NetPerformer® System Reference

Advanced Voice Features

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Voice Traffic Control

1.1 About Voice Traffic Control

The NetPerformer is able to control the maximum number of voice channels that pass through a WAN link, Frame Relay port or PVC.

- If an unlimited number of voice channels were allowed through any port or PVC the result could be voice packet underruns, choppy voice conversations or congestion over the network.
- Voice Traffic Control permits more efficient use of the available bandwidth by blocking new voice connections once the maximum number of channels is reached.

1.2 Configuration

► To configure a NetPerformer for Voice Traffic Control:

- Set the *Maximum number of voice channels* parameter in the **SETUP/PORT** and **SETUP/PVC** menus.

In a NetPerformer network, each destination may be served by several links. For example, a PVCR port and PVCR PVC may be defined with the same remote unit name. Since the NetPerformer performs load balancing of all traffic going to the remote unit, it is not possible to know in advance over which link a voice packet will be transmitted. For this reason, instead of blocking voice channels on a particular link, the decision to block a new voice connection is based on the maximum number of voice channels allowed for all links serving the destination.

- This means that if you set the *Maximum number of voice channels* to 300 on the PVCR port, the PVCR PVC may also be affected.
- Neither the PVCR port nor the Frame Relay port associated with the PVCR PVC can carry more than 300 channels, either individually or as a group.
- A new voice connection will be made only if the *Maximum number of voice channels* has not been reached for the group as a whole, the PVCR ports or the Frame Relay ports serving the destination.

1.3 Operation

When a call is blocked:

- A fast busy tone is emitted for 5 seconds
- Then the Link Down Busy sequence is carried out (see the description of the *Link down busy* parameter in the Analog Voice fascicle of this document series).
- To see whether the fast busy tone is due to Voice Traffic Control or to a Link Down Busy situation, execute the Display States (**DS**) command.
    - The last error listed will specify the cause of the problem (**TOO MANY CALLS** versus **UNREACHABLE**)
    - Also, the State will be listed as **NO LINK** during a Link Down Busy situation.
- You can also carry out a capture of the voice port with the Start Capture (**STC**) command, level 2, and look at the **CONNEC CONF** type. Refer to the *Monitoring and Statistics* fascicle of this document series for details.

Voice Traffic Control is available on data-only NetPerformers as well as integrated voice/data units. For data-only products, this allows for voice traffic control on intermediate units that route voice packets to their final destination.

For details concerning the *Maximum number of voice channels* parameter, refer to the appendix *SE/PORT/#/PVCR Configuration Parameters* in the *WAN/Leased Lines* fascicle of this document series.

2

Enhanced Dialing

2.1 About Enhanced Dialing

The NetPerformer's Enhanced Dialing features provide a multitude of functions previously available only in advanced PBX systems. Traditionally, PBX to PBX connectivity was accomplished using leased *tie-line* circuits. As data connectivity requirements have grown, it has become imperative to combine voice and data circuits in order to increase circuit cost efficiency.

- Channel banks and voice compression multiplexers were deployed between PBX's to allow users to mix voice and data into common communication links.
- Time division and packetized multiplexers and digital circuit multiplication equipment (DCME) use voice compression algorithms and silence detection techniques to intelligently and dynamically redistribute bandwidth on an as-needed basis.

Intelligent telephone communication devices can perform dial string manipulations, which allow the user to enter one dial string to reach a destination, regardless of the number of PBX's, multiplexers, or links that are traversed in the process. NetPerformer Enhanced Dialing features automate the communication path and reduce the need for repetitive user actions.

2.2 Parameters Used

In the following sections discuss the various Enhanced Dialing features of the NetPerformer.

- Each section describes the application in the context of a telephone call which arrives at one NetPerformer (the local site or calling unit) and is forwarded to another NetPerformer (the remote site or called unit).
- Each application discusses how to set up the parameters on both the calling and the called sides to make the application work.

It is important to understand which configuration parameters are used by the calling and the called NetPerformer units. Some of the Enhanced Dialing parameters are global parameters, others are specific to the voice ports or channels, and still others are part of the Voice Mapping Table.

SE Menu	Parameter	Range	Description
GLOBAL	Dial timer	*0 – 10 sec.*	How long should the calling NetPerformer wait for input before timing out?
GLOBAL	Extension number (no. of digits)	*2 - 4*	The number of digits that must be used to define an extension number in the Voice Mapping Table.
SLOT/ CHANNEL	Speed dial number	Determined by *Entry digits* in MAP entries	**AUTODIAL** *Activation type* only. Specifies which speed dial number will be dialed when a off-hook condition occurs on this voice channel.

Table 2-1: Calling Side Parameters

SE Menu	Parameter	Range	Description
MAP	Entry digits	Determined by the best match in MAP entries, or the GLOBAL *Dial timer*.	Actual speed dial digits that the user must dial to get to a destination.
SLOT/ CHANNEL	Remote unit, Remote port number	Determined by the *Unit name* and available channels on remote unit.	**PREDEFINED** *Activation type* only.Determines the destination of the call.
MAP	Destination name	Determined by the *Unit name* on remote unit.	Determines the destination of the call when a Speed Dial Number (*Entry digits*) is used.
MAP	Destination extension source	*HUNT, USER, MAP*	If an extension number at the destination device is required, from where will it originate? It can be selected automatically from a Hunt Group, dialed by the user or programmed into the mapping table.
MAP	Destination extension	Determined by GLOBAL Extension number (no. of digits)	**MAP** *Destination extension source* only. Specifies the extension number
MAP	Extended digits source	*NONE, USER, MAP*	If extended digits are to be forwarded out of the called NetPerformer, from where will they originate? The static choice is to have the extended digits pre-programmed into the mapping table. The dynamic choice is to forward the user-dialed digits. By default no source is required, and no extended digits are forwarded.
MAP	Extended digits to forward	*0 - 30*	**MAP** *Extended digits source* only. The extended digits that will be forwarded to the remote side.
MAP	Number of user extended digits	*0 - 27*	**USER** *Extended digits source* only. The maximum number of user-dialed extended digits that can be entered. User-dialed extended digits specified in the map configuration are used for both **SWITCHED** and **AUTODIAL** *Activation type*.
SLOT/ CHANNEL	Fwd digits	*NONE, ALL, EXT*	**SWITCHED** *Activation type* only. The called channel has the choice of forwarding none of the digits, the speed dial and the extended digits (ALL), or just the extended digits (EXT). The EXT value is not available when the NetPerformer is installed with the SIP VoIP licensed software option.

Table 2-1: Calling Side Parameters

SE Menu	Parameter	Range	Description
SLOT/ CHANNEL	Fwd type	*DTMF, PULSE*	**SWITCHED** *Activation type* only. When the called voice port forwards digits to the attached device, such as a PBX, it can do it using pulse dial or DTMF.
SLOT/ CHANNEL	Fwd delay	*0 – 10000 msec.*	**SWITCHED** *Activation type* only. Set in 250 ms increments, this parameter specifies: • The delay that will be introduced after the called NetPerformer receives the digits from the calling NetPerformer and before they are forwarded to the attached device. • The value of a pause character "," in a dial string. Longer delays can be created by using several pause characters in the string.
SLOT/ CHANNEL	Delete digits	*0 - 4*	**SWITCHED** *Activation type* only. How many digits will be deleted from the dial string coming into the voice channel from the externally attached device (PBX)?

Table 2-1: Calling Side Parameters

NOTE: In the examples on the following pages, the values of some parameters are not relevant to the application, and are marked as **N/A**.

2.3 Converting Pulse to Tone

Use this application to convert pulse dialed digits into DTMF tones at the remote side. This conversion takes place during call setup only. Once the call is established, the feature is no longer operative.

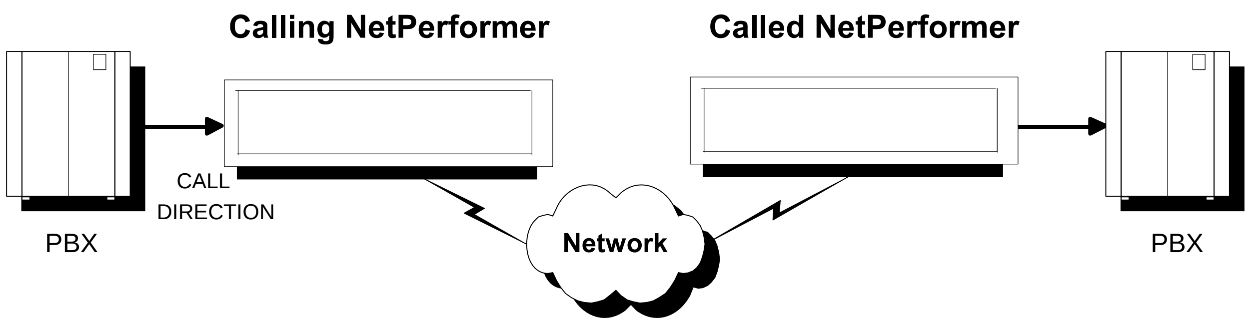


Figure 2-1: Converting Pulse to Tone

Unit	SE Submenu	Parameter Name	Value
Calling	GLOBAL	Dial timer	N/A
		Extension number (no. of digits)	*3*
Calling	SLOT/CHANNEL	Activation type	*SWITCHED*
		Speed dial number	N/A
		Remote unit, Remote port number	N/A
Called	SLOT/CHANNEL	Port extension number	*101*
		Fwd digits	*ALL* or *EXT*
		Fwd type	*DTMF*
		Fwd delay	*0*
		Delete digits	N/A

Table 2-2: Converting Pulse to Tone Parameters

Unit	SE Submenu	Parameter Name	Value
Calling	MAP	Entry digits	*1234*
		Destination name	The called unit name
		Destination extension source	*MAP*
		Destination extension	*101*
		Extended digits source	N/A
		Extended digits to forward	N/A
		Number of user extended digits	N/A

Table 2-2: Converting Pulse to Tone Parameters

If we assume that an analog voice interface card is installed in Slot 1 at both the calling and called NetPerformers, pulse-to-tone conversion can be achieved by providing pulse dialing to the calling NetPerformer. When the dialed digits arrive at the called unit, it will output DTMF digits based on the setting of the *Fwd type* parameter. To accept pulse dialing, the voice port on the calling NetPerformer must be configured with an FXS or E&M interface.

NOTE: If *Fwd digits* (SLOT/CHANNEL) is set to **EXT**, then the NetPerformer also uses the settings for *Extended digits source* (MAP). Whenever the *Extended digits source* is set to **MAP**, the value of *Extended digits to forward* (MAP) is also used.

2.4 Converting Tone to Pulse

Use this application to convert DTMF digits to pulse dialed digits at the remote site. This conversion takes place during call setup only. Once the call is established, the feature is no longer operative.

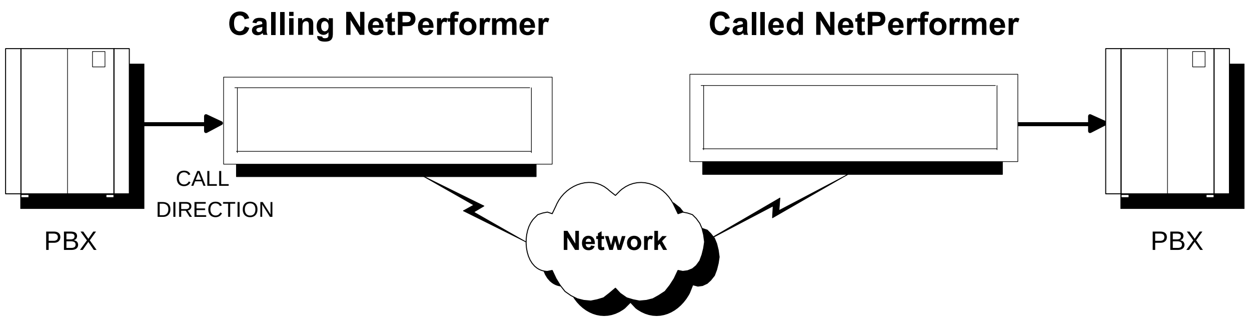


Table 2-3: Converting Tone to Pulse

Unit	SE Submenu	Parameter Name	Value
Calling	GLOBAL	Dial timer	N/A
		Extension number (no. of digits)	*3*
Calling	SLOT/CHANNEL	Activation type	*SWITCHED*
		Speed dial number	N/A
		Remote unit, Remote port number	N/A
Called	SLOT/CHANNEL	Port extension number	*102*
		Fwd digits	*ALL* or *EXT*
		Fwd type	*PULSE*
		Fwd delay	*0*
		Delete digits	N/A

Table 2-4: Converting Tone to Pulse

Unit	SE Submenu	Parameter Name	Value
Calling	MAP	Entry digits	*2345*
		Destination name	The called unit name
		Destination extension source	*MAP*
		Destination extension	*102*
		Extended digits source	N/A
		Extended digits to forward	N/A
		Number of user extended digits	N/A

Table 2-4: Converting Tone to Pulse

If we assume that an analog voice interface card is installed in Slot 1 at both the calling and called NetPerformers, tone-to-pulse conversion can be achieved by providing DTMF dialing to the calling NetPerformer. When the dialed digits arrive at the called unit, it will output the digits using pulse dialing, as specified by the *Fwd type* parameter. To forward pulse dialing, the voice port on the called NetPerformer must be configured with an FXO or E&M interface.

2.5 Forwarding Dialed Digits

Use this application to forward dialed digits from the remote NetPerformer to the attached device (PBX). For example, if you dial 4567 at the calling NetPerformer, the called unit will forward 4567 to the attached PBX when the call to remote NetPerformer is established.

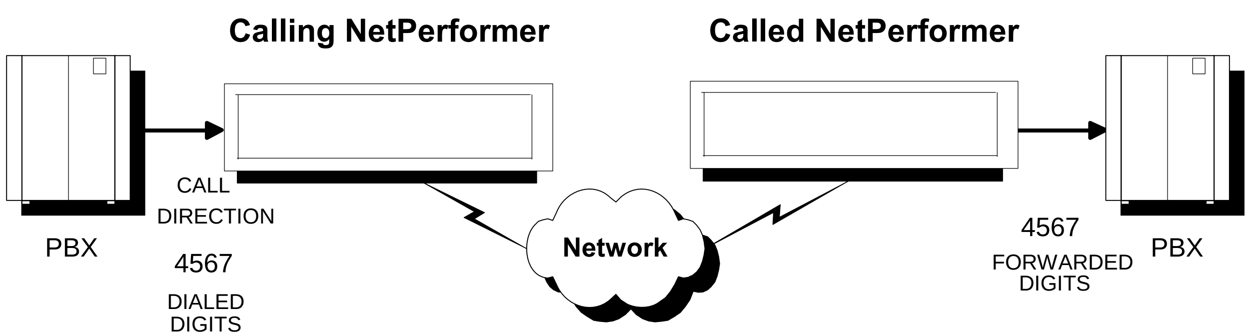


Figure 2-2: Forwarding Dialed Digits

Unit	SE Submenu	Parameter Name	Value
Calling	GLOBAL	Dial timer	N/A
		Extension number (no. of digits)	*3*
Calling	SLOT/CHANNEL	Activation type	*SWITCHED*
		Speed dial number	N/A
		Remote unit, Remote port number	N/A
Called	SLOT/CHANNEL	Port extension number	*103*
		Fwd digits	*ALL*
		Fwd type	*DTMF* or *PULSE*
		Fwd delay	*0*
		Delete digits	N/A

Table 2-5: Forwarding Dialed Digits Parameters

Unit	SE Submenu	Parameter Name	Value
Calling	MAP	Entry digits	*4567*
		Destination name	The called unit name
		Destination extension source	*MAP*
		Destination extension	*103*
		Extended digits source	N/A
		Extended digits to forward	N/A
		Number of user extended digits	N/A

Table 2-5: Forwarding Dialed Digits Parameters

In this example, 4 digits are used for the Speed Dial Number. If *Extended digits to forward* are specified in the calling NetPerformer's Voice Mapping Table entry for *Entry digits* 4567, those digits will also be forwarded from the called NetPerformer to the attached PBX.

2.6 Deleting Leading Digits

Use this application to delete up to 4 dialed digits from the incoming dial string before passing the digits on to the called NetPerformer. For example, if the PBX produces the digit stream 234567 at the calling NetPerformer, the first 2 leading digits may be deleted. The remaining digits, 4567, are then forwarded to the called NetPerformer.

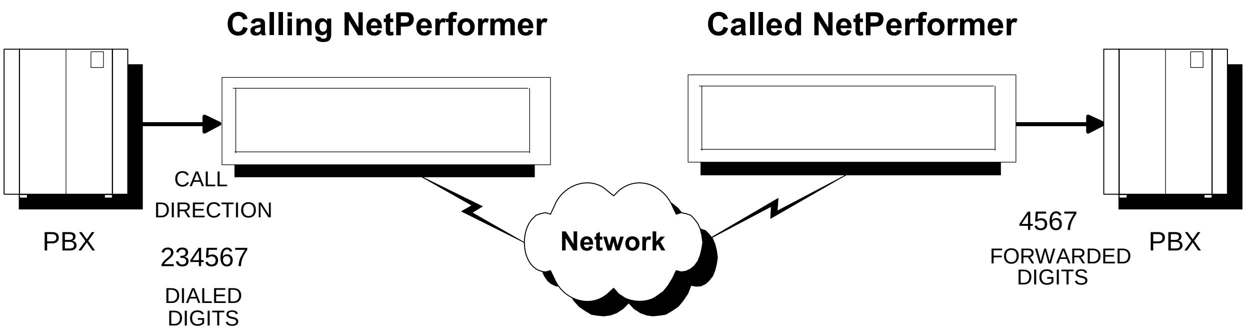


Figure 2-3: Deleting Leading Digits

Unit	SE Submenu	Parameter Name	Value
Calling	GLOBAL	Dial timer	N/A
		Extension number (no. of digits)	*3*
Calling	SLOT/CHANNEL	Activation type	*SWITCHED*
		Speed dial number	N/A
		Remote unit, Remote port number	N/A
		Delete digits	*2*
Called	SLOT/CHANNEL	Port extension number	*104*
		Fwd digits	ALL
		Fwd type	*DTMF* or *PULSE*
		Fwd delay	*0*
		Delete digits	N/A

Table 2-6: Deleting Leading Digits

Unit	SE Submenu	Parameter Name	Value
Calling	MAP	Entry digits	*4567*
		Destination name	The called unit name
		Destination extension source	*MAP*
		Destination extension	*104*
		Extended digits source	N/A
		Extended digits to forward	N/A
		Number of user extended digits	N/A

Table 2-6: Deleting Leading Digits

In this example we are deleting the first 2 digits and using the next 4 digits as the Speed Dial Number. Delete digits may be set from **0** to **4** depending on the requirements of your application.

2.7 Converting the Dialed Number to a Different Number

Use this application to dial a Speed Dial Number at the calling NetPerformer, and have a completely different number sent to the called NetPerformer for forwarding to a PBX. This is accomplished by using Extended Digits with MAP as the source. For example, if you dial 7730 at the calling NetPerformer, it looks up this number in its Voice Mapping Table and forwards the dialed number along with an extended digits number to the called NetPerformer.

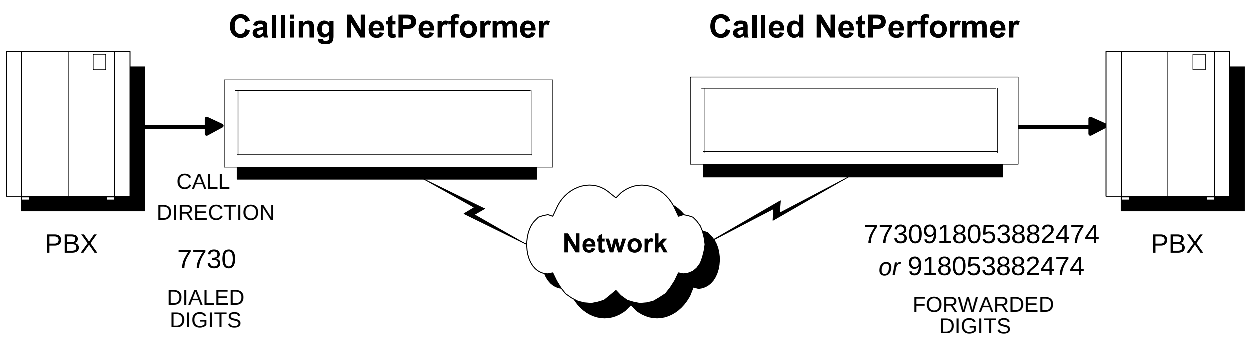


Figure 2-4: Converting Dialed Numbers

Unit	SE Submenu	Parameter Name	Value
Calling	GLOBAL	Dial timer	N/A
		Extension number (no. of digits)	*3*
Calling	SLOT/CHANNEL	Activation type	*SWITCHED*
		Speed dial number	N/A
		Remote unit, Remote port number	N/A
		Delete digits	*0*
Called	SLOT/CHANNEL	Port extension number	*104*
		Fwd digits	*ALL* or *EXT*
		Fwd type	*DTMF* or *PULSE*
		Fwd delay	*1000*
		Delete digits	N/A

Table 2-7: Converting Dialed Numbers Parameters

Unit	SE Submenu	Parameter Name	Value
Calling	MAP	Entry digits	*7730*
		Destination name	The called unit name
		Destination extension source	*MAP*
		Destination extension	*104*
		Extended digits source	*MAP*
		Extended digits to forward	*918053882474*
		Number of user extended digits	N/A

Table 2-7: Converting Dialed Numbers Parameters

In this example 7730 was dialed, and the extended digits 918053882474 were appended and sent to the called NetPerformer for forwarding to the attached PBX. Depending on the setting of the *Fwd digits* parameter at the called NetPerformer, two forwarding scenarios are possible:

- If *Fwd digits* is set to **ALL** the entire string, 7730918053882474, is forwarded. Since the *Fwd delay* parameter is set to **1000 msec**, there will be a one-second delay before any digits are forwarded to the PBX at the remote site.
- If *Fwd digits* is set to **EXT** only the extended digits, 918053882474, are forwarded. Once again, there is a one-second delay before the called unit forwards this string to the attached PBX.

NOTE: The maximum number of extended digits that can be entered is 30. In this application, the total number of digits which may be forwarded is 16 (4 speed dial digits and 12 extended digits).

2.8 Adding Pauses to Forwarded Numbers

The previous application can be modified to include pauses in the digit string that is forwarded from the called NetPerformer to the attached PBX. Pauses cannot be introduced from the user dialed string. They must be added to the *Extended digits to forward* parameter in the Voice Mapping Table. To do this, insert a comma "," in the appropriate position of the *Extended digits to forward* string.

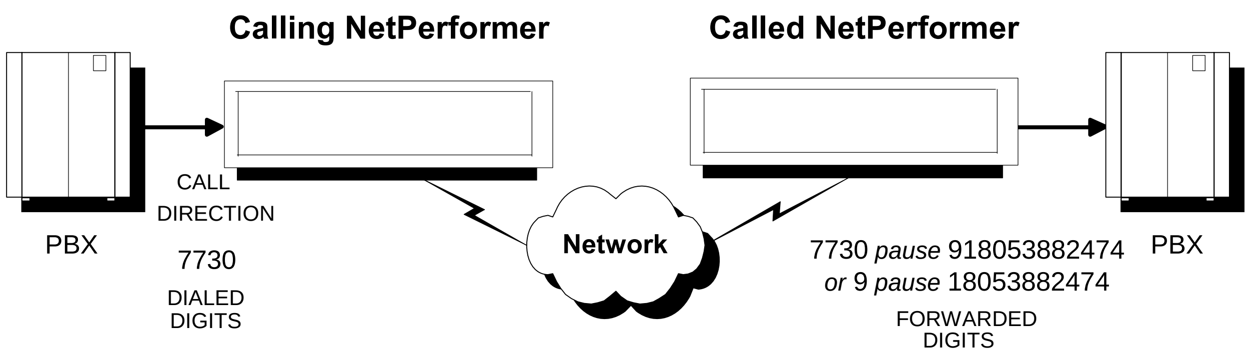


Figure 2-5: Adding Pauses to Forwarded Numbers

Unit	SE Submenu	Parameter Name	Value
Calling	GLOBAL	Dial timer	N/A
		Extension number (no. of digits)	*3*
Calling	SLOT/CHANNEL	Activation type	*SWITCHED*
		Speed dial number	N/A
		Remote unit, Remote port number	N/A
		Delete digits	*0*
Called	SLOT/CHANNEL	Port extension number	*104*
		Fwd digits	*ALL* or *EXT*
		Fwd type	*DTMF* or *PULSE*
		Fwd delay	*2000*
		Delete digits	N/A

Table 2-8: Adding Pauses to Forwarded Numbers Parameters

Unit	SE Submenu	Parameter Name	Value
Calling	MAP	Entry digits	*7730*
		Destination name	The called unit name
		Destination extension source	*MAP*
		Destination extension	*104*
		Extended digits source	*MAP*
		Extended digits to forward	*9,18053882474*
		Number of user extended digits	N/A

Table 2-8: Adding Pauses to Forwarded Numbers Parameters

In this example 7730 was dialed, and the extended digits 9,18053882474 were appended and sent to the called NetPerformer for forwarding to the attached PBX. Depending on the setting of the *Fwd digits* parameter at the called NetPerformer, two forwarding scenarios are possible:

- If *Fwd digits* is set to **ALL** the entire string including a pause, 77309*Pause*18053882474, is forwarded. Since the *Fwd delay* parameter is set to **2000 msec**, there will be a two-second delay before any digits are forwarded to the PBX at the remote site.
- If *Fwd digits* is set to **EXT** only the extended digits including a pause, 9*Pause*18053882474, are forwarded. Once again, there is a two-second delay before the called unit forwards this string to the attached PBX.

Each comma produces a single pause. The length of this pause is determined from the *Fwd delay* parameter (if 0, the value of the comma is 0). Multiple commas can be strung together to achieve a longer pause (if *Fwd delay* is non-zero). Each comma counts as one of the 30 digits permitted in the *Extended digits to forward* string.

2.9 Forwarding Long Dial Strings to the Remote PBX

Use this application to dial a speed dial number followed by up to 16 additional digits that will be forwarded by the called NetPerformer to the remote PBX. This is accomplished by using Extended Digits with USER as the source. For example, if you dial 3456918053883504 at the calling NetPerformer, it parses the string and determines that 3456 is the Speed Dial Number (since it matches an entry in the Voice Mapping Table). The calling unit then determines the destination, places the call and forwards the entire dialed number, including the Speed Dial Number and the user-dialed Extended Digits, to the called NetPerformer.

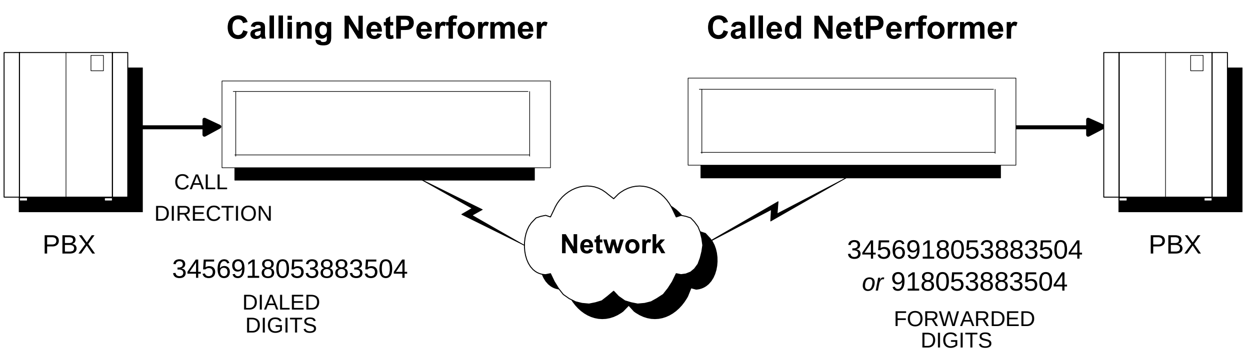


Figure 2-6: Forwarding Long Dial Strings to the Remote PBX

Unit	SE Submenu	Parameter Name	Value
Calling	GLOBAL	Dial timer	N/A
		Extension number (no. of digits)	*3*
Calling	SLOT/CHANNEL	Activation type	*SWITCHED*
		Speed dial number	N/A
		Remote unit, Remote port number	N/A
		Delete digits	*0*
Called	SLOT/CHANNEL	Port extension number	*105*
		Fwd digits	*ALL* or *EXT*
		Fwd type	*DTMF* or *PULSE*
		Fwd delay	N/A
		Delete digits	N/A

Table 2-9: Forwarding Long Dial Strings to the Remote PBX Parameters

Unit	SE Submenu	Parameter Name	Value
Calling	MAP	Entry digits	3456
		Destination name	The called unit name
		Destination extension source	*MAP*
		Destination extension	*105*
		Extended digits source	*USER*
		Extended digits to forward	N/A
		Number of user extended digits	*12*

Table 2-9: Forwarding Long Dial Strings to the Remote PBX Parameters

Depending on the setting of the *Fwd digits* parameter at the called NetPerformer, two forwarding scenarios are possible:

- If *Fwd digits* is set to **ALL** the entire string, 3456918053883504, is forwarded.
- If *Fwd digits* is set to **EXT** only the user-dialed Extended Digits, 918053883504, are forwarded.

In this example, the maximum number of extended digits that can be entered is 12, and the total maximum number of digits which may be forwarded is 16 (4 speed dial digits and 12 extended digits).

NOTE: A **USER** *Extended digits source* can be specified in the Voice Mapping Table only for a voice channel with **SWITCHED** or **AUTODIAL** *Activation type*.

2.10 Wildcard Dialing

Use this application to program a single Voice Mapping Table entry that will allow several similar numbers to get to the same destination. For example, Speed Dial Number 456* will reach the same destination whether you dial 4560, 4561, 4562, and so on up to 4569. By itself this feature does not represent a major advantage. However, when coupled with the fact that a dialed number can be forwarded at the called location to an attached PBX, this feature supports a DID or DISA type dialing plan.

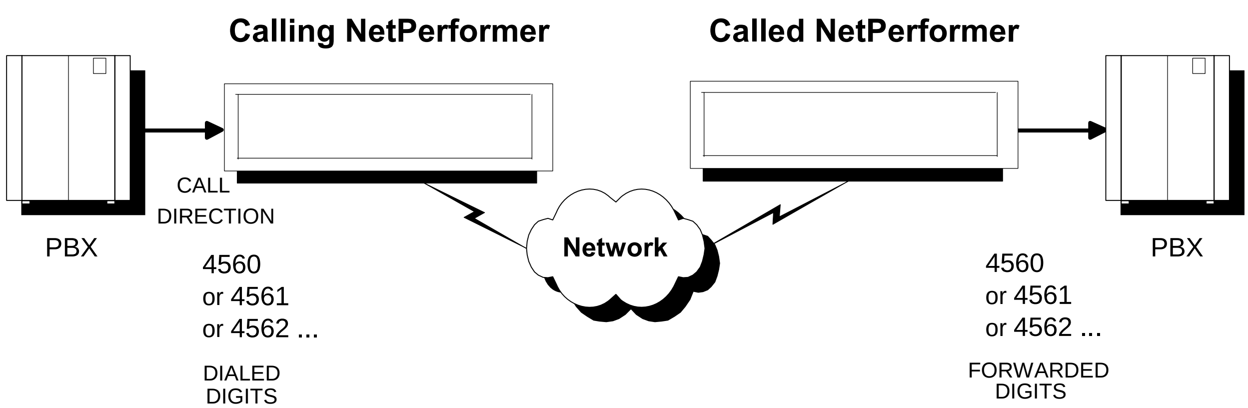


Figure 2-7: Wildcard Dialing

Unit	SE Submenu	Parameter Name	Value
Calling	GLOBAL	Dial timer	N/A
		Extension number (no. of digits)	*3*
Calling	SLOT/CHANNEL	Activation type	*SWITCHED*
		Speed dial number	N/A
		Remote unit, Remote port number	N/A
		Delete digits	*0*
Called	SLOT/CHANNEL	Port extension number	*106*
		Fwd digits	*ALL*
		Fwd type	*DTMF* or *PULSE*
		Fwd delay	N/A
		Delete digits	N/A

Table 2-10: Wildcard Dialing Parameters

Unit	SE Submenu	Parameter Name	Value
Calling	MAP	Entry digits	*456**
		Destination name	The called unit name
		Destination extension source	*MAP*
		Destination extension	*106*
		Extended digits source	*USER*
		Extended digits to forward	N/A
		Number of user extended digits	N/A

Table 2-10: Wildcard Dialing Parameters

In the above example, if you dial 4560, 4561 or 4562 the calling NetPerformer will try to reach the same destination NetPerformer. Once the call is established, however, the dialed number is forwarded to the PBX, which can route the call to different destinations based on that number. In order to forward the speed dial digits from the called NetPerformer to the attached PBX, its *Fwd digits* parameter must be set to **ALL**.

2.11 Wink Start Signaling

The Wink Start signaling method is used in conjunction with E&M signaling only. The voice channel *E&M signaling type* parameter must be set to **WINK START**.

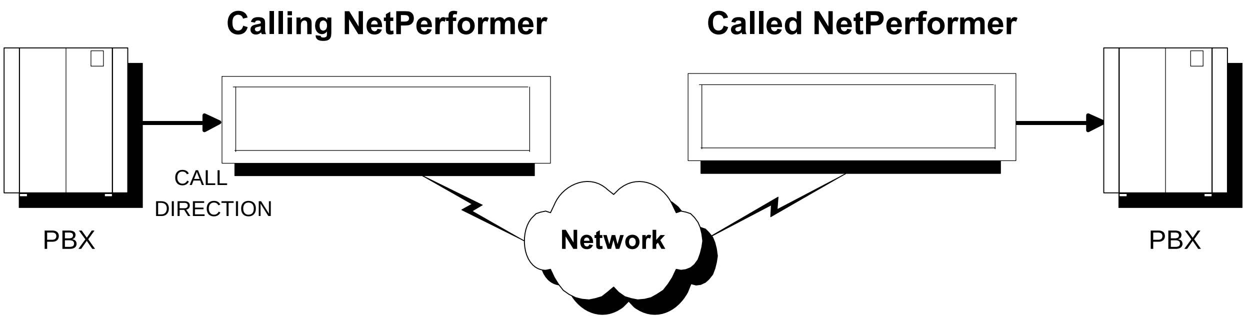


Figure 2-8: Wink Start Signaling Method

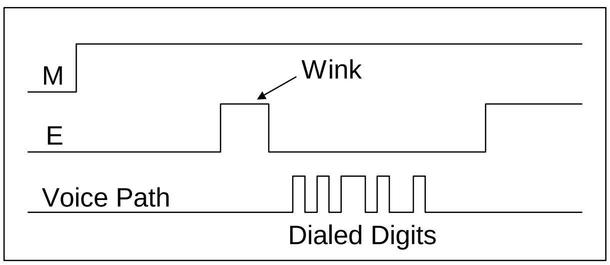
Unit	SE Submenu	Parameter Name	Value
Calling	GLOBAL	Dial timer	N/A
		Extension number (no. of digits)	*3*
Calling	SLOT/CHANNEL	Activation type	*SWITCHED*
		Speed dial number	N/A
		Remote unit, Remote port number	N/A
		Delete digits	N/A
Called	SLOT/CHANNEL	Port extension number	*107*
		Fwd digits	N/A
		Fwd type	N/A
		Fwd delay	N/A
		Delete digits	N/A
		E&M signaling type	*WINK START*

Table 2-11: Wink Start Signaling Method Parameters

Unit	SE Submenu	Parameter Name	Value
Calling	MAP	Entry digits	*7210*
		Destination name	The called unit name
		Destination extension source	*MAP*
		Destination extension	*107*
		Extended digits source	N/A
		Extended digits to forward	N/A
		Number of user extended digits	N/A

Table 2-11: Wink Start Signaling Method Parameters

When Wink Start signaling is used from the calling side PBX to the calling NetPerformer, or from the called NetPerformer to the called side PBX, the timing between NetPerformer and PBX is as follows:



On the calling side, the PBX raises the M lead. Once the NetPerformer winks using the E lead, the PBX can proceed with dialing after a short waiting period.

On the called side, the NetPerformer raises the M lead. Once the called side PBX winks using the E lead, the NetPerformer can proceed with dialing after a short waiting period. The variable delay is between the M lead going high to the start of the wink. The wink itself is a preset interval of 100 milliseconds.

2.12 Enhanced Dialing Summary

Since several dialing methods are available, it is essential to understand the structure of the dial string that is presented from the PBX to the calling NetPerformer. In addition, you need to know how the calling and the called NetPerformers will parse this string:

- Which portions of the string will be acted upon (used to look up a destination),
- Which portions will be forwarded to other devices.

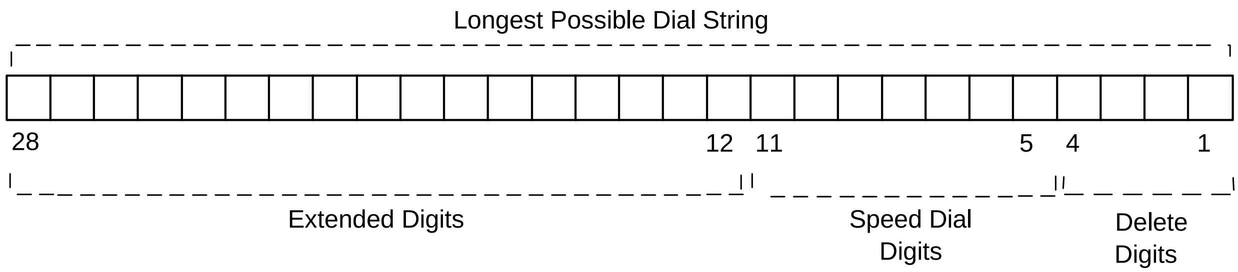


Table 2-12: Enhanced Dialing Summary

As the above illustration shows, the dial string contains a maximum of 28 digits (DTMF or pulse dial). If the Delete Digits parameter is set to any non-zero value, that number of leading digits (up to 4) will be deleted. If Delete Digits is set to **0**, no leading digits are deleted. In this case, the dial string can contain a maximum of 24 digits, where the first digits represent the Speed Dial Number.

The next group of digits is used as the Speed Dial Number, which is variable in length from 1 to 8 digits. The calling NetPerformer looks up these digits in its Voice Mapping Table to determine the destination of the call. The table entry may also specify additional extended digits, with the *Extended digits source* defined as USER or MAP.

If there are no extended digits, then the speed dial number that was dialed will be forwarded to the called NetPerformer. If extended digits (up to 30) are provided in the dial string or if they are present in the *Extended digits to forward* parameter in the Voice Mapping Table, these extended digits will also be forwarded to the called NetPerformer (depending on the setting of the *Extended digits source* parameter).

When the dial string is received at the remote site, the configuration of the called NetPerformer determines whether to forward none of the digits, only the Extended Digits, or the Speed Dial Number plus the Extended Digits to the attached device (usually a PBX).

The minimum user-dialed string is 1 digit in length. Alternatively, the voice port on the calling NetPerformer can be configured for Predefined or Autodial line activation, where no digits at all are needed. In this case, a pre-programmed Speed Dial Number is dialed automatically when the voice channel goes off-hook. Refer to the *Analog Voice* fascicle of this document series for an overview of the Predefined and Autodial line activation types.

NOTE: If a voice port is configured for Autodial line activation, the Delete Digits parameter must be set to **0**. If it is not, the call will not complete, since the autodial voice port will wait for the delete digits to be entered before proceeding. As no digits are entered for an autodial call, the call will time out with a busy signal in 12 seconds.

2.13 Pulse Dial Make/Break Ratios

Using rotary type dialing, a typical make/break ratio is 33/66. This means that when a number is dialed, the contact is made for 33 milliseconds and broken for 66 milliseconds for each successive digit. For example, when you dial a 3, the rotary dial is rotated to the number 3 and then released. What follows is a 66-ms break, a 33-ms make, a 66-ms break, a 33-ms make, a 66-ms break, and then a continuous make (closed loop).

Telephone equipment manufacturers in certain countries have implemented other make/break ratios. To avoid possible misdialing, Memotec has implemented a continuously programmable make/break ratio with a 2-millisecond resolution. The range of possible make/break ratio values is *20/80*, *22/78*, *24/76*, … *80/20*. This range is compatible with all pulse dial equipment found throughout the world. For example, the **33/66** ratio is used in the U.S., Belgium, Denmark, U.K., France, Portugal, and others. The 40/60 ratio is used in Austria, Germany, Italy, Ireland, Sweden, Switzerland, and some other countries.

2.14 DTMF ON/OFF Threshold

Typically, the duration of DTMF (Dual Tone Multi Frequency) tone dialing depends on how quickly you push the buttons on the telephone. In fact, there are two types of DTMF phones:

- A tone of fixed duration is output when a dialing button is pressed, or
- A tone sounds for as long as the dialing button is pressed.

In the latter case, tone duration is typically 100 to 750 milliseconds, and the time between tones is typically 300 to 1500 milliseconds. Much shorter periods are common when using automated dialing such as digits forwarded by a PBX or speed dialing on a telephone. With automated dialing, tones are generally in the range of 60 to 120 milliseconds, and the time between tones is 50 to 150 milliseconds. Some devices can go as fast as 40 milliseconds ON and 40 milliseconds OFF.

To support the fastest speed dialing equipment available, Memotec has implemented a configurable range of 30 to 1000 milliseconds for each of the ON and OFF tone states.

3

Domain Dialing

3.1 NetPerformer Support of Domain Dialing

Domain Dialing is a calling procedure that takes advantage of the hierarchical structure of voice/fax networks for more efficient speed dialing and Voice Mapping Table design. It is designed for very large voice networks, where more than 1000 Speed Dial Numbers would otherwise be required to access all network devices.

NOTE: If you can reach all destination units in your voice network using less than 1000 Speed Dial Numbers, Domain Dialing is not required for your NetPerformer application.

The entire voice network can be configured as set of domains, each with a gateway NetPerformer that handles dialing to other domains. Examples of this are provided in the following section.

The advantage of Domain Dialing is its ability to handle a vast number of clients without overloading the Voice Mapping Tables. Each Voice Mapping Table can be maintained at a reasonable size without compromising the efficiency or size of the voice/fax network.

3.2 Domain Dialing Application Scenarios

Three domains may be set up for a network that spans the Pacific Ocean: USA, CANADA and JAPAN. Several sub-domains are configured in each domain, for example, USA.LA in Los Angeles, and JAPAN.TOK in Tokyo. Each domain accesses its sub-domains through its gateway: USA.GWY, CANADA.GWY and JAPAN.GWY. Each sub-domain can branch further into smaller service areas. The result is a voice network designed as a hierarchy of service levels. This example is illustrated in Figure 3-1.

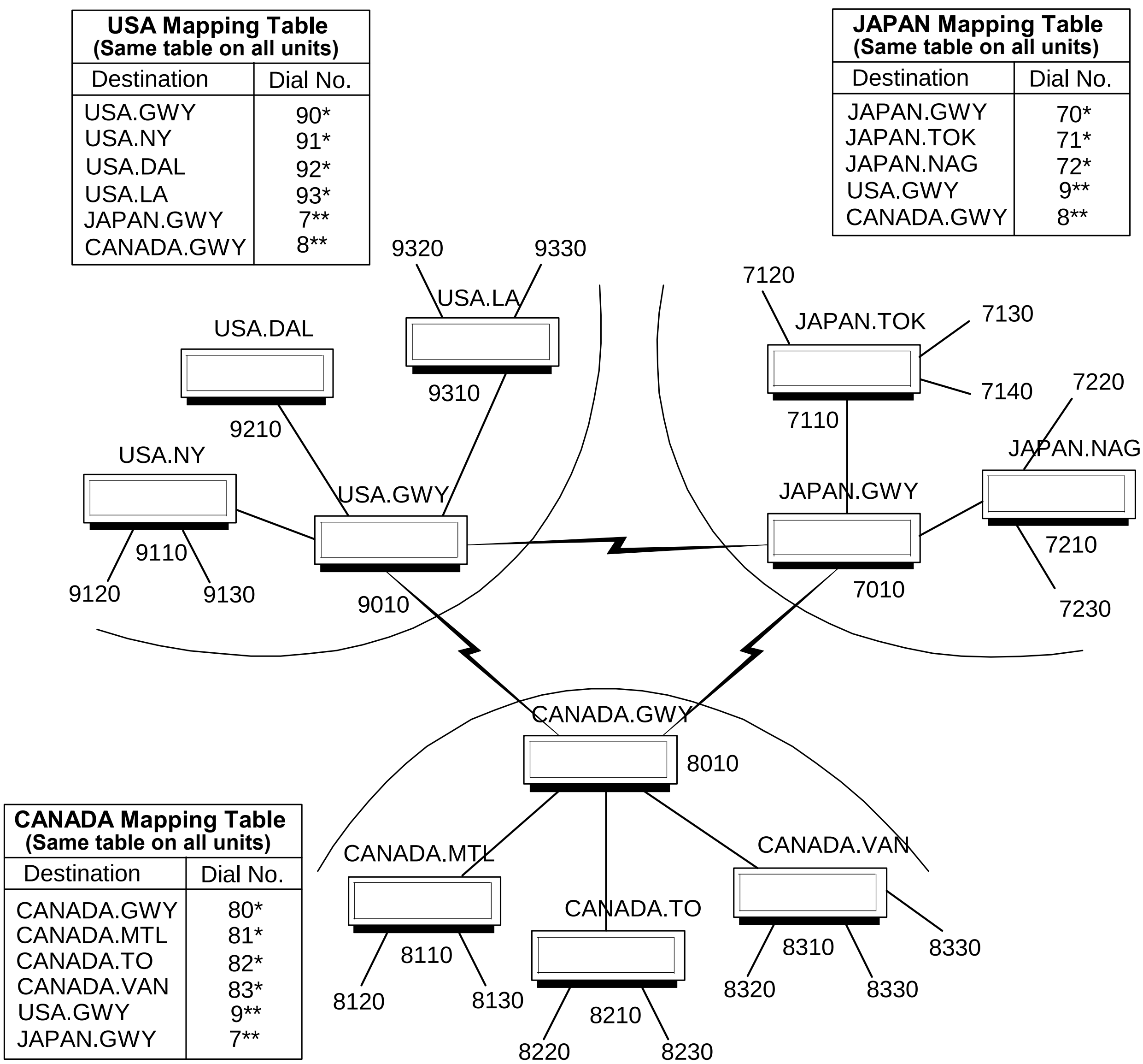


Figure 3-1: Domain Dialing

NOTE: This figure is a conceptual representation of a network that uses Domain Dialing. In an actual application, all units would be linked to the Frame Relay cloud.

3.2.1 Scenario 1

In this network, a caller in Los Angeles (USA.LA) may dial 7113 to reach someone in a service area handled by the NetPerformer in Tokyo (JAPAN.TOK). USA.LA looks in its Voice Mapping Table and determines from the first digit of the speed dial number that the call must be sent to the Japan gateway (JAPAN.GWY).

JAPAN.GWY looks in its Voice Mapping Table and determines from the first two digits of the speed dial number that the call must be sent to the Tokyo sub-domain (JAPAN.TOK). Through a brief negotiation sequence with USA.LA, the Connect Request is sent to JAPAN.TOK.

JAPAN.TOK looks in its Voice Mapping Table and determines from the first three digits of the speed dial number that the call must be established with an end device in its own service area. It sends a Connect Confirm to USA.LA, and the logical connection is opened.

3.2.2 Scenario 2

As a second example, a caller in Toronto (CANADA.TO) may dial 7224. CANADA.TO sends its Connect Request to the Japan gateway (JAPAN.GWY). JAPAN.GWY looks in its Voice Mapping Table and notifies CANADA.TO that the call must be sent to the Nagasaki sub-domain (JAPAN.NAG), as indicated by the first two digits. In turn, CANADA.TO sends back a Connect Request to JAPAN.NAG. Finally, JAPAN.NAG determines from the third digit that the destination of the call is one of its sub-areas, labeled 7220 in Figure 3-1.

3.3 Configuration Tips

To establish Domain Dialing in your network you must plan the domains, sub-domains and service areas carefully. Each area at each level must be distinguished by a significant digit in the speed dial number. Use the wildcard character (*) for non-significant digits. You may also include fully specified speed dial numbers in the Voice Mapping Table for direct Connect Request/Connect Confirm call negotiation. For example, the Voice Mapping Table of USA.LA could contain the number 7113 to permit dialing directly to the specific user in Tokyo. When a number is dialed, the NetPerformer that receives the Connect Request will always choose the best match in its Voice Mapping Table.

NOTE: To enter a sub-domain of another domain in your network, the speed dial number must correspond to a specific branch of that domain. If it does not match, the NetPerformer will try to connect the call locally.

4

Hunt Forwarding

4.1 NetPerformer Support of Hunt Forwarding

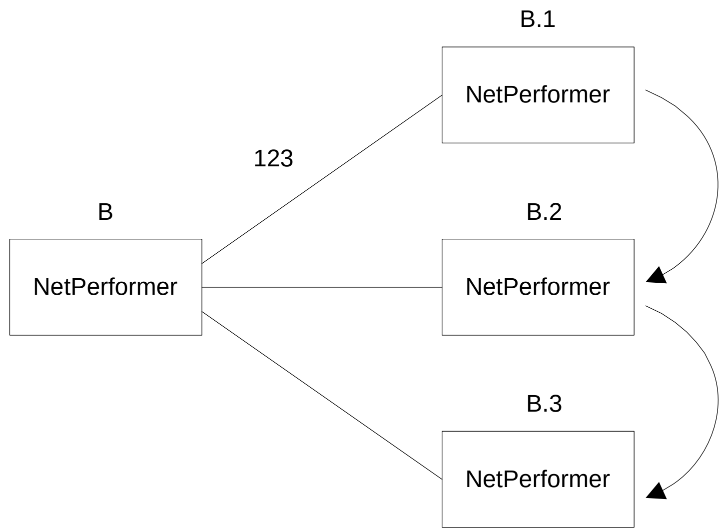
The NetPerformer supports the Hunt Forwarding mechanism, which is mapped on the call forwarding scheme. Coupled with the gateway function, Hunt Forwarding allows a NetPerformer to perform call hunting on multiple units.

Hunt Forwarding defines a specific member of a Hunt Group that should be attempted next when a busy call is received while establishing a switched voice call to that Hunt Group. Hunt Forwarding permits user-defined call attempt sequences in the NetPerformer network. The Hunt Forwarding path can be configured to execute a single call attempt of all units in the Hunt Group, a subset of these units, or a loop that repeats until the DSP timeout is reached.

Without the Hunt Forwarding feature, the calling unit can try to establish a call to any unit belonging to a Hunt Group. In the example below, if Speed Dial Number 123 is defined with Destination Name B.*, NetPerformer B can initiate a call to any remote unit using this Speed Dial Number. If the first member of that group (B.1) is busy, it returns a message to that effect to the calling unit (B). The calling unit (B) then decides which remote unit is the next one to try (B.2).

The drawbacks of this scenario are that the target units need to be located in the same area, and each unit can be attempted only once.

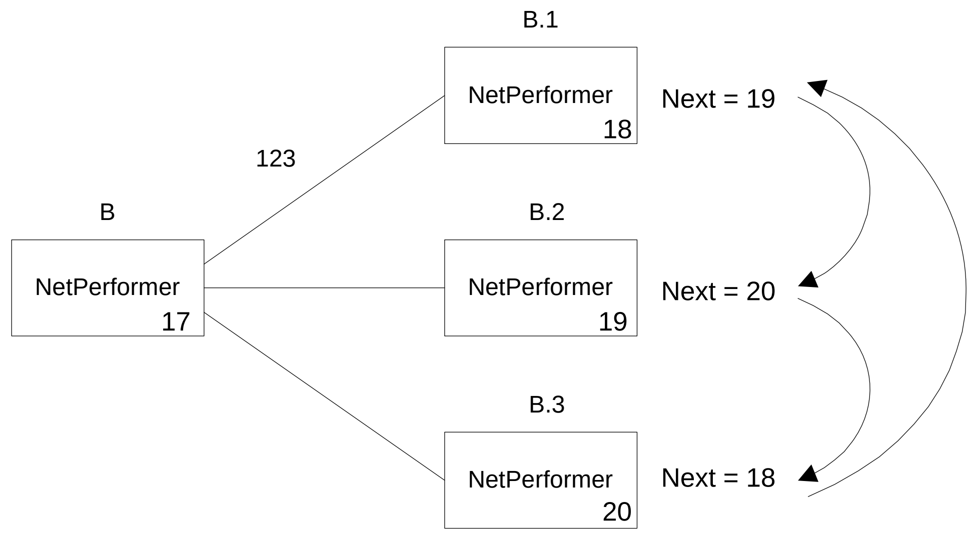
When the NetPerformers are configured for Hunt Forwarding, the calling unit attempts a call to a specific member of the Hunt Group. For example, NetPerformer A in the diagram below may use Speed Dial Number 456, which is defined with Destination Name B. If this call is unsuccessful, the called unit (B) sends back a message indicating which unit to try next. These forwarding messages can be defined to create a continuous loop of call attempts through all remote units in the Hunt Group. In this example, remote unit B specifies unit C as the next destination, unit C specifies unit D and unit D specifies the first remote unit (B) again. A loop is created that continues until the DSP on unit A times out.



Without Hunt Forwarding

- calling unit (B) attempts call to B.*
- Speed Dial Number 123 = B.*
- calling unit (B) decides which remote unit is attempted next
- single attempt of each remote unit in the Hunt Group.

Figure 4-1: Without Hunt Forwarding



With Hunt Forwarding

- calling unit (DLCI 17) attempts call to NetPerformer with DLCI address 18
- called unit (DLCI 18) decides which remote NetPerformer is attempted next (using DLCI address)
- continuous loop through all remote units in the Hunt Group.

Figure 4-2: With Hunt Forwarding

4.2 Configuring the NetPerformer for Hunt Forwarding

In a NetPerformer only network, parameters in the **HUNT** option of the **SETUP** command can be used to define the next unit in the Hunt Forwarding path: *Hunt x Next Destination*, where *x* represents the specific Hunt Group (A to F). These parameters provide the unit name of the member of the desired Hunt Group that should be tried next in the call attempt sequence. A busy call for that Hunt Group will be forwarded to this unit.

NOTE: With Hunt Forwarding, if an intermediary line goes down the final destination may not be reachable. In some cases, Voice Mapping with a wildcard character (for example, *Destination unit* = **B.***) may be a superior solution. An example would be where unit A is connected to city X, and city X is connected to Y with X as the intermediate point. A call defined to go from A to Y via X would not work with Hunt Forwarding, but would be successful if using B.1 for Y and B.2 for X.

4.3 Hunt Group Sorting Rules

In NetPerformer versions prior to V10.2, Hunt Group channels were sorted in the order that they were created:

- Upon termination of a voice call, the voice channel Hunt Group was placed at the end of the list. This process applied to all individual Hunt Groups.
- Due to the random nature of the duration of voice calls, the Hunt Group sorting list was unpredictable, and could lead to potential call collisions between the NetPerformer and the attached voice equipment.

Hunt Group Sorting Rules have been added to the NetPerformer base product in V10.2 at the level of the physical interface (or **LINK**):

- When a sorting rule is selected, it is applied to all Hunt Group channels that send voice traffic over that interface.
- The result is a predictable application of the Hunt Group rules on each slot and for each Hunt Group that has been defined.
- All voice channels that belong to a specific Hunt Group are part of the same linked list.

  For example, if a T1 span is split between Hunt Group A for the first 12 timeslots, and Hunt Group B for the remaining 12 timeslots, the two Hunt Groups (A and B) each have their own sorting list.

NOTE: Hunt Group Sorting Rules are *not available* on a product installed with the SIP VoIP licensed software option.

Four Hunt Group Sorting Rules are available on the NetPerformer base product:

- *Linear Selection Ascending* (**LSA**), described in the next section.
- *Linear Selection Descending* (**LSD**), described on page 4-8
- *Round Robin Ascending* (**RRA**), described on page 4-10
- *Round Robin Descending* (**RRD**), described on page 4-13.

NOTE: The examples in the following sections refer to a T1 interface where all calls are placed from the NetPerformer unit to the PBX.

4.3.1 Linear Selection Ascending (LSA) Order

When the NetPerformer tries to place a call under Linear Selection Ascending (**LSA**) order, **timeslot selection always starts with the lowest numbered timeslot**, and cycles upward to higher numbered timeslots until a free timeslot is found.

Hunt Group Sorting with LSA

Under LSA, when the NetPerformer wants to place a call after receiving a request from the WAN:

- It first attempts to seize timeslot 1 of the interface card
- If timeslot 1 is busy, it attempts to seize timeslot 2, and so on until the first free timeslot is found.

When the next call request comes in:

- Once again, the NetPerformer attempts to seize timeslot 1 first, then timeslot 2, and so on.

Examples of LSA Order

Starting Conditions

The NetPerformer unit is installed with two T1 cards: in Slot 1 and Slot 2. All voice channels belong to Hunt Group A, and all Egress calls request a connection to Hunt Group A.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 1

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: All channels are Idle
- The NetPerformer scans the Hunt Group A list *from left to right*, and selects channel 101.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Seized	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 2

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 101 is Seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from left to right, and selects channel 102
- Channel 101 goes back to Idle state when the first call is completed.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A

Channel	101	102	103	104	124	201	202	203	204	224
Channel State	Idle	Seized	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 3

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 102 is Seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from left to right, and selects channel 101.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Seized	Seized	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 4

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 101 through 124 are seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from left to right, and selects channel 201 on the second T1 interface card.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Seized	Seized	Seized	Seized	Seized	Seized	Idle	Idle	Idle	Idle

4.3.2 Linear Selection Descending (LSD) Order

When the NetPerformer tries to place a call under Linear Selection Descending (**LSD**) order, **timeslot selection always starts with the highest numbered timeslot**, and cycles downward to lower numbered timeslots until a free timeslot is found.

Hunt Group Sorting with LSD

Under LSD, when the NetPerformer wants to place a call after receiving a request from the WAN:

- It first attempts to seize timeslot 24 of the interface card (T1 card)
- If timeslot 24 is busy, it attempts to seize timeslot 23, and so on until the first free timeslot is found.

When the next call request comes in:

- Once again, the NetPerformer attempts to seize timeslot 24 first, then timeslot 23, and so on.

Examples of LSD Order

Starting Conditions

The NetPerformer unit is installed with two T1 cards: in Slot 1 and Slot 2. All voice channels belong to Hunt Group A, and all Egress calls request a connection to Hunt Group A.

Channel	101	102	122	123	124	201	202	222	223	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 1

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: All channels are Idle
- The NetPerformer scans the Hunt Group A list *from right to left*, and selects channel 224.

Channel	101	102	122	123	124	201	202	222	223	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Seized

Example 2

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 224 is Seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from right to left, and selects channel 223
- Channel 224 goes back to Idle state when the first call is completed.

Channel	101	102	122	123	124	201	202	222	223	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Seized	Idle

Example 3

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 223 is Seized, all others are Idle

- The NetPerformer scans the Hunt Group A list from right to left, and selects channel 224.

Channel	**101**	**102**	**122**	**123**	**124**	**201**	**202**	**222**	**223**	**224**
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Seized	Seized

Example 4

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 224 through 201 are seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from right to left, and selects channel 124 on the first T1 interface card.

Channel	**101**	**102**	**122**	**123**	**124**	**201**	**202**	**222**	**223**	**224**
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Seized	Seized	Seized	Seized	Seized	Seized

4.3.3 Round Robin Ascending (RRA) Order

When the NetPerformer tries to place a call under Round Robin Ascending (**RRA**) order, **timeslot selection starts with the timeslot that is one number higher than the last used timeslot on that interface**, and cycles upward to higher numbered timeslots until a free timeslot is found. If the highest numbered timeslot is reached, the cycle continues with timeslot 1.

NOTE: RRA order is the default Hunt Group Sorting Rule on the NetPerformer.

Hunt Group Sorting with RRA

Under RRA, when the NetPerformer wants to place a call after receiving a request from the WAN:

- It first attempts to seize timeslot 1 of the interface card
- If timeslot 1 is busy, it attempts to seize timeslot 2, and so on until the first free timeslot is found.

Suppose that the call was successfully seized on timeslot 22. After the call terminates and the next call request comes in:

- The NetPerformer attempts to seize timeslot 23 first

NOTE: This is the case even if timeslots 1 to 22 are available.

- If timeslot 23 is busy, it attempts to seize timeslot 24
- If timeslot 24 is busy, it starts again at timeslot 1, and keeps attempting higher numbered timeslots until the first free timeslot is found.

Examples of RRA Order

Starting Conditions

The NetPerformer unit is installed with two T1 cards: in Slot 1 and Slot 2. All voice channels belong to Hunt Group A, and all Egress calls request a connection to Hunt Group A.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 1

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: All channels are Idle, and no channels have been previously used
- The NetPerformer scans the Hunt Group A list *from left to right*, and selects channel 101.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Seized	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 2

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 101 is Seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from left to right, and selects channel 102
- Channel 101 goes back to Idle state when the first call is completed.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Seized	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 3

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 102 is Seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from left to right, and selects channel 103.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Seized	Seized	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 4

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: All channels are Idle, the last channel used in Hunt Group A is channel 103
- The NetPerformer scans the Hunt Group A list from left to right, and selects channel 104.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Seized	Idle	Idle	Idle	Idle	Idle	Idle

Example 5

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channels 102, 124, 204 and 224 are seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from left to right, and determines that the last channel on the list (channel 224) is seized
- It starts scanning again from the lowest numbered channel, and selects channel 101.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Seized	Seized	Idle	Idle	Seized	Idle	Idle	Idle	Seized	Seized

4.3.4 Round Robin Descending (RRD) Order

When the NetPerformer tries to place a call under Round Robin Descending (**RRD**) order, **timeslot selection starts with the timeslot that is one number lower than the last used timeslot on that interface**, and cycles downward to lower numbered timeslots until a free timeslot is found. If timeslot 1 is reached, the cycle continues with the highest numbered timeslot on the interface card.

Hunt Group Sorting with RRD

Under RRD, when the NetPerformer wants to place a call after receiving a request from the WAN:

- It first attempts to seize timeslot 24 of the interface card (T1 card)
- If timeslot 24 is busy, it attempts to seize timeslot 23, and so on until the first free timeslot is found.

Suppose that the call was successfully seized on timeslot 3. After the call terminates and the next call request comes in:

- The NetPerformer attempts to seize timeslot 2 first

  This is the case even if timeslots 3 to 24 are available.
- If timeslot 2 is busy, it attempts to seize timeslot 1
- If timeslot 1 is busy, it starts again at timeslot 24, and keeps attempting lower numbered timeslots until the first free timeslot is found.

Examples of RRD order

Starting Conditions

The NetPerformer unit is installed with two T1 cards: in Slot 1 and Slot 2. All voice channels belong to Hunt Group A, and all Egress calls request a connection to Hunt Group A.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle

Example 1

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: All channels are Idle, and no channels have been previously used
- The NetPerformer scans the Hunt group A list *from right to left*, and selects channel 224.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A

Channel	101	102	103	104	124	201	202	203	204	224
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Seized

Example 2

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 224 is Seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from right to left, and selects channel 223
- Channel 224 goes back to Idle state when the first call is completed.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Seized	Idle

Example 3

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channel 223 is Seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from right to left, and selects channel 222.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A
Channel State	Idle	Idle	Idle	Idle	Idle	Idle	Idle	Seized	Seized	Idle

Example 4

- The NetPerformer receives a CONNECT REQUEST for Hunt Group A
- Conditions: Channels 223, 124, 102 and 101 are seized, all others are Idle
- The NetPerformer scans the Hunt Group A list from right to left, and determines that the last channel on the list (channel 101) is seized
- It starts scanning again from the highest numbered channel, and selects channel 224.

Channel	101	102	103	104	124	201	202	203	204	224
Hunt Group	A	A	A	A	A	A	A	A	A	A

Channel	101	102	103	104	124	201	202	203	204	224
Channel State	Seized	Seized	Idle	Idle	Seized	Idle	Idle	Idle	Seized	Seized

4.4 Configuring the NetPerformer for Hunt Group Sorting

Configure the Hunt Group Sorting Rule on each slot using the **SETUP/SLOT/LINK** submenu.

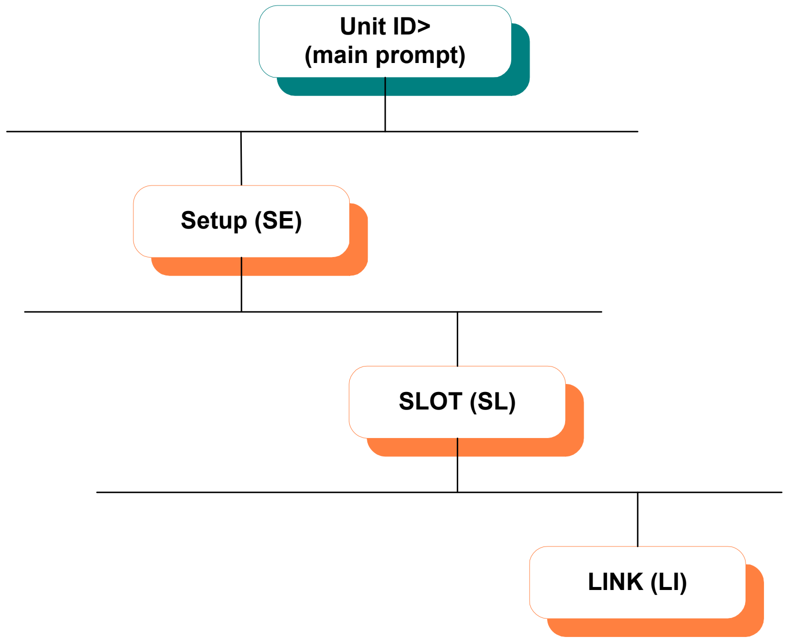


Figure 4-3: SETUP/SLOT/LINK Path on the CLI Tree

► **To configure the Hunt Group Sorting Rule:**

1. At the NetPerformer command line prompt, enter the menu sequence: **SE** ↵ **SLOT**
2. Enter the *Slot number*
3. Enter **LINK**
4. Set the *Status* parameter to **ENABLE**
5. Set *Hunt Group Sorting* to the appropriate sorting order for your application: **LSA**, **LSD**, **RRA** or **RRD**.
6. Change the other parameters from their default values, if desired.

SE/SLOT/#/ LINK 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) ? SLOT
SLOT> Slot number (1/2/4,def:1) ?
Item (LINK/CHANNEL,def:LINK) ?
PORT 100> Status (def:DISABLE) ? ENABLE
PORT 100> Clock recovery (def:ENABLE) ?
```

```
PORT 100> Digital port clock source (def:1) ?
PORT 100> Signaling mode (def:NONE) ? ROB BIT
PORT 100> Pcm encoding law (def:MU-LAW) ?
PORT 100> Hunt Group Sorting (def:RRA) ? ?
CHOICE: LSA LSD RRA RRD

PORT 100> Hunt Group Sorting (Default value:RRA, Current value:RRA) ?
```

Hunt Group Sorting

Console	SNMLP	Test-based Config
Hunt Group Sorting	ifwanHuntRules	[ifwan#] HuntRules

Sets the sorting order that is applied when the NetPerformer searches for an empty voice channel belonging to a Hunt Group on this interface card:

- **LSA:** Linear Selection Ascending, starting at timeslot 1 (see page 4-6)
- **LSD:** Linear Selection Descending, starting at the highest numbered timeslot on the interface card (see page 4-8)
- **RRA:** Round Robin Ascending, starting from the last used timeslot plus 1 (see page 4-10)
- **RRD:** Round Robin Descending, starting from the last used timeslot minus 1 (see page 4-13).

NOTE: All Hunt Groups on an interface card are searched using the same sorting rule. However, the starting point for RRA and RRD is maintained separately for each Hunt Group list.

Values: *LSA, LSD, RRA, RRD*

Default: *RRA*

5

Voice Traffic Routing

5.1 About Voice Traffic Routing

The Voice Traffic Routing (VTR) feature of the NetPerformer permits local control of how voice traffic will be forwarded to its intended destination. With VTR, alternate MAP entries are used in sequence if a call cannot be established using the first MAP entry. Various routes using local and long distance services can thus be constructed at the local site. Call attempts using matching MAP entries can be looped until a successful connection is made.

5.2 Virtual Connections Technology

VTR is based on the concept of *Virtual Connections*, which is proprietary to the NetPerformer product line.

- Virtual Connections control the WAN aspects of the network, including routing, alternate routing, load balancing, bandwidth on demand (on private networks) and dial back-up.
- VTR routes the voice call based on the digits dialed by the user, and selects the intended destination using the destination Unit Name.

5.2.1 Routing Functions

With VTR you can define a numbering plan for routing calls on the WAN transport to the destination node, or even locally on the originating node.

- Routing of the voice call includes the ability to search for alternate nodes depending on the status of the network (such as link down, all channels busy, and so on).
- VTR also includes dialed digits manipulation, which is dependent on the destination node that is used for the outbound portion of the voice call.

5.2.2 Uses and Goals

VTR is particularly useful for call back operators (or alternate carrier). Such operators offer international voice traffic transport through private or public Voice over Packet networks, e.g. VoFR and VoIP.

- To offer an attractive service, call back operators should be able to route the call to alternate destinations in case the primary destination is unreachable.
- The goal is to complete as many calls as possible, providing a transparent interface to the caller.
- In some cases, the service may even revert to use of the Public Switched Telephone Network at the originating or destination node in order to complete a call.
- This transparent approach lets the user dial the same number no matter what service is used, without having to try different access codes for the different routes that may be available at a particular time.

5.2.3 VTR versus LCR

VTR is sometimes referred to as Least Cost Routing (LCR) by some users. However, the VTR function on the NetPerformer is intended to supplement an LCR service that may already be implemented on some PBXs.

NOTE: A full-featured LCR might imply using an external server that is capable of providing a route with the least cost, based on a per minute rate as a function of the time of day. This is not part of an embedded system such as VTR on the NetPerformer.

5.3 Other Rerouting Methods

In a NetPerformer-based network, traffic can be rerouted in three ways:

- Unit Name containing a wildcard character:
  - The wildcard character allows for a hunt of multiple destinations.
  - For example, a MAP entry could be defined with the destination name BOSTON.* and assigned to Hunt Group A.
  - A call will be attempted to all ports on unit BOSTON.1 that belong to Hunt Group A.
  - If all of these ports are busy or unreachable, the ports on unit BOSTON.2 are attempted next.
  - The hunt continues through all units named BOSTON.* until an available port is found to place the call.
- Domain Dialing: refer to the chapter “Domain Dialing” on page 3-1.
  - The entire telephony network is configured as set of domains, each with a gateway NetPerformer that handles dialing to other domains.
  - Several sub-domains are configured in each domain, and each sub-domain can branch further into smaller service areas.
  - The result is a telephony network designed as a hierarchy of service levels handling a vast number of clients without overloading the Voice Mapping Tables.
  - Ideal for very large networks where over 1000 speed dial numbers would otherwise be required.
- Hunt Forwarding: refer to “Hunt Forwarding” on page 4-1.
  - Permits call forwarding to and from legacy NetPerformer products.
  - Defines a specific member of a Hunt Group that should be attempted next when a busy call is received while establishing a switched voice call to that Hunt Group.
  - The Hunt Forwarding path can be configured to execute a single call attempt of all units in the Hunt Group, a subset of these units, or a loop that repeats until a predefined timeout is reached.

5.3.1 Remote Versus Local Rerouting

Both Domain Dialing and Hunt Forwarding are *remote rerouting methods*. The decision as to where to reroute the call is taken at the remote end. For these rerouting methods, a call cannot be completed if the link goes down, since the remote unit must be able to send either a busy indication or an alternate destination decision to the local unit.

VTR, on the other hand, is a *local rerouting method*. Its goal is to reroute a call without depending on information provided by the remote unit. It is thus able to place a call even if a link to a particular destination is down. Multiple call attempts are not required.

NOTE: Rerouting using a wildcard character also relies on a local unit decision as to the next unit or port to attempt in the Hunt Group. The local unit searches its Voice Mapping Table for all destinations that match the wildcard, and tries them one by one.

5.3.2 Optional Methods

The three rerouting methods, Domain Dialing, Hunt Forwarding and Voice Traffic Routing, can be controlled by different units in the network (local versus remote). In NetPerformer V9.1.0 and higher, these rerouting methods can be enabled or disabled separately on each unit. Three new configuration parameters, described in Table 5-1, have been added to the Setup Global menu to control whether a particular method is available.

Parameter	SNMP Variable	Values	Function
Enable VTR	*sysEnableVtr*	*YES*, *NO* (def.)	Enable (YES) or disable (NO) Voice Traffic Routing on this unit.
Enable Domain Dialing	*sysEnableDomain*	*YES* (def.), *NO*	Enable (YES) or disable (NO) Domain Dialing on this unit.
Enable Hunt Forwarding	*sysEnableHuntFwd*	*YES* (def.), *NO*	Enable (YES) or disable (NO) Hunt Forwarding on this unit.

Table 5-1: Global Parameters for Voice Routing

Note that Domain Dialing and Hunt Forwarding are enabled by default, whereas VTR is disabled by default.

- For most applications, if you enable VTR you should disable Domain Dialing and Hunt Forwarding.
  - With VTR it is the local unit that decides which route will be used and which digit stream must be forwarded, depending on whether the call is local or long distance. The prefix digit stream is stored in the MAP entry.
  - With Domain Dialing and Hunt Forwarding the local Voice Mapping Table is not used to reroute voice calls. This could lead to undesirable results if VTR is also enabled.
- For some applications, however, it may be convenient to have either Domain Dialing or Hunt Forwarding enabled at the same time as VTR.
  - **VTR and Domain Dialing:** If a domain gateway indicates to the local unit that it cannot complete a call, the local unit will send a new connection request to a new destination, using the same extended digits as for the initial call attempt.
  - **VTR and Hunt Forwarding:** If the local unit receives a busy indication containing a next destination name, it will send a new connection request to

this destination rather than use another MAP entry. The extended digits for this connection will be the same as for the initial call attempt.

5.4 How VTR Works

When VTR is enabled, you can configure more than one MAP entry with the same speed dial number.

- The best match in the Voice Mapping Table will be used first.
- Other matching speed dial numbers will be used sequentially until no match is possible or the call is established.
- All digits, including the speed dial number and any extended digits, are sent to the remote unit, which can decide whether to forward the extended digits to the attached equipment.
- A call can be routed to a local port, even if previous attempts have been made to remote destinations.
- Attempts of all matching speed dial numbers can be looped, using a new extended parameter, VTRLOOP. This repeats the call sequence if the call was not established after all MAP entries are used. See “Global Parameters” on page 5-13.

5.5 Configuring Voice Mapping Tables

Some changes were made to the Voice Mapping Table in NetPerformer V9.1.0 and V9.2.0 to support VTR configuration. These changes are noted below.

5.5.1 Adding a MAP Entry

To accommodate multiple MAP entries with the same speed dial number, you are prompted for the speed dial number when you add another entry.

- Enter **SE** at the NetPerformer console command line.
- Enter **MAP**.
- Enter **ADD** to add a new entry.
- Enter the *Speed Dial Number* that you want to use for this entry. Use * (asterisk) as a wildcard character, if desired.

NOTE: In NetPerformer V9.1.0, the required number of digits was determined by the global *Speed dial number (no. of digits)* parameter. **In NetPerformer V9.2.0 and higher, you can enter a speed dial number of any length (maximum 8 digits).**

Once you enter the speed dial number, the NetPerformer console responds with a list of all MAP entries that have already been created with this speed dial number. Each MAP entry is numbered, as in this example where a MAP entry with Destination Name MONTREAL is being added. It has a speed dial number that is already used for 3 other entries.

SE/MAP/ADD example

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? MAP
MAP> Operation (ADD/MODIFY/DELETE,def:ADD) ? ADD
MAP> Speed Dial Number? : 001212
MAP #1> Speed Dial Number...................... 001212
MAP #1> Destination Name....................... NEW-YORK
MAP #1> Destination Extension number........... HUNT A
MAP #1> Extended digits to forward............. 9,!
MAP #2> Speed Dial Number...................... 001212
MAP #2> Destination Name....................... PHILADELPHIA
MAP #2> Destination Extension number........... HUNT A
MAP #2> Extended digits to forward............. NONE
MAP #3> Speed Dial Number...................... 001212
MAP #3> Destination Name....................... PARIS
MAP #3> Destination Extension number........... 3456
MAP #3> Extended digits to forward............. 001212!
MAP> MAP Number to add? : 3
MAP> Destination Name? : MONTREAL
```

```
MAP> Remote extension number source (HUNT/USER/MAP,def:HUNT) ?
MAP> Hunt group (A/B,def:A) ? B
MAP> Extended digits source (NONE/USER/MAP,def:NONE) ? MAP
MAP> Extended digits to forward? : 9,1212!
MAP> Add another map entry (NO/YES,def:NO) ?
```

- Enter the number of the MAP entry you want to add.
  - If you enter a new number, the MAP entry will be placed in the list in the appropriate sequence.
  - If, as in the above example, you enter a number that is already being used for another entry, the original entry will be shifted down to the next number. The entry you are currently defining will have the number you select.
- Enter the appropriate values for the following:
  - *Destination Name*. The name of the remote unit. Use * (asterisk) as a wildcard character, if desired.
  - *Remote extension number source*: HUNT, USER or MAP.
  - For MAP extension number source, enter the *Destination Extension number*. The required number of digits is determined by the global *Extension number (no. of digits)* parameter.
  - For HUNT extension number source, enter the *Hunt group*: A or B.
  - *Extended digits source*: NONE, USER or MAP. (default NONE).
  - For MAP extended digits source, enter the *Extended digits to forward*. The maximum number of digits is determined by the global *Extension number (no. of digits)* parameter. Use the wildcard character ! (exclamation mark) to concatenate user-dialed digits to the extended digits. During call setup the ! is replaced by the extended digits dialed by the user.

NOTE: The ! wildcard character cannot be used in the first MAP entry. The wildcard character * (asterisk) *can* be used in any speed dial number.

In our example, the result of the ADD operation is 4 MAP entries that use the same speed dial number, with the newest entry appearing as MAP #3. The original MAP #3 (with destination PARIS) now appears as MAP #4.

DMF example

```
BOSTON>DMF
DISPLAY MAP FILE
MAP #1> Speed Dial Number...................... 001212
MAP #1> Destination Name....................... NEW-YORK
MAP #1> Destination Extension number........... HUNT A
MAP #1> Extended digits to forward............. 9,!
MAP #2> Speed Dial Number...................... 001212
MAP #2> Destination Name....................... PHILADELPHIA
MAP #2> Destination Extension number........... HUNT A
MAP #2> Extended digits to forward............. NONE
MAP #3> Speed Dial Number...................... 001212
MAP #3> Destination Name....................... MONTREAL
```

```
MAP #3> Destination Extension number........... HUNT B
MAP #3> Extended digits to forward............. 9,1212!
MAP #4> Speed Dial Number...................... 001212
MAP #4> Destination Name....................... PARIS
MAP #4> Destination Extension number........... 3456
MAP #4> Extended digits to forward............. 001212!
```

5.5.2 Deleting a MAP Entry

To delete a MAP entry, use the DELETE operation of the SETUP/MAP menu. You must specify the *Speed Dial Number* and the *MAP Number to delete*.

In this example the MAP entry with *Destination Name* MONTREAL (MAP #3)is deleted.

SE/MAP/ DELETE example

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? MAP
MAP> Operation (ADD/MODIFY/DELETE,def:ADD) ? DELETE
MAP> Speed Dial Number? : 001212
MAP #1> Speed Dial Number...................... 001212
MAP #1> Destination Name....................... NEW-YORK
MAP #1> Destination Extension number........... HUNT A
MAP #1> Extended digits to forward............. 9,!
MAP #2> Speed Dial Number...................... 001212
MAP #2> Destination Name....................... PHILADELPHIA
MAP #2> Destination Extension number........... HUNT A
MAP #2> Extended digits to forward............. NONE
MAP #3> Speed Dial Number...................... 001212
MAP #3> Destination Name....................... MONTREAL
MAP #3> Destination Extension number........... HUNT B
MAP #3> Extended digits to forward............. 9,1212!
MAP #4> Speed Dial Number...................... 001212
MAP #4> Destination Name....................... PARIS
MAP #4> Destination Extension number........... 3456
MAP #4> Extended digits to forward............. 001212!
MAP> MAP Number to delete? : 3
MAP> Delete another map entry (NO/YES,def:NO) ? NO
```

As a result of this operation, the original MAP #4 has shifted up to MAP #3, taking the place of the entry that was deleted.

DMF example: after deletion

```
BOSTON>DMF
DISPLAY MAP FILE
MAP #1> Speed Dial Number...................... 001212
MAP #1> Destination Name....................... NEW-YORK
MAP #1> Destination Extension number........... HUNT A
MAP #1> Extended digits to forward............. 9,!
MAP #2> Speed Dial Number...................... 001212
```

```
MAP #2> Destination Name...................... PHILADELPHIA
MAP #2> Destination Extension number........... HUNT A
MAP #2> Extended digits to forward............. NONE
MAP #3> Speed Dial Number...................... 001212
MAP #3> Destination Name...................... PARIS
MAP #3> Destination Extension number........... 3456
MAP #3> Extended digits to forward............. 001212!
```

5.5.3 Modifying a MAP Entry

To modify an existing MAP entry, use the MODIFY operation of the SETUP/MAP menu. You must specify the *Speed Dial Number* and the *MAP Number to modify*.

In this example the MAP entry with *Destination Name* PARIS has its extension number replaced by a hunt group.

SE/MAP/ MODIFY example

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? MAP
MAP> Operation (ADD/MODIFY/DELETE,def:ADD) ? MODIFY
MAP> Speed Dial Number? : 001212
MAP #1> Speed Dial Number...................... 001212
MAP #1> Destination Name...................... NEW-YORK
MAP #1> Destination Extension number........... HUNT A
MAP #1> Extended digits to forward............. 9,!
MAP #2> Speed Dial Number...................... 001212
MAP #2> Destination Name...................... PHILADELPHIA
MAP #2> Destination Extension number........... HUNT A
MAP #2> Extended digits to forward............. NONE
MAP #3> Speed Dial Number...................... 001212
MAP #3> Destination Name...................... PARIS
MAP #3> Destination Extension number........... 3456
MAP #3> Extended digits to forward............. 001212!
MAP> MAP Number to modify? : 3
MAP #1> Destination Name (def: PARIS) ?
MAP> Remote extension number source (HUNT/USER/MAP,def: MAP) ? HUNT
MAP> Hunt group (A/B,def:B) ? A
MAP> Extended digits source (NONE/USER/MAP,def:MAP) ?
MAP #1> Extended digits to forward (def:9,1212!) ?
MAP> Modify another map entry (NO/YES,def:NO) ? NO
```

The new MAP file contents show the modified PARIS entry.

DMF example: after modification

```
BOSTON>DMF
DISPLAY MAP FILE
MAP #1> Speed Dial Number...................... 001212
MAP #1> Destination Name...................... NEW-YORK
MAP #1> Destination Extension number........... HUNT A
```

```
MAP #1> Extended digits to forward............. 9,!
MAP #2> Speed Dial Number...................... 001212
MAP #2> Destination Name....................... PHILADELPHIA
MAP #2> Destination Extension number........... HUNT A
MAP #2> Extended digits to forward............. NONE
MAP #3> Speed Dial Number...................... 001212
MAP #3> Destination Name....................... PARIS
MAP #3> Destination Extension number........... HUNT A
MAP #3> Extended digits to forward............. 001212!
```

5.6 Voice Traffic Routing Parameters

5.6.1 Global Parameters

Parameters in the SETUP/GLOBAL menu that affect VTR are described in Table 5-2.

Parameter	SNMP Variable	Values	Function
Speed Dial Number (no. of digits) NetPerformer V9.1.0 only	*sysSpeedDial-NumLength*	*1 - 8, VAR* (def. *4*)	Defines the number of digits that must be used to define a speed dial number in the Voice Mapping Table. VAR allows for variable length numbers. **In NetPerformer V9.2.0 and higher this parameter is no longer required.** You can enter a speed dial number of any length, up to a maximum of 8 digits.
Extended digits (no. of digits) NetPerformer V9.1.0 only	*sysExtended-DigitsLength*	*0 - 30*	Determines the maximum number of digits in the extended number that will be forwarded to the attached equipment at the remote end. When VTR is used, this parameter should be set to its maximum value. The dial timer can be used to stop collecting the extended digits to forward. **In NetPerformer V9.2.0 and higher this parameter has been replaced by *Number of user extended digits* in the SETUP/ MAP submenu**.
Enable VTR	*sysEnableVtr*	*YES*, *NO* (def.)	Enable (YES) or disable (NO) Voice Traffic Routing on this unit. This parameter takes effect at the originating end of the connection.
Enable Domain Dialing	*sysEnableDomain*	*YES* (def.), *NO*	Enable (YES) or disable (NO) Domain Dialing on this unit. This parameter takes effect at the receiving end of the connection.
Enable Hunt Forwarding	*sysEnableHuntFwd*	*YES* (def.), *NO*	Enable (YES) or disable (NO) Hunt Forwarding on this unit. This parameter takes effect at the originating end of the connection.

Table 5-2: Global Parameters for VTR

5.6.2 Extended Parameters

One extended parameter in the **EP GLOBAL** group also affects VTR operations: VTRLOOP.

Parameter	Values	Function
VTRLOOP	*YES*, *NO* (def.)	Allows repeating the call sequence in a loop if the call was not established after trying all MAP entries. The loop continues until the call is established or a call timer has expired (approximately 11 seconds).

Table 5-3: Extended Parameters for VTR

5.7 VTR Application

This section provides an example of three different dialing scenarios in an international enterprise network. In the example shown in Figure 5-1, users in Paris can call area code 212 (New York) or 514 (Montreal), as well as specific phones connected on the PBX in Montreal having extension numbers 294, 295 and 296. The MAP table in Paris contains a set of MAP entries for each scenario.

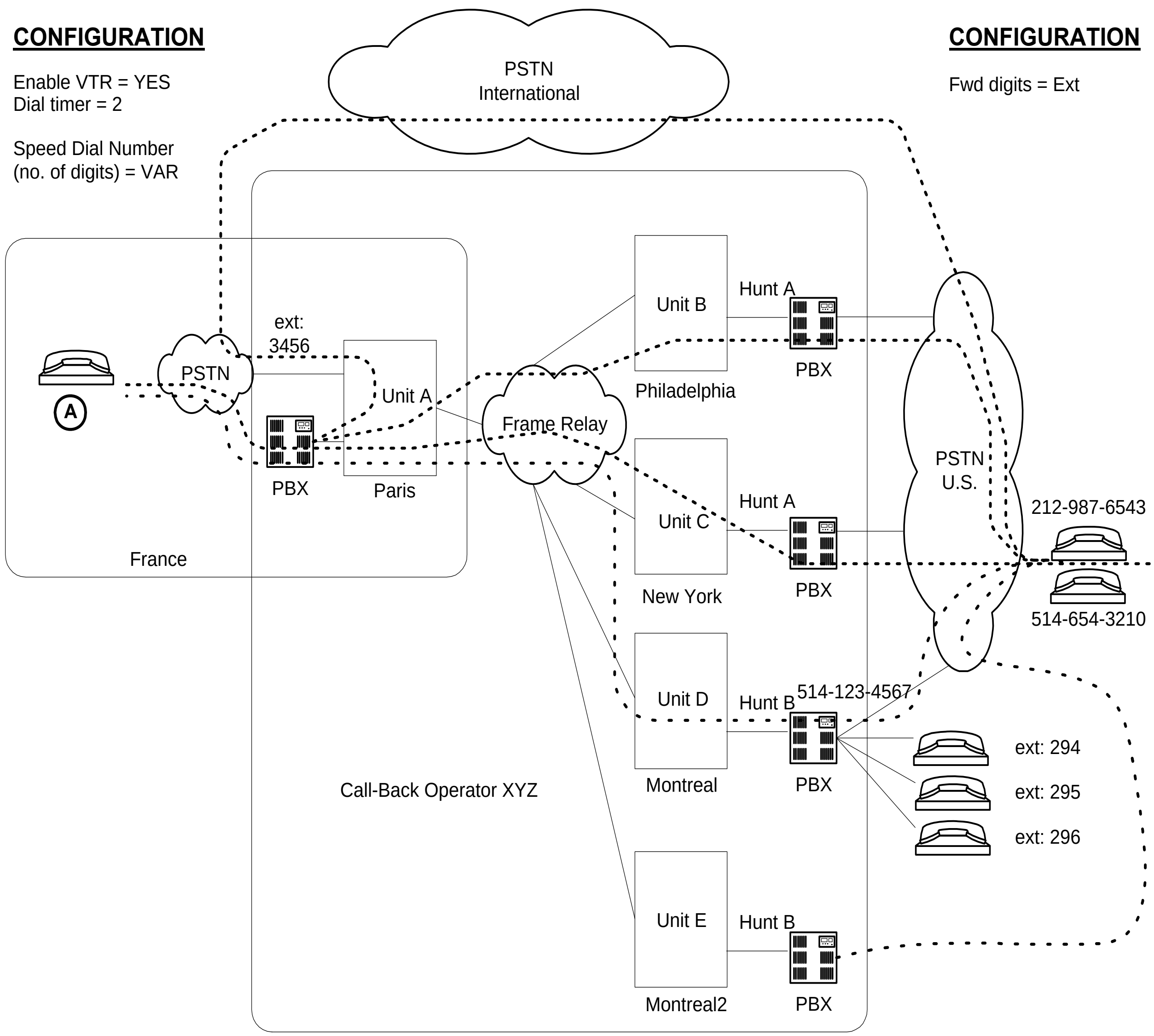


Figure 5-1: VTR Application Example

5.7.1 First Dialing Scenario

The goal of the first dialing scenario is to reach any phone number in New York (area code 212) from Paris using the best available path.

Alternate Routes

To permit the cost savings of a local call, the first attempt should go to the NetPerformer in New York. If this unit is busy or unreachable the successive alternate paths are the following:

- Second attempt to the NetPerformer in Philadelphia.
- Third attempt to the NetPerformer in Montreal.
- Final attempt via the local PSTN in Paris and the international PSTN.

MAP Entries

The NetPerformer in Paris requires the following entries in its Voice Mapping Table:

MAP Entry	Speed Dial Number	Destination	Extended Digits to Forward
#1	001212	NEW-YORK	9,!
#2	001212	PHILADELPHIA	9,1212!
#3	001212	MONTREAL	9,1212!
#4	001212	PARIS	001212!

Table 5-4: Voice Mapping Table

NOTE: The user needs to dial only one number to reach New York, 001212, no matter which route is chosen. The user may also dial extended digits after the speed dial number. The exclamation mark ! in the *Extended digits to forward* field is replaced by the extended digits dialed by the user.

VTR Operation

In this scenario, VTR operates in the following way:

- The user (A) dials a local phone number in France to reach the PBX of the call back operator in Paris. This PBX returns a dial tone to the user's phone.
- The user enters a phone number, for example, 8-001-212-987-6543.
  - The PBX recognizes the digit 8 as a request to send the call to the Voice over Frame Relay network.
  - It picks up a free line on the NetPerformer in Paris and forwards 0012129876543.
  - In this case the first 6 digits (001212) match the speed dial number of 4 alternate entries in the MAP table, given on page 5-16.
- The NetPerformer in Paris attempts the best match first (MAP Entry #1), and tries to establish a call to the PBX in New York. In this MAP entry, the extended number is 9,! which the NetPerformer interprets as:

  - 9 is the beginning of the extended number.
  - The comma , indicates that a pause is required. The duration of the pause is determined by the value of the *Fwd Delay* parameter on the remote unit voice port.
- The NetPerformer in Paris concatenates the user-dialed digits, 9876543, with the *Extended digits to forward* that have been configured in the MAP entry (9,!). The resulting extended number is 9 <PAUSE> 987-6543.

  The NetPerformer sends this extended number in the Connect Request for the call.
- The remote NetPerformer in New York must have the *Fwd Digits* parameter on the voice port set to EXT to be able to forward the extended number to the attached equipment.

  The NetPerformer in New York sends the extended number to the attached PBX to establish a local call through the PSTN.
- If the NetPerformer in Paris cannot open a connection to the PBX in New York (for example, no line is available or the trunk is down), it will attempt the second MAP entry that matches speed dial number 001212.
  - It tries to establish a call to the PBX in Philadelphia.
  - Using this MAP entry the extended number is 9 <PAUSE> 1 (212) 987-6543, which defines a long distance call that the Philadelphia PBX can dial to reach New York.
- If the NetPerformer in Paris is unable to open a connection to the PBX in Philadelphia, it will use the third match (MAP Entry #3).
  - It tries to establish a call to the PBX in Montreal.
  - The extended number is once again 9 <PAUSE> 1 (212) 987-6543, which defines a long distance call that the Montreal PBX can dial to reach New York.
- If the PBX in Montreal cannot be reached, for example because all lines are busy, the NetPerformer in Montreal will respond with an indication that MONTREAL2 is the next destination.
  - If on the unit in Paris the global parameter Enable Hunt Forwarding is set to YES, a call will be attempted to the MONTREAL2 unit.
  - In this case, the extended digits that were last used are repeated: 9 <PAUSE> 1 (212) 987-6543.
- If this attempt also fails, the NetPerformer in Paris will use the next MAP entry that matches the speed dial number, MAP Entry #4.
  - It tries to establish a call using one of the free ports to the local PSTN in France.
  - Using this MAP entry the extended number is 001 (212) 987-6543, which defines an international call to reach New York.

5.7.2 Second Dialing Scenario

The goal of the second dialing scenario is to reach any phone number in Montreal (area code 514) from Paris using the best available path.

Alternate Routes

To permit the cost savings of a local call, the first attempt should go to the NetPerformer in Montreal. If this unit is busy or unreachable the successive alternate paths are as follows:

- Second attempt to the NetPerformer in New York.
- Third attempt to the NetPerformer in Philadelphia.
- Final attempt via the local PSTN in Paris and the international PSTN.

MAP Entries

The NetPerformer in Paris requires the following entries in its Voice Mapping Table:

MAP Entry	Speed Dial Number	Destination	Extended Digits to Forward
#5	001514	MONTREAL	9,!
#6	001514	NEW-YORK	9,1514!
#7	001514	PHILADELPHIA	9,1514!
#8	001514	PARIS	001514!

Table 5-5: Voice Mapping Table

NOTE: The user needs to dial only one number to reach Montreal, 001514, no matter which route is chosen.

If these were the only MAP entries in the Voice Mapping Table, they would be numbered from #1 to #4. MAP entry numbers are provided for reference purposes only, and are always numbered from 1 to *n*. They are generated by the NetPerformer for the specific console display, and do not form an integral part of the MAP entry definition.

VTR Operation

In this scenario, VTR operates in the following way:

- The user (A) dials a local phone number in France to reach the PBX of the call back operator in Paris. This PBX returns a dial tone to the user’s phone.
- The user enters a phone number, for example, 8-001-514-654-3210.
  - The PBX recognizes the digit 8 as a request to send the call to the Voice over Frame Relay network.
  - It picks up a free line on the NetPerformer in Paris and forwards 0015146543210.
  - The first 6 digits (001514) match the speed dial number of 4 alternate entries in the MAP table, given above.
- The subsequent steps are similar to those described for Scenario 1, starting on page 5-16.

5.7.3 Third Dialing Scenario

The goal of the third dialing scenario is to reach any extension number on the PBX in Montreal from Paris using the best available path. The extension numbers in Montreal are numbered from 200 to 299. Only one call is attempted to the unit in Montreal.

Alternate Routes

To permit the cost savings of a local call, the first attempt should go to the NetPerformer in Montreal. If this unit is busy or unreachable a second attempt is made to the NetPerformer in New York.

MAP Entries

The NetPerformer in Paris requires the following entries in its Voice Mapping Table:

MAP Entry	Speed Dial Number	Destination	Extended Digits to Forward
#9	514	MONTREAL	USER
#10	514	NEW-YORK	9,15141234567,!

Table 5-6: Voice Mapping Table

NOTE: The user needs to dial only one number to reach Montreal, 514, no matter which route is chosen.

VTR Operation

- The user (A) dials a local phone number in France to reach the PBX of the call back operator in Paris. This PBX returns a dial tone to the user's phone.
- The user enters a phone number, for example, 8-514-295.
  - The PBX recognizes the digit 8 as a request to send the call to the Voice over Frame Relay network.
  - It picks up a free line on the NetPerformer in Paris and forwards 514295.
  - The first 3 digits (514) match the speed dial number of 2 alternate entries in the MAP table, given above.
- The NetPerformer in Paris attempts the best match first (MAP Entry #9), and tries to establish a call to the unit in Montreal.

  In this MAP entry the *Extended digits source* is USER, which means it must be provided by the user.
- The NetPerformer in Paris inserts the user-dialed digits, 295, in the extended number, and sends this extended number to the Montreal unit in the Connect Request for the call.
- If the NetPerformer in Paris is able to open a connection to Montreal, the NetPerformer in Montreal sends the extended number, 295, to the attached PBX.
  - The remote NetPerformer in Montreal must have the *Fwd Digits* parameter on the voice port set to EXT to be able to forward the extended number to the attached equipment.

- When the attached PBX receives the extended number, it tries to put the call through to extension 295.

- If the NetPerformer in Paris cannot open a connection to the PBX in Montreal (for example, no line is available or the trunk is down), it will attempt the second MAP entry that matches speed dial number 514.
  - It tries to establish a call to the PBX in New York.
  - Using this MAP entry the extended number is 9 <PAUSE> 1 (514) 123-4567 <PAUSE> 295, which defines a long distance call that the New York PBX can dial to reach Montreal.
  - The first digit, 9 is used to seize a line on the PSTN. It is followed by a pause.
  - The number 1 (514) 123-4567 is used to reach the PBX in Montreal via the PSTN. It is also followed by a pause.
  - The PBX in Montreal uses the last three digits to put the call through to extension 295.

5.7.4 Combining VTR with a Hunt Group

Each dialing attempt during a VTR scenario can be combined with the Hunt Group function.

- Set the *Remote extension number source* for the MAP entry to HUNT.
- Define the *Hunt group* as A or B.
- Each available voice port in the Hunt Group will be attempted in turn before attempting a different destination unit.
- Dialing attempts that begin via a local PSTN will require a specific extension number rather than a Hunt Group.
  - Set the *Remote extension number source* for the MAP entry as MAP.
  - Define the *Destination extension number* with the specific extension number you want to use.

MAP Entries

When combined with the Hunt Group function, the NetPerformer in Paris will have the following entries in its Voice Mapping Table:

MAP Entry	Speed Dial Number	Destination	Hunt Group or Ext. No.	Extended Digits to Forward
#1	001212	NEW-YORK	HUNT A	9,!
#2	001212	PHILADELPHIA	HUNT A	9,1212!
#3	001212	MONTREAL	HUNT B	9,1212!
#4	001212	PARIS	3456	001212!
#5	001514	MONTREAL	HUNT B	9,!
#6	001514	NEW-YORK	HUNT A	9,1514!

Table 5-7: Voice Mapping Table

MAP Entry	Speed Dial Number	Destination	Hunt Group or Ext. No.	Extended Digits to Forward
#7	001514	PHILADELPHIA	HUNT A	9,1514!
#8	001514	PARIS	3456	001514!
#9	514	MONTREAL	HUNT B	USER
#10	514	NEW-YORK	HUNT A	9,15141234567,!

Table 5-7: Voice Mapping Table

5.7.5 Configuration Notes

- In all of the preceding examples, the digit string that is forwarded to the attached equipment at the remote end is always:
  - A prefix containing the international code, country code and area code, and
  - The local phone number.
- The *Speed Dial Number* is not used at the remote end. Since it is only used locally, you must ensure that the remote voice port has its *Fwd Digits* parameter set to EXT.

  This is the only way to ensure that the user-dialed digits will be forwarded to the equipment attached to the NetPerformer at the remote site.
- To design your speed dial numbering scheme for maximum efficiency:
  - The *Delete Digits* parameter on a voice port determines the number of leading digits that will be deleted from the *Speed Dial Number*. It must not be used to delete digits that are part of the speed dial number. For example, if all speed dial numbers begin with the international code 001, do not set the *Delete Digits* parameter to 2.
  - Make sure each *Speed Dial Number* begins with a unique sequence that cannot be confused with the international code or country code. For example, if 001514 is the *Speed Dial Number* for Montreal, do not use the sequence 001 to define another *Speed Dial Number*.

6

Supplementary Services on an Analog Interface

On the SDM-9XXX Series, the following supplementary services are now available on an analog interface:

- *Generation of billing signals on an FXS channel (see next section)*
- *Retransmission of Caller ID (ANI) over an FXS interface ("Retransmission of Caller ID over an FXS Interface" on page 6-6)*
- *Detection of Caller ID (ANI) on an FXO interface ("Detection of Caller ID on an FXO Interface" on page 6-8)*

6.1 Generation of Billing Signals on an FXS Channel

The SDM-9220 and SDM-9230 are able to keep track of billing information on voice calls. For analog FXS channels, the unit 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, you must:

- Configure the **LINK** to generate billing signals with the appropriate *Billing signal type* for your network (For details, refer to the Analog Voice fascicle of this document series).

  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 one of the following values:
  - **For NetPerformer versions later than V10.1.0 R03**: *EGRESS*, *INGRESS* or *BOTH ENDS*
  - **For NetPerformer V10.1.0 R03 or earlier:** *TERMINATE*, *ORIGINATE* or *BOTH ENDS*

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.

6.1.1 LINK Configuration

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

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

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,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 6-1):
  - **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 6-1: Characteristics of the Billing Signal Type Parameter

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

6.1.2 CHANNEL Configuration

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

SE/SLOT/#/CHANNEL example: on FXS interface card for Billing Signals

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,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> 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 (sec) (0-600,def:1) ?
VOICE 201> Billing signal duration (ms) (20-1000,inc:20,def:20) ?
VOICE 201> Billing signal intervals (sec) (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* NetPerformer V10.1.0 R03 or earlier: *DISABLE, TERMINATE, ORIGINATE, BOTH ENDS*	*DISABLE*	Determines the conditions for generating billing signals on this FXS channel: **EGRESS (or TERMINATE):** Billing signals are generated if this FXS channel **places a call** to the phone that is plugged into it. **INGRESS (or ORIGINATE):** 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.
First billing signal time (sec) *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.

Table 6-2: Parameters for Billing Signals on an FXS Channel

Parameter	Range of Values	Default	Function
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 (sec) *ifvceAnalog-BillingTone-Intervals*	0 to 600	1	Sets the **wait time**, in seconds, between the billing signals that are generated on this channel.

Table 6-2: Parameters for Billing Signals on an FXS Channel

6.2 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, execute **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.**

SE/SLOT/#/CHANNEL example: on FXS interface card for Caller ID retransmission

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,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> 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
```

6.3 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, execute **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.**

SE/SLOT/#/CHANNEL example: on FXO interface card for Caller ID detection

```
BOSTON>SE
SETUP
Item (BRIDGE/CALLER ID/CLASS/CUSTOM/FILTER/GLOBAL/HUNT/IP/IPX/MAP/PHONE/
PORT/PU/PPPOE/PPPUSER/PVC/REDUNDANCY/SCHEDULE/SLOT/USER/VLAN,
def:BRIDGE) ? SLOT
SLOT> Slot number (1/2/3,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 (sec) (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> 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
```

7

Custom Signaling

Custom signaling parameters are available on NetPerformer to fine-tune the line signaling characteristics of analog and digital voice connections. When these characteristics are carefully adjusted, NetPerformer voice channels can support non-standard equipment.

7.1 Line Signaling Characteristics

Line signaling is the call state signaling used to establish and clear a call. The way this signaling is accomplished varies according to the interface type: digital or analog (E&M).

7.1.1 Digital Line Signaling

Digital E1 interfaces can use Channel Associated Signaling (CAS) for line signaling. CAS signaling uses 4 bits, referred to as ABCD. The signaling bits are in timeslot 16. The other timeslots take turns using timeslot 16 to transmit and receive call state indications.

Digital T1 interfaces can use ESF or D4 framing. Four bits (ABCD) are used with ESF framing. Two bits (AB) are used for signaling when the framing is set to D4. However, some PBXs require the CD bits to be 00 or 11, or to mirror the AB bits (A=C, B=D). PBX-dependent variations can be configured using the Custom Signaling submenu of the Setup (**SE**) command.

The NetPerformer supports the following digital line signaling types:

- R2
- R2-China, which accommodates special E1 signaling requirements in China
- E&M Immediate Start
- E&M Wink Start
- E&M PLAR (Private Line Automatic Ringdown)
- E&M Delay Dial (this type is not specifically configured. A delay dial channel parameter is used to delay the output of the dial digits when the NetPerformer is the calling side.)

You can also use the following analog signaling types on digital ports:

- FXO
- FXS
- GND FXO
- GND FXS

7.1.2 Analog Line Signaling

Analog line signaling is accomplished through a change on a physical connection.

- For FXS and FXO, a ringing voltage is applied to the Ring lead to indicate a call, and the loop between the Tip and Ring leads is closed to indicate that the call is accepted (Loop Start).
- For E&M, the call states are indicated by controlling the voltage level on the E and M leads. In general, the PBX raises the M lead to indicate an incoming call, and the NetPerformer raises the E lead to indicate a call accept.

  The NetPerformer supports the following analog line signaling types:

  - FXS Loop Start
  - FXO Loop Start
  - E&M Immediate Start
  - E&M Wink Start

7.2 Adjusting the Line Signaling Characteristics

Any Custom Signaling characteristics required for the application can be configured from the NetPerformer console. The unit sends these parameter values to the Signaling Engine for control of custom signaling and custom ring.

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

7.2.1 Configuring Custom Signaling

Caution

The procedures described in this section are intended for experienced users and technical personnel only. Inappropriate changes to line signaling characteristics can introduce unwanted delays and prevent call completion or proper call termination. If you are not fully familiar with line signaling characteristics and the requirements of your voice equipment, do not make any changes to the NetPerformer standard configuration using the CUSTOM menu.

Custom signaling characteristics are set using the NetPerformer console. To access the CUSTOM menu:

1. Enter **SE** at the console command prompt to access the SETUP menu.
2. Enter **CUSTOM** at the Item prompt
3. Change the Custom Signaling parameters from their default values, if desired.

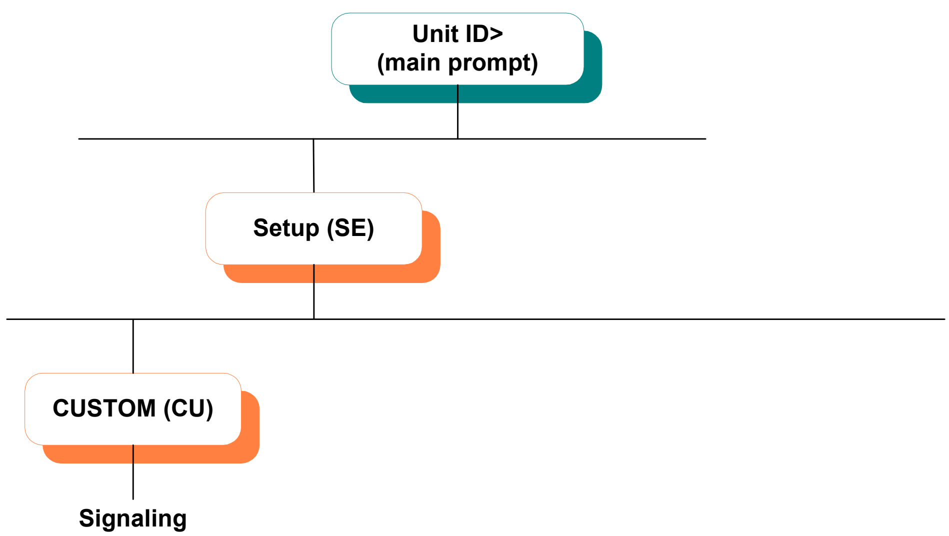


Figure 7-1: SETUP/CUTSOM Path in the CLI Tree

7.2.2 Customizing the Line Signaling Characteristics

1. Enter **SIGNALING** at the second Item prompt.
2. Load a standard signaling template, if desired (default **NO**). Each template defines a standard line signaling protocol, including all parameter values associated with that protocol. When you start with a standard signaling template, you need to change only those parameters that are unique to your application.

NOTE: If you load a standard signaling template (parameter set to **YES**), it will overwrite the current Custom Signaling configuration. You must confirm this action. If you do not load a standard signaling template (parameter set to **NO**), the software will take the last- configured Custom Signaling values as the starting definition. The default signaling definition used is **IMMEDIATE START**.

3. Select the standard signaling template that you would like to load: **IMMEDIATE START, R2, PLAR, FXO, FXS, GND FXO, GND FXS, R2-CHINA** or **WINK START**, default **IMMEDIATE START**.

Caution

The full roster of custom signaling templates has not yet been tested in the field, but preliminary lab tests with the IMMEDIATE START template reveal no problematic areas.

4. Select the line signaling characteristic that you want to configure with a non-standard value: **IDLE, SEIZURE, PULSE DIAL, WINK, CLEAR BACK, CLEAR FORWARD, ANSWER, RING** or **ALL**, default **ALL**.
5. You are then presented with a list of parameters that can be adjusted for the line signaling characteristic you have chosen. These parameters are listed in the following table. The way you define these parameters will affect the way the voice channel is brought up, maintained and terminated when it is configured with the CUSTOM signaling type.

Caution

If you are unsure of the exact values required to support your voice equipment, **DO NOT CHANGE** the Custom Signaling parameters or put the Signaling Type parameter to **CUSTOM**.

NOTE: If the voice channel is configured with the Signaling Type parameter set to any other value than CUSTOM, the Custom Signaling parameters you define will have no effect.

7.2.3 Custom Line Signaling Parameters

The following table shows the parameters that are available at the console for fine-tuning the line signaling characteristics. All console entries must be made in hexadecimal notation. For all parameters the range of values is **00000000 - FFFFFFFF**.

NOTE: These parameters can be accessed from the console or through dial-up or Telnet connection only. There are no SNMP equivalents.

The default value of each parameter is also shown in the following table. This is the value provided by the standard **IMMEDIATE START** template. The actual default value you see at the console will be this value or the value you last entered for this parameter.

Line Signaling Characteristic	Parameter Name	Default Value
IDLE	Idle detect minimum time	*001715D8*
	Idle detect maximum time	*00494504*
	Idle detect start pattern	*001715B8*
	Idle detect start pattern mask	*1000450C*
	Idle detect finish pattern	*FEE00CFC*
	Idle detect finish pattern mask	*004944EC*
	Idle generate start time	*FEE00CFC*
	Idle generate finish time	*004944F4*
	Idle generate start pattern	*FEE00CFC*
	Idle generate start pattern mask	*00494504*
	Idle generate finish pattern	*000A9030*
	Idle generate finish pattern mask	*00494524*

Table 7-1: Custom Signaling Parameters

Line Signaling Characteristic	Parameter Name	Default Value
SEIZURE	Seizure detect minimum time	*00000000*
	Seizure detect maximum time	*00000004*
	Seizure detect start pattern	*00355104*
	Seizure detect start pattern mask	*0049451C*
	Seizure detect finish pattern	*000AB258*
	Seizure detect finish pattern mask	*00000008*
	Seizure generate start time	*002AD1C0*
	Seizure generate finish time	*00000000*
	Seizure generate start pattern	*00000000*
	Seizure generate start pattern mask	*0049455C*
	Seizure generate finish pattern	*000A3EEC*
	Seizure generate finish pattern mask	*00000C13*
PULSE DIAL	Pulse dial detect make minimum time	*00651094*
	Pulse dial detect make maximum time	*002AD1C0*
	Pulse dial detect break minimum time	*00494568*
	Pulse dial detect break maximum time	*00000800*
	Pulse dial detect start pattern	*FEE00CFC*
	Pulse dial detect start pattern mask	*0049454C*
	Pulse dial detect finish pattern	*00000001*
	Pulse dial detect finish pattern mask	*000003E4*
	Pulse dial generate start time	*FEE00CFC*
	Pulse dial generate finish time	*00494564*
	Pulse dial generate inter digit time	*000A9030*
	Pulse dial generate start pattern	*00000001*
	Pulse dial generate start pattern mask	*00000000*
	Pulse dial generate finish pattern	*00000004*
	Pulse dial generate finish pattern mask	*00355104*

Table 7-1: Custom Signaling Parameters

Line Signaling Characteristic	Parameter Name	Default Value
WINK	Wink detect minimum time	*0049457C*
	Wink detect maximum time	*000AB258*
	Wink detect start pattern	*0049457C*
	Wink detect start pattern mask	*004945D4*
	Wink detect finish pattern	*0000000D*
	Wink detect finish pattern mask	*002AD1B0*
	Wink generate start time	*00494594*
	Wink generate finish time	*0003E604*
	Wink generate start pattern	*00000001*
	Wink generate start pattern mask	*004945D4*
	Wink generate finish pattern	*00000001*
	Wink generate finish pattern mask	*002AD1B0*
CLEAR BACK	Clear back detect minimum time	*004945AC*
	Clear back detect maximum time	*0003D680*
	Clear back detect start pattern	*00000000*
	Clear back detect start pattern mask	*002AD1B0*
	Clear back detect finish pattern	*00000000*
	Clear back detect finish pattern mask	*0000001D*
	Clear back generate start time	*004945CC*
	Clear back generate finish time	*00040C70*
	Clear back generate start pattern	*004945CC*
	Clear back generate start pattern mask	*00000001*
	Clear back generate finish pattern	*001F1188*
	Clear back generate finish pattern mask	*001F1138*

Table 7-1: Custom Signaling Parameters

Line Signaling Characteristic	Parameter Name	Default Value
CLEAR FORWARD	Clear forward detect minimum time	*00000000*
	Clear forward detect maximum time	*0000001D*
	Clear forward detect start pattern	*00494644*
	Clear forward detect start pattern mask	*00042900*
	Clear forward detect finish pattern	*00004644*
	Clear forward detect finish pattern mask	*0004132C*
	Clear forward generate start time	*00494604*
	Clear forward generate finish time	*00000000*
	Clear forward generate start pattern	*00000000*
	Clear forward generate start pattern mask	*494D4D45*
	Clear forward generate finish pattern	*44494154*
	Clear forward generate finish pattern mask	*45205354*
ANSWER	Answer detect minimum time	*41525400*
	Answer detect maximum time	*0000000F*
	Answer detect start pattern	*001F1180*
	Answer detect start pattern mask	*00000000*
	Answer detect finish pattern	*00000000*
	Answer detect finish pattern mask	*00000000*
	Answer generate start time	*00000000*
	Answer generate finish time	*00000000*
	Answer generate start pattern	*00000000*
	Answer generate start pattern mask	*00000000*
	Answer generate finish pattern	*00000000*
	Answer generate finish pattern mask	*00000000*
RING	Ring generate start pattern	*00000000*
	Ring generate start pattern mask	*00000000*
	Ring generate finish pattern	*00000000*
	Ring generate finish pattern mask	*00000001*

Table 7-1: Custom Signaling Parameters

7.3 Customizing the Ring Cadence Characteristics

Ring cadence characteristics are configured separately from the line signaling characteristics. These parameters determine the type of cadence that the NetPerformer will use when generating a ring.

► To access the RING CADENCE submenu:

1. Enter **SE** at the console command prompt to access the SETUP menu.
2. Enter **CUSTOM** at the Item prompt.
3. Enter **RING CADENCE** at the second Item prompt.
4. Specify initial cadence (**YES**, **NO**, default **NO**). This parameter determines whether an initial cadence is required (**YES,** NO, default **NO**).
   - The default, **NO**, means all rings are identical. Only a loop cadence needs to be defined in this case (with the loop cadence parameters, below).
   - For example, the ring used in the United Kingdom has a slightly different cadence on the first ring (the initial cadence) than the following rings (the loop cadence).
5. If an initial cadence is required (Specify Initial Cadence is set to **YES**), enter the appropriate values for the following:
   - Initial ring on period (1-10000 msec, default 2000 msec). This is the duration that the phone will ring on the first ring.
   - Initial ring off period (1-10000 msec, default 4000 msec). This is the interval between the end of the first ring and the start of the second ring.
   - Add more initial cadence parameters (**YES, NO**, default **NO**). This determines whether more initial cadence parameters will be displayed for configuration. If set to **YES**, the Initial Ring On Period and Initial Ring Off Period parameters are listed again.
6. To define the loop cadence, enter the appropriate values for the following:
   - Loop ring on period (1-10000 msec, default 2000 msec). This is the duration that the phone will ring on the second and subsequent rings (or on all rings if an initial cadence is not defined). For example, the North American ring cadence has a Loop Ring On Period of 2000 ms, with no initial cadence.
   - Loop ring off period (1-10000 msec, default 4000 msec). This is the interval between the end of the second ring and the start of the third ring, and between all subsequent rings (or between all rings if an initial cadence is not defined). For example, the North American ring cadence has a Loop Ring Off Period of 4000 ms, with no initial cadence.

Add more cadence parameters (**YES, NO**, default **NO**). This determines whether more loop cadence parameters will be displayed for configuration. If set to **YES**, the Loop Ring On Period and Loop Ring Off Period parameters are listed again.

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