*not applicable*

Specifies the type of operation you would like to execute at the console:

- **ADD:** To add a new MAP entry to the Voice Mapping Table
- **MODIFY:** To change an existing MAP entry
- **DELETE:** To delete a MAP entry from the Voice Mapping Table.

Values: ADD, MODIFY, DELETE

Default: ADD

8.2 Entry digits

Console	SNMP	Text-based Map
Entry digits	*not available*	[map#] MappingEntry

Defines the digit sequence for the Speed Dial Number. This is the number that the user will dial to reach the desired destination and port. The *Entry digits* can be any length up to 19 digits; all telephone pad keys can be used.

Use the wildcard character * (asterisk) to represent a range of digits. This is especially useful when designing the voice network for Domain Dialing, discussed in the *Advanced Voice Features* fascicle of this document series.

Each wildcard character must be replaced with a digit when dialing the number. For example, the speed dial number 1* may be defined for several destinations in Chicago. When dialing to this location, the user can enter **10**, **11**, **12** and so on up to **19**, but cannot enter **1*** from the telephone keypad.

Values: 1 to 8-digit numeric string: 0 - 9, * for each digit

Default: no value

8.3 Destination name

Console	SNMP	Text-based Map
Destination name	*not available*	[map#] UnitName

Specifies the NetPerformer at the remote site to which the call will be directed. Enter the *Unit name* of the remote NetPerformer unit.

You can use the * wildcard character, if desired. For example, if you have 3 units at one site, named **BOSTON.1**, **BOSTON.2** and **BOSTON.3**, you can refer to all 3 units by setting the *Destination name* to **BOSTON.***, **BOSTON***, **BOST*** or any other shortened form ending in *****. When you dial the *Entry digits* for this MAP entry, all 3 units will be examined for an available voice port.

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

Values: Maximum 32-character alphanumeric string

Default: NONE

8.4 Destination extension source

Console	SNMP	Text-based Map
Destination extension source	*not available*	[map#] ExtensionDigitsSrc

Specifies from which source the NetPerformer will take the destination extension number:

- **HUNT:** The NetPerformer will use a Hunt Group to complete the call. The call will be connected to the first available voice channel on the destination unit that belongs to a specific *Hunt group*.

  Configure the Hunt Group with the *Hunt group* parameter, described below.
- **USER:** The call will be connected to the destination extension number that the user dials (after dialing the Speed Dial Number)
- **MAP:** The call will be connected to the extension number defined in this MAP entry.

  Configure this extension number with the *Destination extension* parameter, described below.

Values: HUNT, USER, MAP

Default: HUNT

If the *Destination extension source* is set to **HUNT**, the following parameter is also required:

8.5 Hunt group

Console	SNMP	Text-based Map
Hunt group	*not available*	[map#] HuntExtension-Digits

For HUNT Destination extension source only
Specifies the Hunt Group on the destination unit that the NetPerformer will attempt in order to complete the call.

The destination unit will try to complete the call to the lowest numbered voice channel that has the specified Hunt Group active. If this channel is busy, the next voice channel in that Hunt Group will be attempted.

Values: A, B, C, D, E, F

Default: A

If the *Destination extension source* is set to **MAP**, the following parameter is also required:

8.6 Destination extension

Console	SNMP	Text-based Map
Destination extension	*not available*	[map#] DestinationExten-sionNb

For MAP Destination extension source only
Defines the extension number that will be attempted on the remote NetPerformer. The remote unit will try to connect the call on the voice channel corresponding to the specified extension number.

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 *Destination extension* is determined by the *Extension number (no. of digits)* parameter of the **SETUP/GLOBAL** menu
- The default length is set to 3 digits
- You must enter the correct number of digits, as specified by the *Extension number (no. of digits)* parameter. If you enter the wrong number of digits, the message **Invalid Extension Number** is displayed at the console.
- At least one digit of the *Destination extension*, must be non-zero. That is, values such as **00**, **000** and **0000** are not permitted.

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

Default: no value

8.7 Extended digits source

Console	SNMP	Text-based Map
Extended digits source	*not available*	[map#] ExtendedDigitSrc

Specifies from which source the NetPerformer will take the extended digits that are to be forwarded over this voice channel to the remote side:

- **NONE:** No source is required for the extended digits. Use this value when no extended digits are to be forwarded to the remote unit (the voice channel *Fwd digits* parameter is set to **NONE**).
- **USER:** The user dials the extended digits that are forwarded to the remote side. The local voice channel will provide a dial tone and wait for the user to dial the extended digits. The user at the local site should dial the appropriate extended digits immediately after the speed dial number (and user-dialed extension number, if required).
  - The maximum number of digits that can be entered is determined by the *Number of user extended digits* parameter in this MAP entry.
  - The extended digits terminator, **#**, can be used to place the call before the maximum number of digits has been dialed and before the *Dial timer* has expired. Enter the desired digits, followed by **#**. The **#** terminator signifies that there are no more extended digits to process, and allows the call to be placed immediately.
- **MAP:** When placing the call, the local voice channel will use the extended digits defined in the Voice Mapping Table with the *Extended digits to forward* parameter.

Values: NONE, USER, MAP

Default: NONE

When the *Extended digits source* is set to **USER**, the following parameter is also required.

8.8 Number of user extended digits

Console	SNMP	Text-based Map
Number of user extended digits	*not available*	[map#] UserExtended-Digits

For USER Extended digits source only

Defines the maximum number of user-dialed extended digits that can be forwarded to the remote unit.

Values: 0 - 27

Default: 0

When the *Extended digits source* is set to **MAP**, the following parameter is also required.

8.9 Extended digits to forward

Console	SNMP	Text-based Map
Extended digits to forward	*not available*	[map#] ExtendedDigitsString

For MAP Extended digits source only
Defines the extended digits sequence that will be forwarded to the remote NetPerformer. Up to 30 digits can be forwarded.

You can include the characters **A**, **B**, **C** and **D** in the dial string to permit digit mapping that is tone type dependent. **A**, **B** and **C** are valid MF combinations used for ST3P, STP and ST2P, respectively.

NOTE: The global *Dial timer* parameter, if set to a non-zero value, can be applied to the forwarded digits. If the maximum number of extended digits is not reached, the *Extended digits to forward* will be sent automatically after the *Dial timer* expires.

Values: Maximum 30 digits, 0 - 9, A - D, , (pause), #, *

Default: no value

8.10 Use SVC connection

Console	SNMP	Text-based Map
Use SVC connection	*not available*	[map#] SvcConnection

Determines whether the remote NetPerformer will be accessed using SVCs in a Frame Relay network when this speed dial number is used. Set this parameter to **YES** to activate an SVC connection and continue with SVC configuration at the console. Set it to **NO** to disable an SVC connection. In this case, a Frame Relay connection must be made using PVCs.

If you are using SVCs, the following conditions must be met:

- The *Management interface* of the FR-USER port must be set to **ANNEX-D**. Refer to the *WAN/Frame Relay* fascicle of this document series
- The *Link down busy* parameter must be set to **NO** on all voice channels that may use SVCs to reach their destination.

- If you are using SVCs, but not fax or modem communications, the *Fax Relay* and *Modem Relay* parameters on the voice channel must both be set to **NONE**. This will prevent the SVCs from taking too much bandwidth.
- If you are using both SVCs and fax or modem communications, set the *Maximum fax rate* and *Maximum modem rate* parameters to the maximum speed the voice channel may require for fax or modem communications. This will provide sufficient bandwidth to both the SVCs and fax/modem transmissions.

The *Link down busy*, *Fax relay*, *Modem relay*, *Maximum fax rate* and *Maximum modem rate* parameters are described in SE/SLOT/#/CHANNEL Configuration Parameters on page 1.

Values: NO, YES

Default: NO

If *Use SVC connection* is set to **YES**, the following parameters are also required.

8.11 SVC address type

Console	SNMP	Text-based Map
SVC address type	*not available*	[map#] SvcAdressType

Determines the type of SVC network addressing that will be used to access the remote NetPerformer. Select either **E.164** or **X.121**. This parameter is required if two Frame Relay ports with different addressing types are available on the same NetPerformer unit.

Values: E.164, X.121

Default: E.164

8.12 SVC network address

Console	SNMP	Text-based Map
SVC network address	*not available*	[map#] SvcNetAddr

Specifies the SVC network address that will be used to access the remote NetPerformer. This may be an E.164 or X.121 address, depending on the value of the SVC address type parameter.

Values: Maximum 15 digits, **0 - 9** for each digit

Default: no value

8.13 Add another map entry

Console	SNMP	Text-based Map
Add another map entry	*not available*	*not applicable*

Determines whether parameter prompts for another MAP entry will be provided at the console. If you select **NO**, the NetPerformer will save the MAP entry you have just defined, and display the message **Saving map entry...** at the console.

Values: NO, YES

Default: NO

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