NORTEL NETWORKS

Passport 7400

# Voice Networking

Guide

241-7401-755

Passport 7400

# Voice Networking

Guide

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# About this document

This guide describes the Passport Voice Networking service, including

- operation and capabilities
- installation
- application examples
- how to interpret and resolve problems

## Who should read this document and why

This guide is for persons who perform the following tasks for Voice Networking:

- planning
- engineering
- installing and configuring
- configuration
- operating and maintaining
- troubleshooting

## What you need to know

This guide assumes that you understand the Passport network architecture. You can learn more about the product by reading 241-5701-030 *Passport 7400, 15000, 20000 Overview*.

## How this guide is organized

The 241-7401-755 *Passport 7400 Voice Networking Guide* contains the following sections:

- "Voice Networking configuration" (page 23) provides an overview of the tasks for configuring the Voice Networking service.
- "Basic Voice Networking components configuration" (page 27) contains procedures for configuring the basic Voice Networking components.
- "Signaling channel and protocols configuration" (page 37) contains procedures for configuring the SignallingChannel component and the signaling protocols.
- "Voice Networking call server configuration" (page 59) contains procedures for configuring the Voice Networking call server (VNCS).
- "Voice route and subroute configuration" (page 69) contains procedures for configuring the voice route and voice subroute.
- "Other Voice Networking options configuration" (page 83) contains procedures for configuring digit manipulation, call hunting through a hunt group server, and Voice Networking accounting.
- "Voice Networking fundamentals" (page 93) describes the Voice Networking service's capabilities and system parameters.
- "Signaling protocols fundamentals" (page 113) provides details on the channel associated signaling (CAS) and common channel signaling (CCS) protocols supported by Voice Networking.
- "Routing fundamentals" (page 133) describes how Voice Networking routes audio calls through the Passport subnet.
- "Compliance with standards—Voice Networking signaling protocols" (page 181) describes the ITU-T standards to which Voice Networking complies.
- "Network migration considerations" (page 187) describes how to introduce Voice Networking to existing, PVC-based networks.
- "Glare processing" (page 217) describes how Voice Networking handles call collisions.

- “Call release cause codes” (page 219) contains information about the call release cause values that are supported by Voice Networking.

## What’s new in this document

The following feature was added to this document:

- “EuroISDN on 4pMVPe” (page 20)

Other changes to this document include the following:

- Updated this document to remove references to these MVP FPs which are support discontinued (SDed):
  - 1-port DS1 MVP with cardtype 1pDS1MVP and PEC NTFN62
  - 1-port DS1Voice with cardtype 1pDS1V and PEC NTFP41
  - 1-port E1 MVP with cardtype 1pE1MVP and PEC NTFN20
  - 1-port E1Voice with cardtype 1pE1V and PEC NTFP43
  - 1-port J2MV with cardtype J2MV and PEC NTBP96
  - 1-port TTC2M MVP with cardtype 1pTTC2mMVP and PEC NTFN64

This document was restructured into a modular, task-based format to improve the usability of the information. The following changes were made to this document:

- Procedures were grouped into higher-level tasks.
- Task flow charts were added to improve navigation through tasks and procedures, to set tasks and procedures in context, and to provide a visual representation of prerequisites and configuration paths.
- Procedures were restructured into a modular format.
- Purpose statements were added to tasks and procedures to provide context.
- Prerequisites were divided into those applicable to an entire task. those applicable only to a specific procedure, and those applicable only to a specific procedure step. Prerequisites applicable to an entire task were placed in the appropriate task-level prerequisite section, prerequisites

applicable only to a specific procedure were placed in the prerequisite section of the procedure, and prerequisites applicable only to a specific step were placed in the step.

- 'Where' statements were removed from procedures and the content placed in the 'Variable definitions' table following the procedure.
- A 'Procedure Job Aid' section was added to procedures where appropriate. This consists of information that supports the procedure, such as a component hierarchy figure, a checklist, or a diagram.
- Conceptual and reference information were removed from procedures, placed in the appropriate conceptual or reference section, and cross-referenced from the procedure where appropriate.
- The chapters "Basic Voice Networking components configuration" (page 27), "Voice Networking call server configuration" (page 59), "Voice route and subroute configuration" (page 69), "Voice Networking fundamentals" (page 93), "Signaling protocols fundamentals" (page 113), and "Routing fundamentals" (page 133) were updated to remove references to obsolete MVP FPs.

### EuroISDN on 4pMVPe

The following sections were added or updated for this feature:

- "Configuring the Euro-ISDN signaling protocol" (page 55)
- "Card types and supported Voice Networking protocols" (page 109)

## Conventions

There are a number of documentation conventions you should know about.

- nonproportional spaced plain type
  Nonproportional spaced plain type represents system generated text or text that appears on your screen.
- **nonproportional spaced bold type**
  Nonproportional spaced bold type represents words that you should type or that you should select on the screen.

- *italics*
  Statements that appear in italics in a procedure explain the results of a particular step and appear immediately following the step.

  Words that appear in italics in text are for naming.
- [optional_parameter]
  Words in square brackets represent optional parameters. The command can be entered with or without the words in the square brackets.
- <general_term>
  Words in angle brackets represent variables which are to be replaced with specific values.
- UPPERCASE,lowercase
  Passport commands are not case-sensitive and do not have to match commands and parameters exactly as shown in this document, with the exception of string option values (for example, file and directory names) and string attribute values.
- |
  This symbol separates items from which you may select one; for example, ON|OFF indicates that you may specify ON or OFF. If you do not make a choice, a default ON is assumed.
- ...
  Three dots in a command indicate that the parameter may be repeated more than once in succession.

  The term absolute pathname refers to the full specification of a path starting from the root directory. Absolute pathnames always begin with the slash ( / ) symbol. A relative pathname takes the current directory as its starting point, and starts with any alphanumeric character (other than /).

## Related documents

For the complete list of documents contained in the Passport documentation library, see 241-5701-001 *Passport 7400, 15000, 20000 Documentation Guide*.

Refer to the following documents for information on configuring and operating Voice Networking:

- 241-5701-500 *Passport 6400, 7400, 15000, 20000 Alarms*
- 241-6001-022 *Preside MDM Network Reporting System User Guide*
- 241-5701-050 *Passport 7400, 15000, 20000 Commands*
- 241-5701-060 *Passport 7400, 15000, 20000 Components*
- 241-7401-110 *Passport 7400, DPN-100 Interworking Guide*
- 241-7401-200 *Passport 7400 Hardware Description*
- 241-7401-240 *Passport 7400 Hardware Installation, Maintenance and Upgrade*
- 241-5701-400 *Passport 7400, 15000, 20000 Networking Overview*
- 241-5701-405 *Passport 7400, 15000, 20000 Call Server Guide*
- 241-5701-410 *Passport 7400, 15000, 20000 Call Redirection Server Guide*
- 241-5701-415 *Passport 7400, 15000, 20000 Hunt Group Server Guide*
- 241-5701-420 *Passport 7400, 15000, 20000 Trunking Guide*
- 241-5701-425 *Passport 7400, 15000, 20000 Dynamic Packet Routing System Guide*
- 241-5701-435 *Passport 7400, 15000, 20000 Path-Oriented Routing System Guide*
- 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide*
- 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference*
- 241-5701-650 *Passport 7400, 15000, 20000 Accounting Fundamentals*
- 800-1942-31, *ISDN BRI Voice Module Manual*
- NN10600-605 *Passport - MDM Network Security: Operations*

# Chapter 1
# Voice Networking configuration

Configure Voice Networking software on both the control processor (CP) and function processor (FP) on Passport nodes that directly connect to PBXs to

- configure the Software component with the Voice Networking feature corresponding to the software package you downloaded and installed on the Passport node
- specify the signaling protocol software to use and any function processor-specific feature software (such as a particular encoding choice)


- "Prerequisites to Voice Networking configuration" (page 23)
- "Voice Networking configuration task" (page 24)


## Prerequisites to Voice Networking configuration

- Use the procedures in 241-5701-270 *Passport 7400, 15000, 20000 Software Installation Guide* to install Voice Networking and related software.
- Install and configure path-oriented routing system (PORS) and the dynamic packet routing system (DPRS). PORS and DPRS enable the Voice Networking service connections between Passport nodes and route the audio traffic. Ensure that your version of PORS and DPRS are compatible with the version of Voice Networking software you plan to deploy.

  For information on configuring PORS, see 241-5701-435 *Passport 7400, 15000, 20000 Path-Oriented Routing System Guide* and 241-5701-420 *Passport 7400, 15000, 20000 Trunking Guide*. For information on

configuring DPRS and the call server, see 241-5701-425 *Passport 7400, 15000, 20000 Dynamic Packet Routing System Guide* and 241-7401-110 *Passport 7400, DPN-100 Interworking Guide*, and 241-5701-405 *Passport 7400, 15000, 20000 Call Server Guide*, respectively.

- If you plan to use PORS with DNA-based routing, you must configure the Routing DpnAddressPlan components and the ModuleData VirtualCircuitSystem components. See "Voice Networking based on DNAs and DPRS" (page 151).

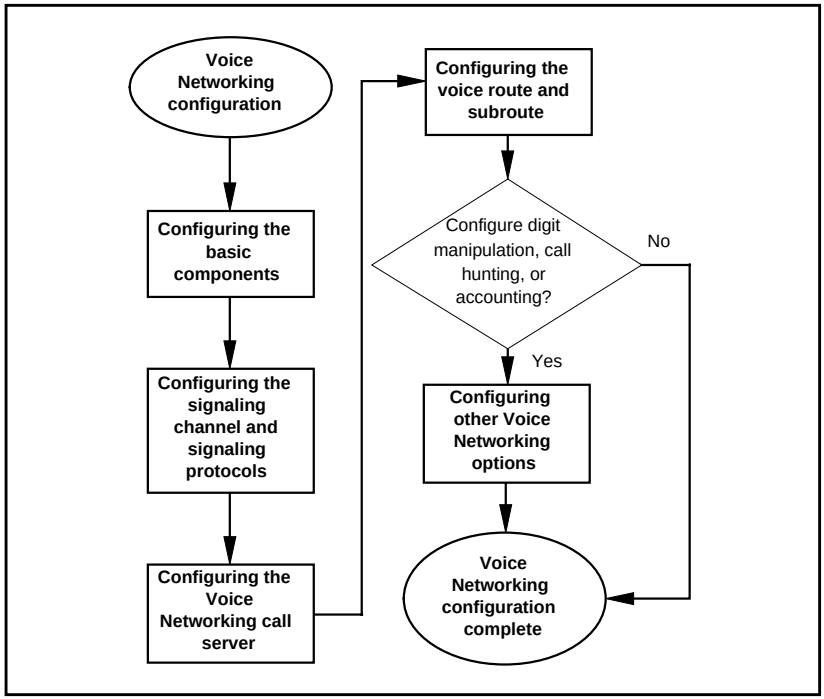
  For more information about configuring the Routing DpnAddressPlan components, see 241-7401-110 *Passport 7400, DPN-100 Interworking Guide*. For more information about configuring the ModuleData VirtualCircuitSystem components, see 241-5701-902 *Passport 7400, 15000, 20000 Configuring Frame Relay*.

- Install network clock synchronization (NCS). NCS is described in 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide*.
- Install and configure hunt group software on each node that hosts a hunt group server. For information on installing and configuring hunt group software, see 241-5701-415 *Passport 7400, 15000, 20000 Hunt Group Server Guide*.
- Install the required hardware. For more information, see "What hardware does Voice Networking require?" (page 97).
- To simplify the configuration process, relate like-numbered component and subcomponent instances.

## Voice Networking configuration task

"Voice Networking configuration task flow" (page 25) shows you the sequence of tasks and procedures you perform to configure Voice Networking. To link to any task or procedure, go to "Task navigation" (page 25).

**Figure 1**
**Voice Networking configuration task flow**




## Task navigation

- “Basic Voice Networking components configuration” (page 27)
- “Signaling channel and protocols configuration” (page 37)
- “Voice Networking call server configuration” (page 59)
- “Voice route and subroute configuration” (page 69)
- “Other Voice Networking options configuration” (page 83)

# Chapter 2
# Basic Voice Networking components configuration

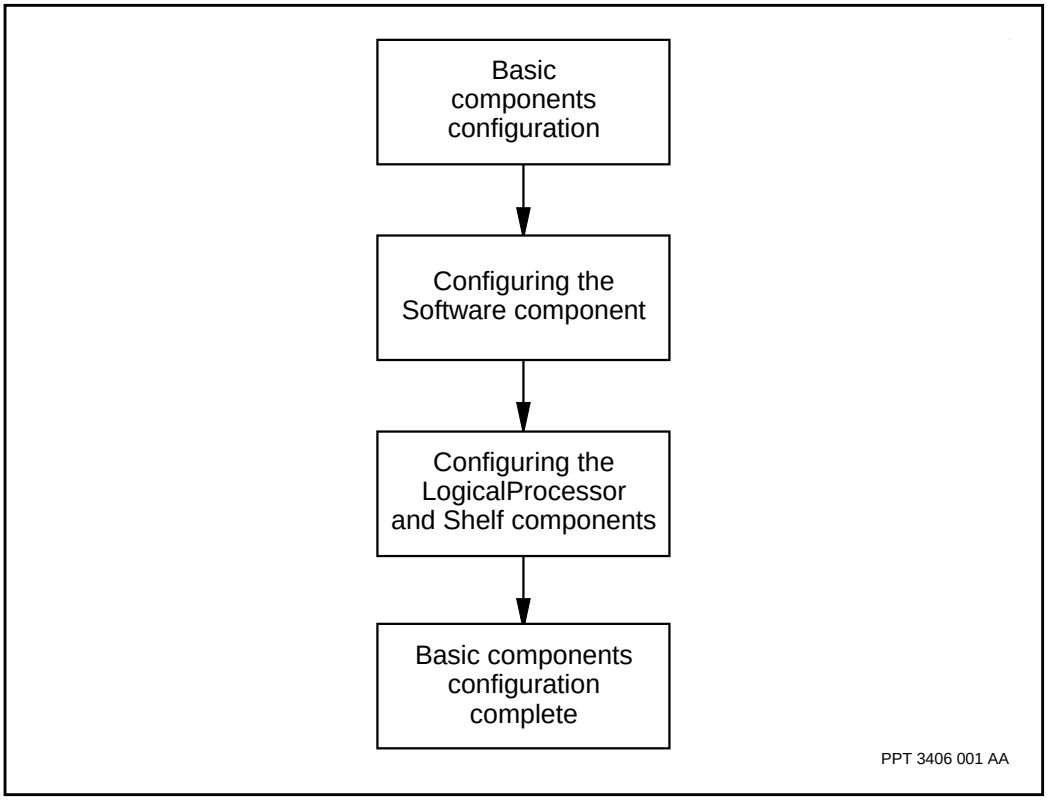
Configure the basic components of the Voice Networking service.


- "Basic Voice Networking components configuration task" (page 27)


## Basic Voice Networking components configuration task

"Basic Voice Networking components configuration task flow" (page 28) shows you the sequence of tasks and procedures you perform to configure the basic Voice Networking components. To link to any of the tasks or procedures, go to "Task navigation" (page 28).

**Figure 2**
**Basic Voice Networking components configuration task flow**




## Task navigation

- “Configuring the Software component” (page 29)
- “Configuring the LogicalProcessor and Shelf components” (page 31)

# Configuring the Software component

Configure Voice Networking software on each Passport node in the network that connects to a PBX.

## Prerequisites

- If you use DNA-based call routing, ensure that the callServer feature is also present under the featureList attribute for Lpt/CP.

## Procedure steps

1. Add voice networking call server software to the LogicalProcessorType component instance for the control processor (CP).

   ```
   set sw lpt/CP featureList vncsCallServer
   ```

2. Add an Lpt component instance for the Voice Networking service.

   ```
   add sw lpt/<name>
   ```

3. Set the featureList attribute by specifying the Voice Networking software package, defining the signaling protocol, and if necessary, adding MVP-E function processor (FP) specific software features.

   ```
   set sw lpt/<name> featureList voiceNetworking
   <protocol> <optional_feature>
   ```

4. Repeat step 1 to step 3 to configure Voice Networking software on each Passport node in the network that connects to a PBX.

## Variable definitions

| Variable | Value |
|---|---|
| <name> | Up to 25 ASCII characters. Use a descriptive word that signifies the signaling protocol you plan to use, such as MCDN or E1CAS. |
| <protocol> | The name of the software package containing the signaling protocol you plan to use.<br><br>For channel associated signaling (CAS), the value casSig applies to all 3 supported CAS types. For common channel signaling, the values are<br><br>• etsiQsig<br>• nisSig<br>• mcdnSig<br>• euroIsdn |
| <optional_feature> | One or more of the following MVP-E FP-specific features:<br><br>• g728<br>• g729 (includes G.729A for MVP-E FPs)<br>• faxRelay<br>• tandemPassThrough |
| | |

## Procedure job aid

**Figure 3**
**Configuring the Software component hierarchy**

# Configuring the LogicalProcessor and Shelf components

Configure the LogicalProcessor and Shelf components to define and link LPs and FPs.

## Prerequisites

- To help simplify configuration, link like-numbered components. For example, link the component instance Lp/1 to the FP represented by the component instance Shelf Card/1.
- Verify the channel configuration on the connected PBX before defining channel and timeslot pairs on the Passport node.

## Procedure steps

1. Add logical processors.

   **add lp/<lp_number>**

2. Link the LPs to the configured Voice Networking software.

   **set lp/<lp_number> logicalProcessorType sw lpt/<name>**

3. For each FP, define its slot number.

   **add shelf card/<card_number>**

4. For each FP, define its card type.

   **set shelf card/<card_number> cardType <FP>**

5. Link each LP to an FP.

   **set lp/<lp_number> mainCard shelf card/<card_number>**

6. Perform a semantic check of your changes.

   **check Prov**

7. Add a port.

   **add lp/<lp_number> <port>/<port_number>**

8. Specify the type of framing format to use for the port.

   **set lp/<lp_number> <port>/<port_number> linetype <framing_format>**

9. Specify the clocking source to use for each port.

   **set lp/<lp_number> <port>/<port_number> clockingSource <value>**

**10** Add channels and timeslots to each port. The type of signaling protocol and port determines how to provision channels and timeslots. For both CAS and CCS protocols interfacing to an E1 port, define 30 channels to process calls and one to carry signaling information. For CCS protocols interfacing to a DS1 port, define 23 channels for bearer services and one to carry signaling information. For CAS protocols interfacing to a DS1 port, define 24 channels to carry traffic.

```
add lp/<lp_number> <port>/<port_number> chan/
<signaling_channel>
```

```
set lp/<lp_number> <port>/<port_number> chan/
<signaling_channel> timeslots <timeslot_number>
```

**11** Optionally, for an E1 port using either a CAS or CCS protocol, delete channel 0 before you add channel and timeslot pairs to allow you to align channel and timeslot numbers with those on the connected PBX, simplifying the configuration and monitoring processes.

```
del lp/<lp_number> e1/0 chan/0
```

**12** Optionally, for a DS1 port using a CCS protocol (NIS A211-1, ETSI QSIG, or MCDN), align channel and timeslot numbers by assigning timeslot 24 to channel 0 (timeslot 24 carries signaling information) to simplify the configuration and monitoring processes.

```
set lp/<lp_number> ds1/0 chan/0 timeslots 24
```

**13** Optionally, for a DS1 port using a CAS protocol (DS1 CAS), align channel and timeslot numbers by assigning timeslot 25 to channel 0 (timeslot 25 is a virtual timeslot that carries DS1 CAS signaling information) to simplify the configuration and monitoring processes.

```
set lp/<lp_number> ds1/0 chan/0 timeslots 25
```

**14** Repeat step 1 to step 13 for each LogicalProcessor component instance you require.

**15** Repeat step 1 to step 14 to configure logical and function processors on each Passport node in the network that connects to a PBX.

## Variable definitions

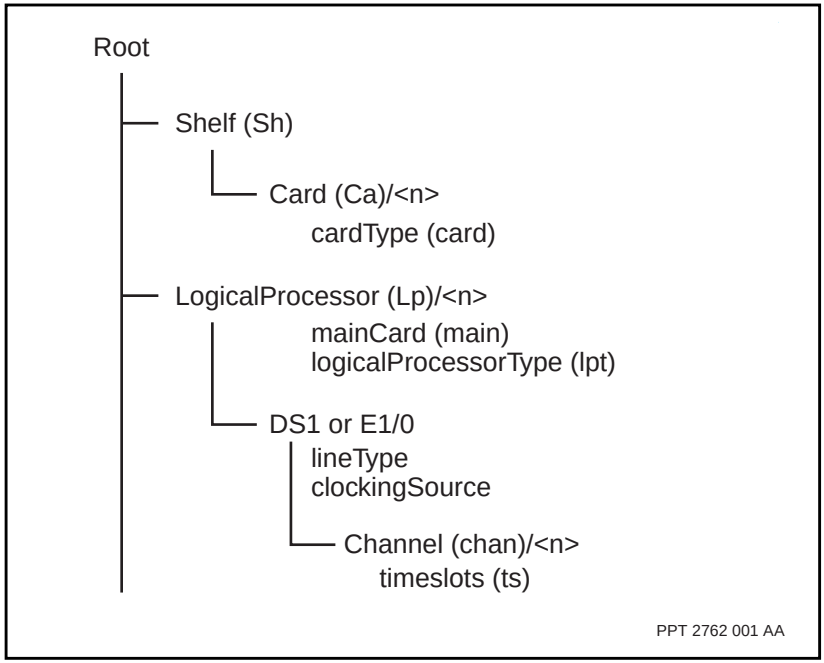
| Variable | Value |
|---|---|
| <card number> | The instance number of the card. |
| <FP> | The card type. To determine the value for the FP you are configuring, refer to 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference*. |
| | Ensure that the FP you specify supports the protocol and features you configured under the Software LogicalProcessorType component. |
| <framing_format> | The signaling protocol you specify determines what type of framing format to use. |
| | For CCS protocols, use ccs for an E1 component and d4 or esf for a Ds1 component, depending on the framing format being used by the connected PBX. |
| | For CAS protocols, use cas for an E1 component and d4Cas or esfCas for a DS1 component, depending on the framing format being used by the connected PBX. |
| <lp_number> | The instance number of the LP. |
| <name> | The title given to the Software LogicalProcessorType component instance. |
| <port> | Either e1 or ds1. |
| | The FP you specify under the Card component determines the type of port you provision. To determine the value for the FP you are configuring, refer to 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference*. |
| | When you add an E1 or Ds1 component, the system automatically creates the component instance Channel/0. |
| <port_number> | The instance number of the port. |

(Sheet 1 of 2)

| Variable | Value |
|---|---|
| <signaling_channel> | The channel you assign to carry signaling information.<br><br>Timeslot 16 on an E1 link carries signaling information for both CAS and CCS protocols. Typically, you assign timeslot 16 to channel 16, as most PBXs define channels 1 to 15 and 17 to 31 to carry traffic. However, certain PBXs running CCS protocols (for example, ETSI QSIG) define channels 1 to 30 for bearer services and assign timeslot 16 to channel 31. |
| <timeslot_number> | The timeslot number. |
| <value> | Local, line or module.<br><br>If you configured the NetworkSynchronization component, the clockingSource attribute defaults to module, otherwise it defaults to line. A value of module specifies that the E1 or DS1 port synchronizes to the Stratum-3 clock on the active CP. |
| (Sheet 2 of 2) | |

## Procedure job aid

**Figure 4**
**Configuring LogicalProcessor and Shelf component hierarchy**



# Chapter 3
# Signaling channel and protocols configuration

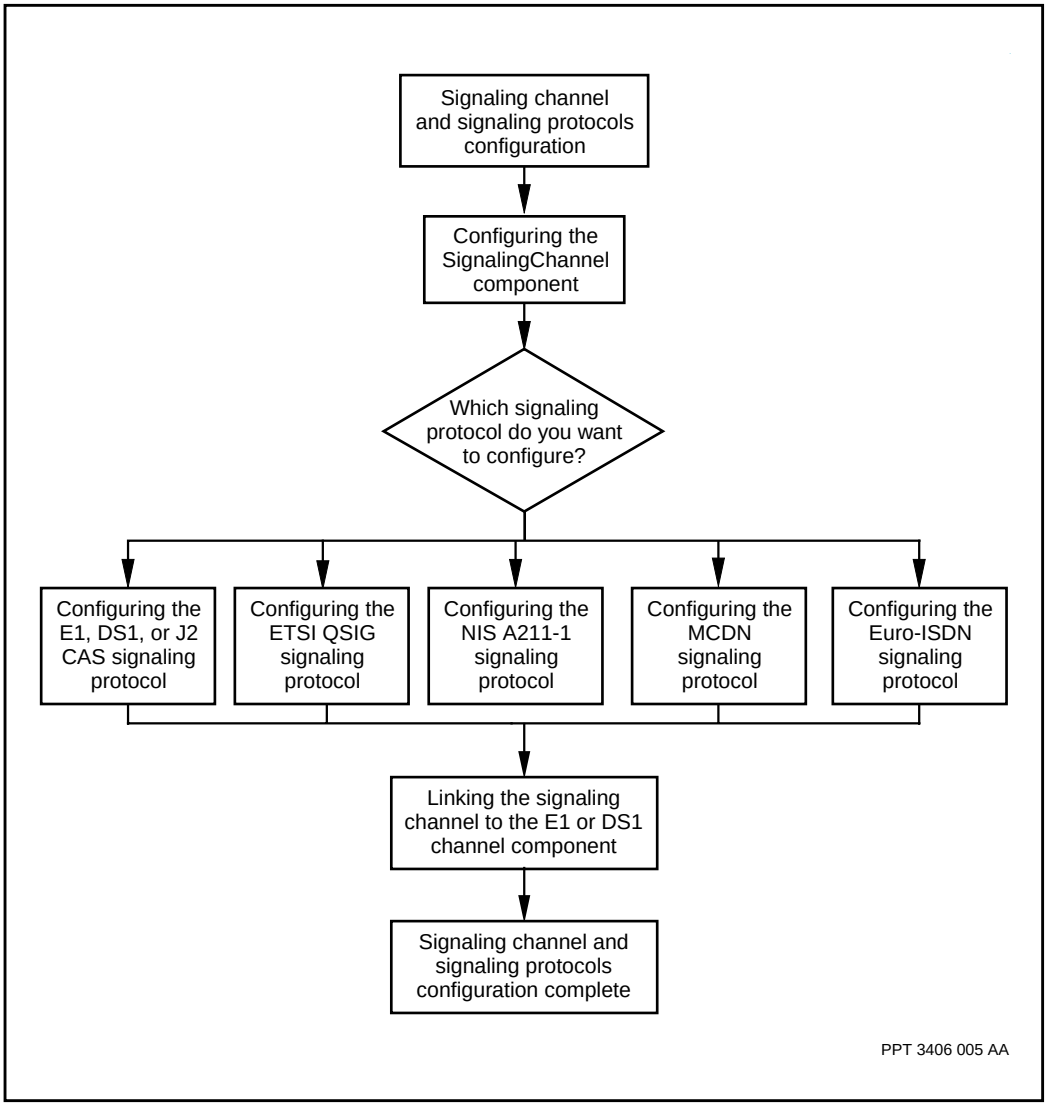
Configure the signaling channels and their associated protocols.


- "Signaling channel and protocols configuration task" (page 37)


## Signaling channel and protocols configuration task

"Signaling channels and protocols configuration task flow" (page 38) shows you the sequence of tasks and procedures you perform to configure the signaling channel and protocols. To link to any task or procedure, go to "Task navigation" (page 38).

**Figure 5**
**Signaling channels and protocols configuration task flow**




## Task navigation

- “Configuring the SignalingChannel component” (page 40)

- “Configuring the E1, DS1, or J2 CAS signaling protocol” (page 45)
- “Configuring the ETSI QSIG signaling protocol” (page 48)
- “Configuring the NIS A211-1 signaling protocol” (page 50)
- “Configuring the MCDN signaling protocol” (page 52)
- “Configuring the Euro-ISDN signaling protocol” (page 55)

# Configuring the SignalingChannel component

Configure the SignalingChannel component to define the signaling characteristics of the connection to the PBX and to define internal cause code values for certain call clearing events.

## Prerequisites

- See the table “Card types and supported Voice Networking protocols” (page 109) for the signaling protocols supported by each FP.
- See the table “Voice Networking protocol interworking” (page 110) for the matrix of supported protocol interworking.
- See the table “Voice Networking CCS-to-CAS protocol gateways” (page 110) for the matrix of supported protocol gateways. If necessary, see also “Protocol interworking and gateways” (page 125) for details about the functionality included with protocol interworking and gateways.
- See the table “Relationship between defaultNpiTon attribute and PBX NPI/TON values” (page 130).
- To automatically enable CCS-to-CAS protocol gateways, add the MCDN, etsiQsig or nisSig and casSig protocol software features on the appropriate Passport nodes (that is, add the protocol that matches the protocol running on the connected PBX) and configure the SigChan component on each node accordingly.
- To automatically enable interworking between NIS A211-1 and MCDN, add the nisSig and mcdnSig protocol software features on the appropriate Passport nodes (that is, add the protocol that matches the protocol running on the connected PBX) and configure the SigChan component on each node accordingly.

## Procedure steps

**1** Add a a signaling channel.

```
add sigchan/<signaling_channel>
```

**2** Specify the signaling protocol to use.

```
add sigchan/<signaling_channel> <protocol>
```

**3** If necessary, override or replace missing or possibly corrupted numbering plan indicator (NPI) information supplied by the calling PBX.

```
set sigchan/<signaling_channel> forceNpiTon <decision>
```

4 If necessary, override or replace missing or possibly corrupted type of number (TON) information supplied by the calling PBX.

```
set sigchan/<signaling_channel> defaultNpiTon <value>
```

5 Optionally, add the InternalCauseMap component.

```
add sigchan/<signaling_channel> InternalCauseMap
```

6 Optionally, verify that the default settings of the InternalCauseMap component's provisionable attributes meet network requirements, and make changes as required.

```
display sigchan/<signaling_channel> InternalCauseMap
```

7 Repeat step 1 to step 6 to configure the signaling channel on each Passport node that connects to a PBX.

## Variable definitions

| Variable | Value |
|---|---|
| <decision> | yes or no<br><br>No is the default.<br><br>If set to yes, Voice Networking ignores NPI and TON information sent by the calling PBX and uses the values configured under the defaultNpiTon attribute.<br><br>Even when the forceNpiTon attribute is set to no, Voice Networking can use the defaultNpiTon attribute's default value, casUnknown, when NPI and TON information in the call setup message received from the calling PBX is missing or cannot be interpreted. |
| <signaling_channel> | The channel you defined to carry timeslot 16 signaling information for CAS or CCS protocols interfacing to an E1 port or 0 for CAS or CCS protocols interfacing to a Ds1 port. |
| <port> | E1 or Ds1 |

(Sheet 1 of 2)

| Variable | Value |
|---|---|
| <protocol> | E1Cas, Ds1Cas, J2Cas, EtsiQsig, NisSig, McdnSig, or EuroIsdn<br><br>The protocol you specify corresponds to the protocol software you configured under the Software component and matches the signaling format—CAS or CCS—you specified under the E1 or DS1 port component. |
| <value> | One or more of the following: casUnknown (default), unknown, international, national, subscriber, and p0 up to p7. |
| (Sheet 2 of 2) | |

## Procedure job aid

**Figure 6**
**Configuring the SignallingChannel component hierarchy**

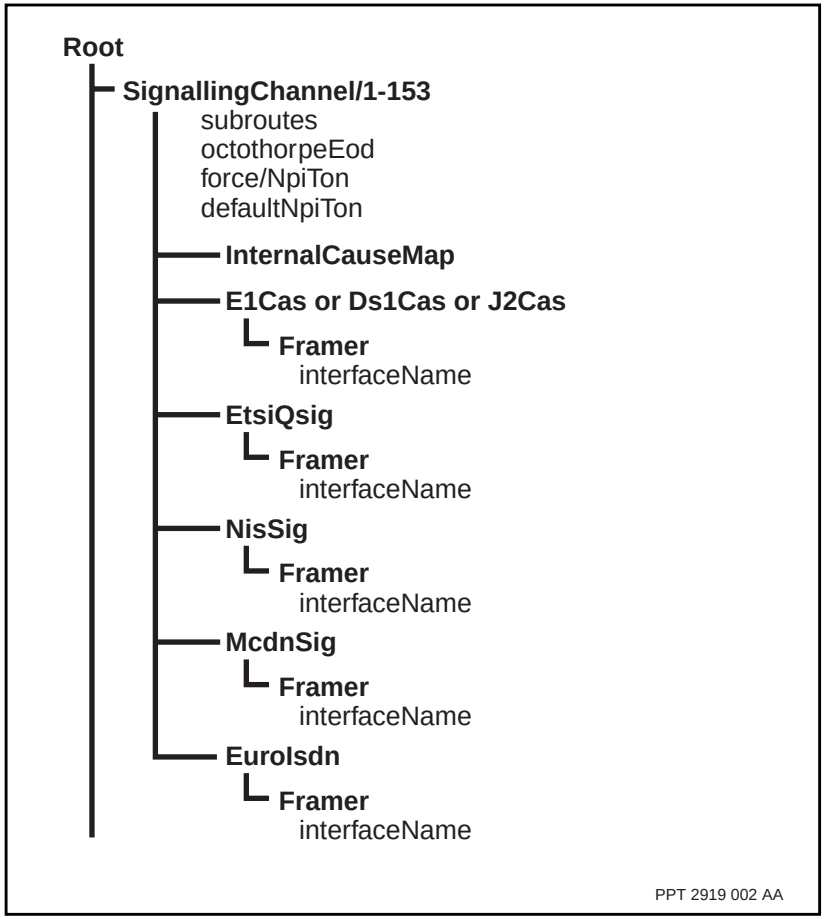
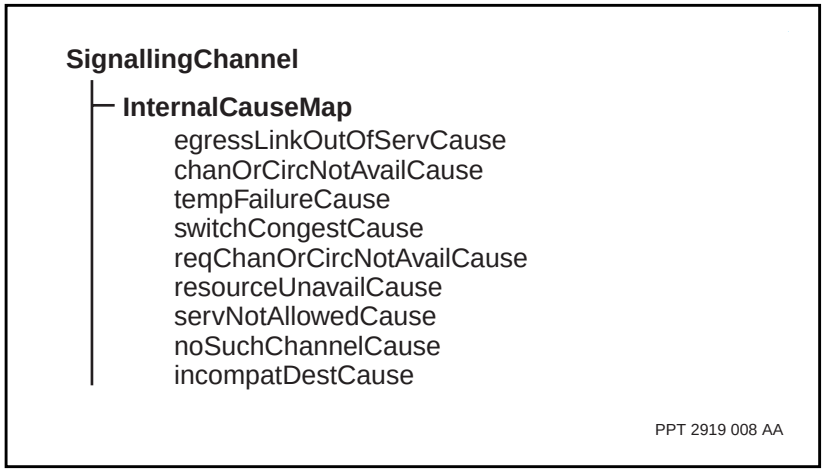


**Figure 7**
**Defining internal cause codes component hierarchy**



# Configuring the E1, DS1, or J2 CAS signaling protocol

Configure the E1, DS1, or J2 CAS signaling protocol to define the operational characteristics of the signaling channel's connection to the PBX for channel associated signaling (CAS) links:

- the tone table to use according to the country of origin
- the signaling bit combinations to use for a range of supported signaling states
- the minimum and maximum durations (measured in milliseconds) for specific signals to persist in order to be considered valid
- the burst length of DTMF digits

## Prerequisites

- Review 241-5701-060 *Passport 7400, 15000, 20000 Components* for attribute descriptions and values related to configuring the E1, DS1, or J2 CAS signaling protocol (see "Configuring the E1Cas, Ds1Cas, or J2Cas signaling protocol component hierarchy" (page 47)).

## Procedure steps

1. Select the appropriate tone table to use according to country.

   ```
   set sigchan/<signaling_channel> <signaling_protocol>
   tonetableselection <country_number>
   ```

2. Verify that the default settings of the AbcdProv group of provisionable attributes meet network requirements.

   ```
   display -p sigchan/<signaling_channel>
   <signaling_protocol> abcdprov
   ```

3. Verify that the default settings of the TimerProv group of provisionable attributes meet network requirements.

   ```
   display -p sigchan/<signaling_channel>
   <signaling_protocol> timerprov
   ```

4. Add a DTMF component.

   ```
   add sigchan/<signaling_channel> <signaling_protocol>
   DTMF
   ```

5. Set the amount of time a DTMF tone persists between dialed digits.

```
set sigchan/<signaling_channel> <signaling_protocol>
DTMF burstTime <value>
```

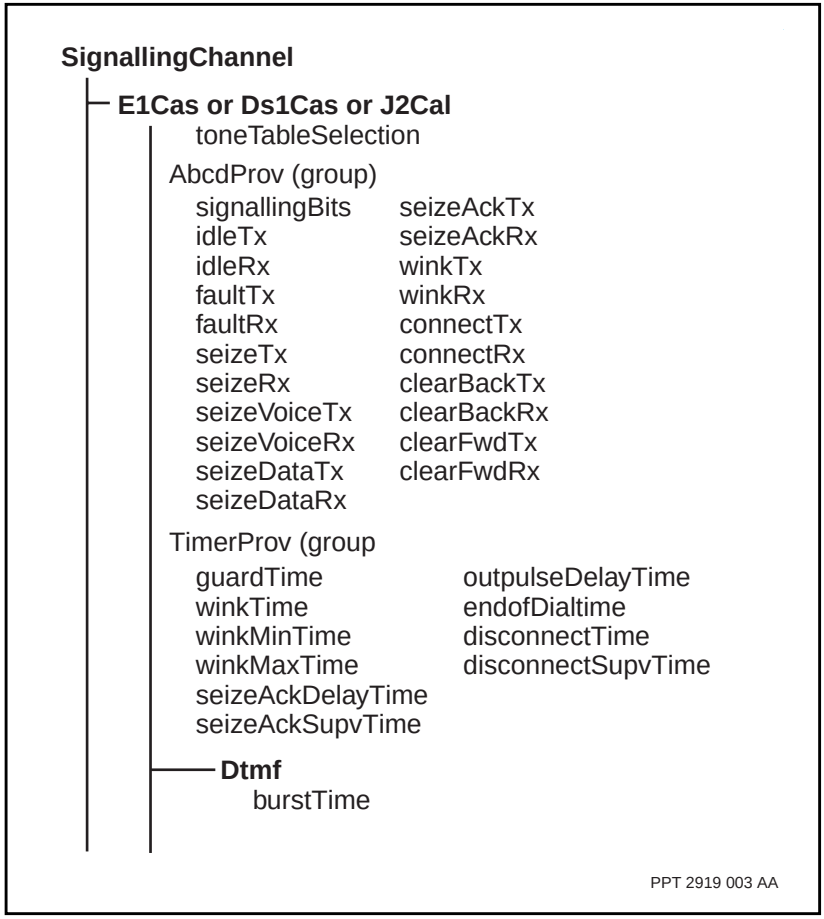
6 Repeat step 1 to step 5 for each signaling channel on each Passport node connected to a PBX that uses a CAS protocol.

## Variable definitions

| Variable | Value |
|---|---|
| <country_number> | 0 for North America, 1 for Japan, 2 for the United Kingdom, 3 for Germany, 4 for France, 5 for Central America (Brazil, Mexico), 6 for Chile, or 7 for Australia<br><br>The default value for the E1Cas component is 2 (United Kingdom). The default value for the Ds1Cas component is 0 (North America). The default value for the J2Cas component is 1 (Japan). |
| <signaling_channel> | The channel you defined to carry timeslot 16 signaling information for CAS or CCS protocols interfacing to an E1 port or 0 for CAS or CCS protocols interfacing to a Ds1 port. |
| <signaling_protocol> | E1Cas, DS1Cas, or J2Cas |
| <value> | 50, 60, 70, or 100 (default) milliseconds |
| | |

## Procedure job aid

**Figure 8**
**Configuring the E1Cas, Ds1Cas, or J2Cas signaling protocol component hierarchy**



# Configuring the ETSI QSIG signaling protocol

Configure the ETSI QSIG signaling protocol to define the functional characteristics of the signaling channel's connection to the PBX, for example:

- OSI Layer 2 and 3 timer settings
- whether to enable or disable support for overlap dialing (in both the sending and receiving directions)
- whether to enable or disable the capability to detect, segment, and reassemble large-sized ISDN messages

## Prerequisites

- Review 241-5701-060 *Passport 7400, 15000, 20000 Components* for attribute descriptions and values related to configuring the ETSI QSIG signaling protocol (see "Configuring the ETSI QSIG signaling protocol component hierarchy" (page 49)).

## Procedure steps

**1** Verify that the default settings of the EtsiQsig component's provisionable attributes meet network requirements, and make changes as required.

```
display sigchan/<signaling_channel> EtsiQsig
```

**2** If necessary, enable the EtsiQsig component to detect, segment, and reassemble large-sized ISDN messages.

```
set sigchan/<signaling_channel> EtsiQsig
msgSegmentation enabled
```

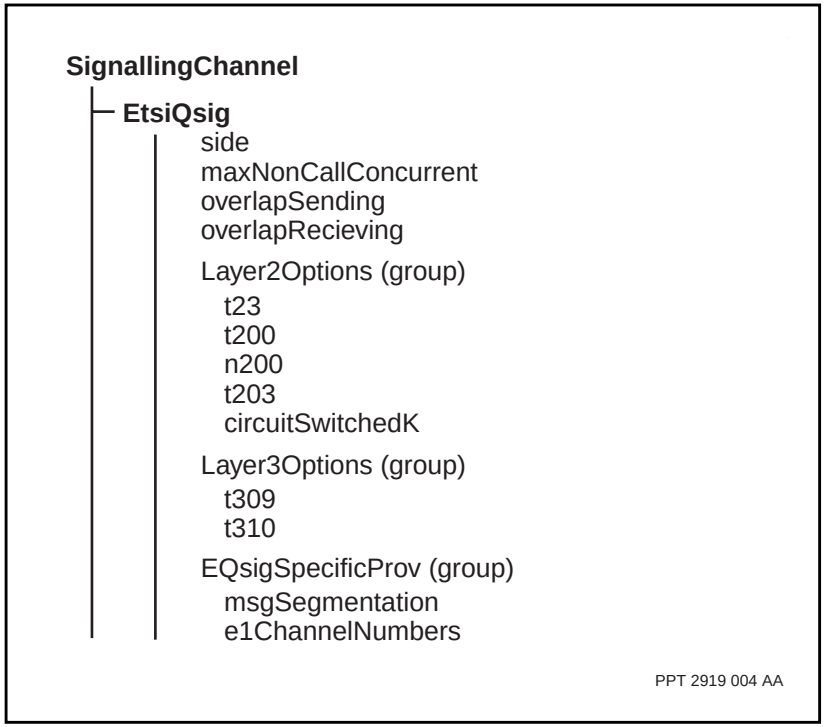
**3** If the connected PBX defines channels 1 to 30 for bearer services and the Passport node defines channels 1 to 15 and 17 to 31, map the PBX's channel assignments to those configured on the Passport node.

```
set sigchan/<signaling_channel> EtsiQsig
e1ChannelNumbers contiguous
```

**4** Repeat step 1 to step 3 for each signaling channel on each Passport node connected to a PBX that uses the ETSI QSIG signaling protocol.

## Procedure job aid

**Figure 9**
**Configuring the ETSI QSIG signaling protocol component hierarchy**



# Configuring the NIS A211-1 signaling protocol

Configure the NIS A211-1 signaling protocol to define the functional characteristics of the signaling channel's connection to the external equipment, for example, OSI Layer 2 and 3 timer settings.

## Prerequisites

- Review 241-5701-060 *Passport 7400, 15000, 20000 Components* for attribute descriptions and values related to configuring the NIS A211-1 signaling protocol (see "Configuring the NIS A211-1 signaling protocol component hierarchy" (page 51)).

## Procedure steps

1. Verify that the default settings of the NisSig component's provisionable attributes meet network requirements, and make changes as required.

   ```
   display sigchan/<signaling_channel> NisSig
   ```

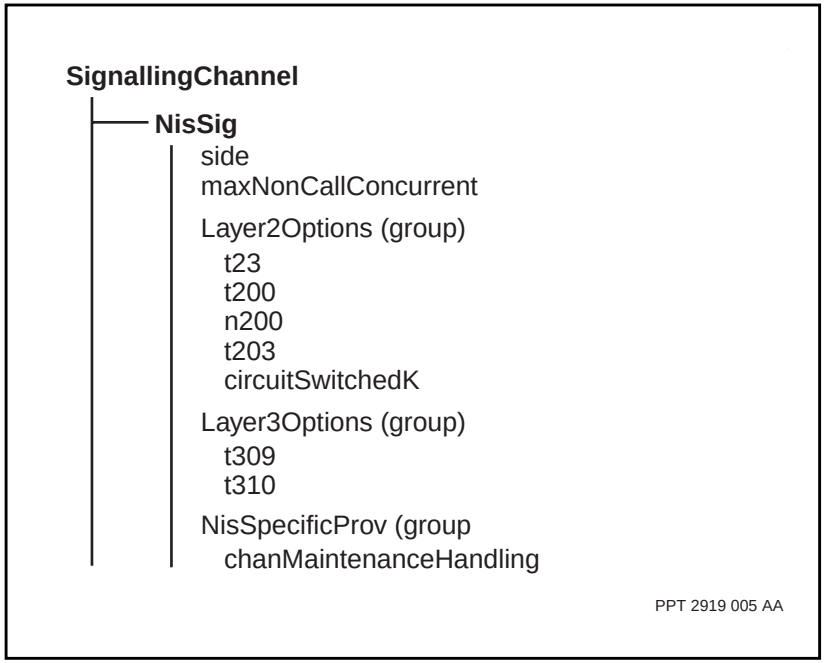
2. Repeat step 1 for each signaling channel on each Passport node connected to a PBX that uses the NIS A211-1 signaling protocol.

## Variable definitions

| Variable | Value |
|---|---|
| <signaling_channel> | The channel you defined to carry timeslot 16 signaling information for CAS or CCS protocols interfacing to an E1 port or 0 for CAS or CCS protocols interfacing to a Ds1 port. |
| | |

## Procedure job aid

**Figure 10**
**Configuring the NIS A211-1 signaling protocol component hierarchy**



# Configuring the MCDN signaling protocol

Configure the MCDN signaling protocol to define the functional characteristics of the signaling channel's connection to the Meridian 1 PBX, for example:

- OSI Layer 2 and 3 timer settings
- whether to enable or disable support for overlap dialing (in both the sending and receiving directions)
- how to handle channel registration with the Meridian 1 PBX, and the types of messages to send, on start up
- which MCDN connection-oriented and connectionless services to support

## Prerequisites

- Review 241-5701-060 *Passport 7400, 15000, 20000 Components* for attribute descriptions and values related to configuring the MCDN signaling protocol (see "Configuring the MCDN signaling protocol component hierarchy" (page 54)).

## Procedure steps

**1** Verify that the default settings of the McdnSig component's provisionable attributes meet network requirements, and make changes as required.

```
display -p sigchan/<signaling_channel> McdnSig
<provisionable_attribute>
```

**2** If, in your Passport network, certain nodes must have access restrictions (that is, calls must flow through certain tandem nodes), disable anti-tromboning on the minimum-access node.

```
set sigchan/<signaling_channel> McdnSig
conOrFeaturesSupported ~antiTromboning
```

**3** If required, make changes to the drop back busy service for congestion handling.

```
set sigchan/<signaling_channel> McdnSig
dropBackCongestion <dropback_value>
```

**4** If required, configure a particular type of network name display service. By default, Voice Networking does not enable MCDN network name display services.

```
set sigchan/<signaling_channel> McdnSig
networkNameDisplay <display_value>
```

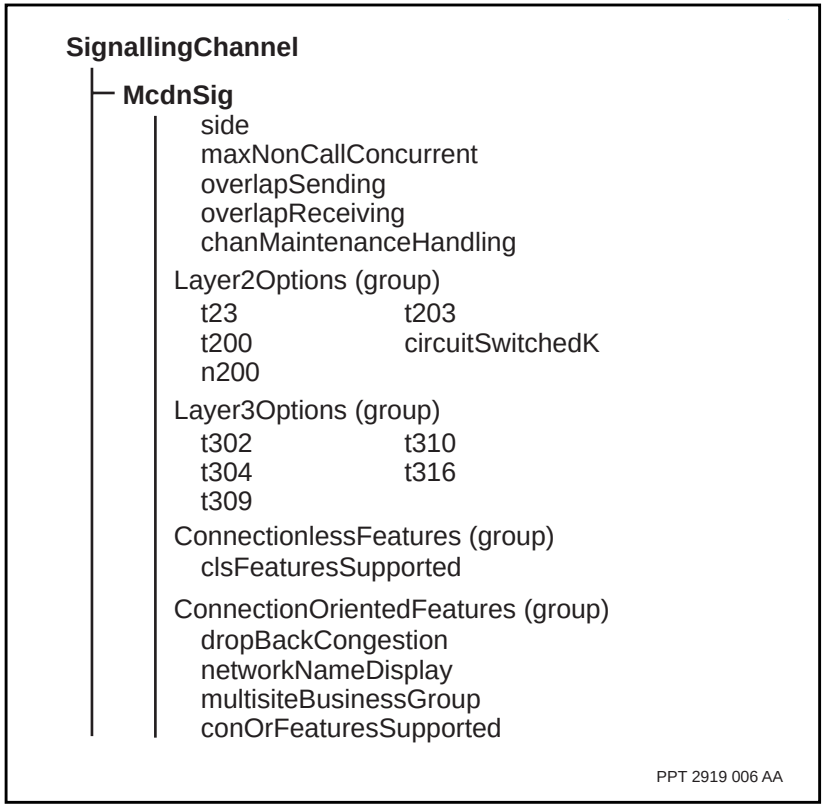
5 Repeat step 1 to step 4 for each signaling channel on each Passport node connected to a Meridian 1 PBX that uses the MCDN signaling protocol.

## Variable definitions

| Variable | Value |
|---|---|
| <display_value> | One of the following: nd1 (nameDisplay1), nd2 (nameDisplay2), or nd3 (nameDisplay3). |
| <dropback_value> | noDropBackAllowed, dropBackToOriginator (default), or dropBackToPriorNode |
| <provisionable_attribute> | Layer2Options, Layer3Options, Provisioned, ConnectionlessFeatures, or ConnectionOrientedFeatures |
| <signaling_channel> | The channel you defined to carry timeslot 16 signaling information for CAS or CCS protocols interfacing to an E1 port or 0 for CAS or CCS protocols interfacing to a Ds1 port. |
| | |

## Procedure job aid

**Figure 11**
**Configuring the MCDN signaling protocol component hierarchy**



# Configuring the Euro-ISDN signaling protocol

Configure the Euro-ISDN signaling protocol to define the functional characteristics of the signaling channel's connection to the PBX, for example:

- OSI Layer 2 and 3 timer settings
- various timer settings for non call-associated sessions (also referred to as virtual calls). Virtual calls provide bearer-independent connection-oriented transport for Euro-ISDN supplementary services.
- whether to enable or disable support for overlap dialing (in both the sending and receiving directions)
- the particular Euro-ISDN protocol variant to use

***Note:*** T309 for Euro-ISDN is calculated according to the specification as follows. T309 : 6 to 12 seconds, according to the formula (N200 + 1) x T200 + 2 seconds.

## Prerequisites

- Review 241-5701-060 *Passport 7400, 15000, 20000 Components* for attribute descriptions and values related to configuring the Euro-ISDN signaling protocol (see "Configuring the Euro-ISDN signaling protocol component hierarchy" (page 57)).
- For the 4-port E1 MVP-E FP, you must provision the protocol variant identically on all four ports.

## Procedure steps

**1** Verify that the default settings of the EuroIsdn component's provisionable attributes meet network requirements.

```
display SigChan/<signaling_channel> EuroIsdn
```

**2** Define the protocol variant.

```
set sigchan/<signaling_channel> euroisdn variant
<type>
```

**3** If you specified either austria or germany under the variant attribute, set the side attribute to user (the default value network applies if you set the variant attribute to etsiGeneric).

```
set sigchan/<signaling_channel> euroisdn side user
```

4 If necessary, change the default settings for tracking the amount of time involved in setting up and monitoring non call-associated sessions or virtual calls.

```
set sigchan/<signaling_channel> euroisdn
connectservicetimer <seconds>
```

```
set sigchan/<signaling_channel> euroisdn
responseservicetimer <seconds>
```

```
set sigchan/<signaling_channel> euroisdn
lifetimeservicetimer <minutes>
```

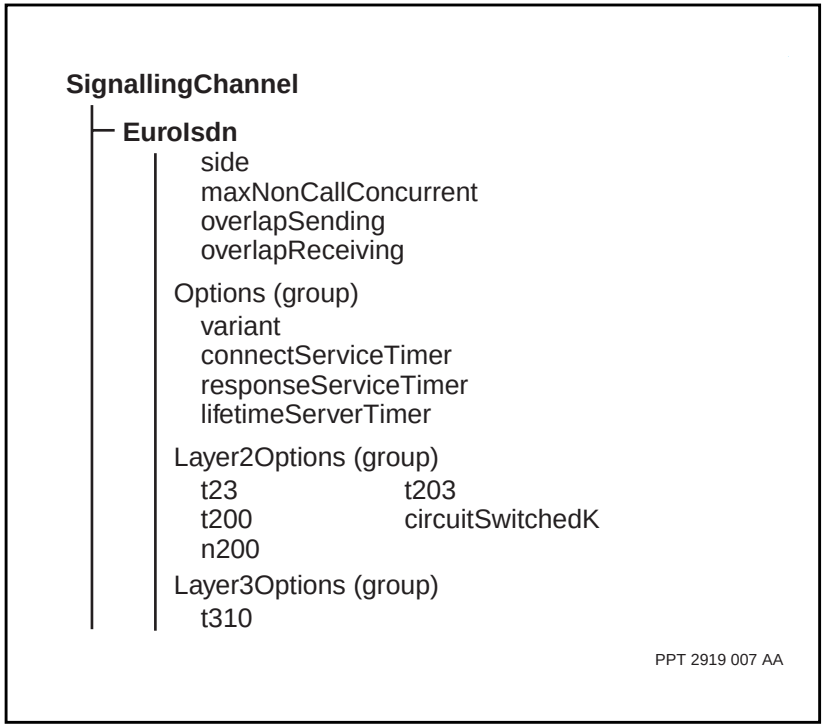
5 Repeat step 1 to step 4 for each signaling channel on each Passport node connected to a PBX or PSTN CO that uses the Euro-ISDN signaling protocol.

## Variable definitions

| Variable | Value |
|---|---|
| <minutes> | from 1 to 200 |
| <seconds> | from 1 to 10 |
| <signaling_channel> | The channel you defined to carry timeslot 16 signaling information for CAS or CCS protocols interfacing to an E1 port or 0 for CAS or CCS protocols interfacing to a Ds1 port. |
| <type> | etsiGeneric, austria, or germany<br><br>The default value etsiGeneric defines the network side of a connection.<br><br>The values austria and germany define the two possible user sides of a connection. |
| | |

## Procedure job aid

**Figure 12**
**Configuring the Euro-ISDN signaling protocol component hierarchy**



# Linking the signaling channel to the DS1 or E1 channel component

Link the signaling channels to the appropriate DS1 or E1 channel components.

## Procedure steps

1. Link each signaling channel to the E1 or DS1 Channel component instance that carries signaling information for the particular signaling protocol.

   **`set sigchan/<signaling_channel> <protocol> framer interfacename lp/<lp_number> <port>/<port_number> chan/<signaling_channel>`**

## Variable definitions

| Variable | Value |
|---|---|
| <lp_number> | The instance number of the LP. |
| <port> | E1 or Ds1 |
| <port_number> | The instance number of the port. |
| <protocol> | E1Cas, Ds1Cas, J2Cas, EtsiQsig, NisSig, McdnSig, or EuroIsdn<br><br>The protocol you specify corresponds to the protocol software you configured under the Software component and matches the signaling format—CAS or CCS—you specified under the E1 or DS1 port component. |
| <signaling_channel> | The channel you defined to carry timeslot 16 signaling information for CAS or CCS protocols interfacing to an E1 port or 0 for CAS or CCS protocols interfacing to a Ds1 port. |
| | |

# Chapter 4
# Voice Networking call server configuration

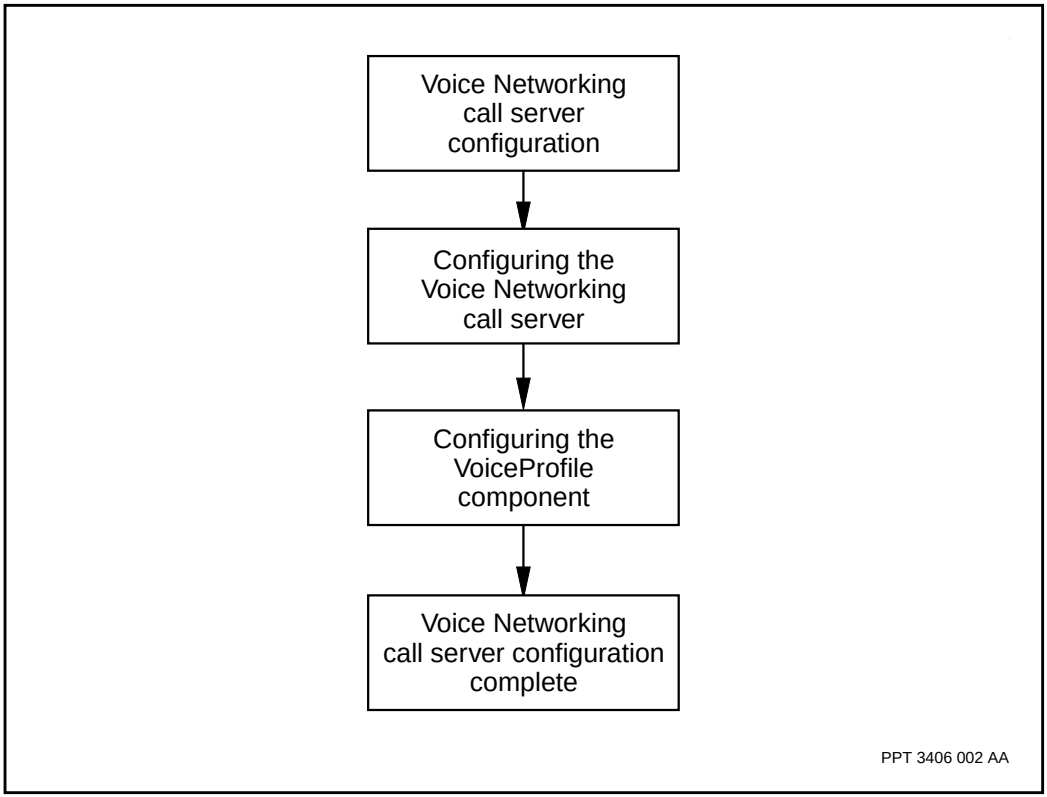
Configure the Voice Networking call server and its VoiceProfile component.


- “Voice Networking call server configuration task” (page 59)


## Voice Networking call server configuration task

“Voice Networking call server configuration task flow” (page 60) shows you the sequence of tasks and procedures you perform to configure the Voice Networking call server and the VoiceProfile component. To link to any task or procedure, go to “Task navigation” (page 60).

**Figure 13**
**Voice Networking call server configuration task flow**




## Task navigation

- “Configuring the Voice Networking call server” (page 61)
- “Configuring the VoiceProfile component” (page 65)

# Configuring the Voice Networking call server

Configure the voice networking call server (VNCS) to define quality of service parameters and dialing plan information for all channels and timeslots interfacing to a particular voice route.

## Prerequisites

- For more information about the CallRouter component, refer to 241-5701-405 *Passport 7400, 15000, 20000 Call Server Guide*.
- For more information about the DPN CSRM, refer to 241-7401-110 *Passport 7400, DPN-100 Interworking Guide*.
- For more information about DPRS, see 241-5701-425 *Passport 7400, 15000, 20000 Dynamic Packet Routing System Guide*.
- For more information on introducing DNA support, see "Voice Networking based on DNAs and DPRS" (page 151).

## Procedure steps

1. Add a voice networking call server (VNCS) for each numbering plan.

   **add vncs/<vncs_number>**

2. Add a directory number (DN).

   **add vncs/<vncs_number> dp/0 dn/<prefix>**

3. For Voice Networking without DNA support (where the migration to DNA-based addressing support is in the planning stages or in progress) configure the VNCS's directory number to use address resolution based on PORS destination node and component identification.

   **set vncs/<vncs_number> dp/<dp_number> dn/<prefix> destinationNodeId <nodeId>**

   **set vncs/<vncs_number> dp/<dp_number> dn/<prefix> destinationComponentId <Vroute>**

4. For Voice Networking with DNA support, configure the VNCS's directory number to use address resolution based on data network address (DNA) parameters.

   **set vncs/<vncs_number> dp/<dp_number> dn/<prefix> npi <npi>**

**set vncs/<vncs_number> dp/<dp_number> dn/<prefix> dna <dna>**

5 For a voice route on a remote node, ensure that the Passport CallRouter component or Data Packet Network (DPN) call server resource module (CSRM), is configured to recognize the DNAs that you added to the VNCS database.

6 For a hunt group server, see “Configuring call hunting through a hunt group server” (page 88) for the configuration procedure.

7 For Voice Networking with DNA support—where all directory numbers have been configured with DNA routing parameters—remove all destination node and component identification values.

**set vncs/<vncs_number> dp/<dp_number> dn/<prefix> destinationNodeId 0**

**set vncs/<vncs_number> dp/<dp_number> dn/<prefix> destinationComponentId**

8 Specify a particular voice profile for the configured directory number to use. Verify that the voice profile’s settings meet network requirements.

**display vncs/<vncs_number> vp/<vp_number>**

9 Repeat step 1 to step 8 to configure each VNCS on each Passport node connected to a PBX.

## Variable definitions

| Variable | Definition |
|---|---|
| <dna> | A binary coded decimal number between 0 and 15 digits in length that uniquely represents either the address of a voice route on a remote Passport node or a hunt group server that supports call hunting.<br><br>You can configure destination node and component identification and NPI and DNA values under directory numbers. If you do plan to migrate to DNA-based routing, having both sets of values configured makes the migration process easier. |
| <dp_number> | The dialing plan number. |
| <nodeId> | A decimal number between 0 and 4095 that corresponds to the value configured under the ModuleData component’s nodeId attribute on a remote Passport node. |

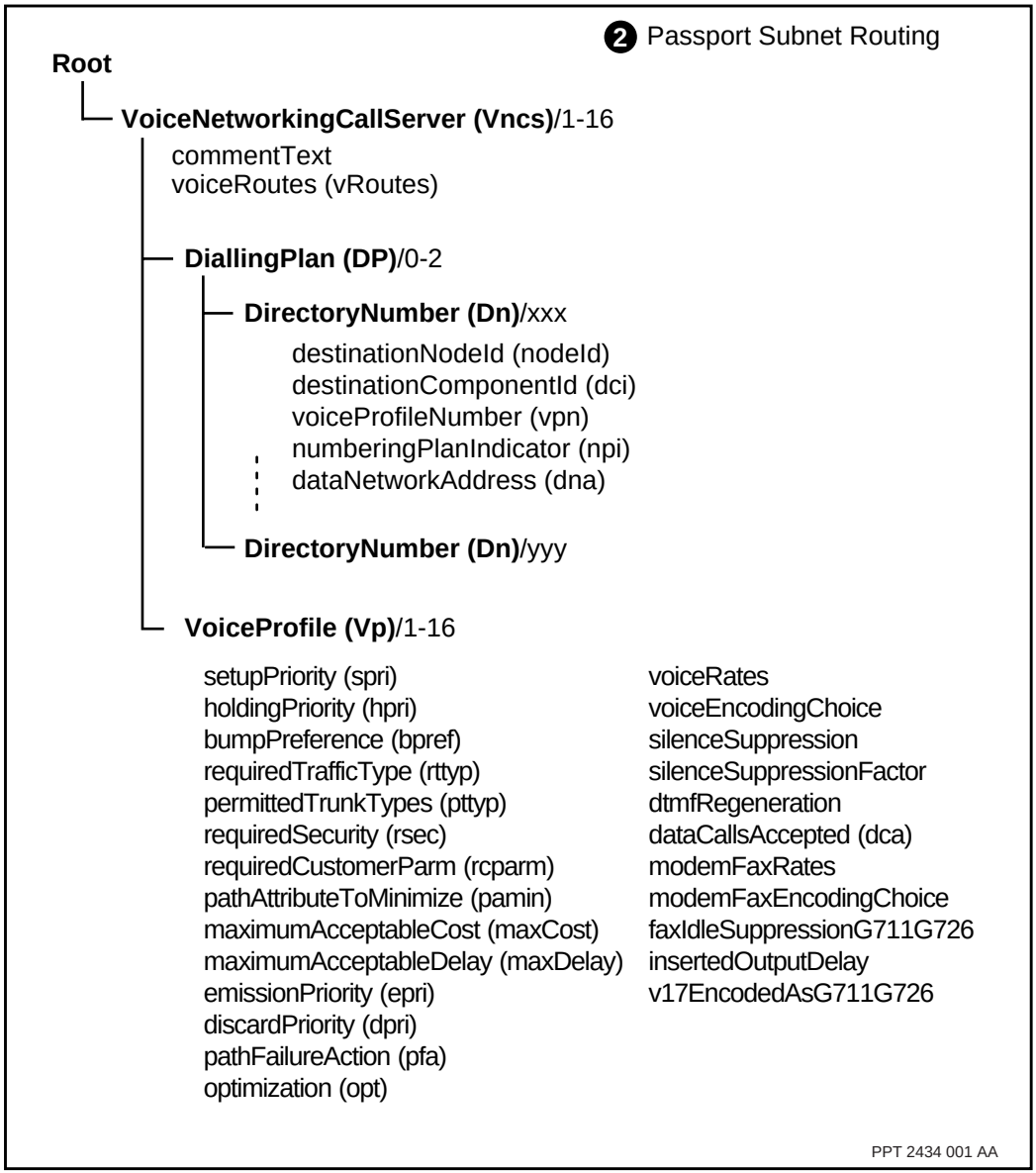
(Sheet 1 of 2)

| Variable | Definition |
|---|---|
| <npi> | Either x121or the default value e164. |
| <prefix> | You can provision up to 10 000 DirectoryNumber component instances for each DiallingPlan component instance.<br>1 to 40 digits in length; can include wildcards (represented by the? character).<br>For more information, see “Directory numbers” (page 135). |
| <vncs_number> | You can add up to 16 VoiceNetworkingCallServer component instances. |
| <vp_number> | The voice profile number. The system provides the VoiceProfile component instance Vp/1 by default. |
| <Vroute> | Represents a particular VoiceRoute component instance on a remote Passport node. |

(Sheet 2 of 2)

## Procedure job aid

**Figure 14**
**VoiceNetworkingCallServer component hierarchy**



# Configuring the VoiceProfile component

Configure the VoiceProfile component, including the definition of voice and modem/fax encoding choices and rates if the default settings do not meet network requirements.

## Prerequisites

**CAUTION**
**Possibility of service disruption**
By invoking the check, activate and confirm configuration commands, you risk disrupting the establishment of new calls as the new data network addresses take effect.

- For information on defining encoding choices and rates for voice, modem/fax, and fax traffic, see “Voice, modem, and facsimile encoding choices and rates” (page 168) and “Defining traffic handling options” (page 175).

## Procedure steps

1. To change the default voice encoding choice and rate values, use the following syntax as an example:

   ```
   set Vncs/1 Vp/1 voiceRates encoding g711G726 rate
   max 32

   set Vncs/1 Vp/1 voiceEncodingChoice first g728Only
   ```

2. To change the default modem/fax encoding choice and rate values, use the following syntax as an example:

   ```
   set Vncs/1 Vp/1 modemFaxRates encoding g711G726 rate
   max 32

   set Vncs/1 Vp/1 modemFaxEncodingChoice first g711G726
   ```

3. If necessary, alter the minimum encoding rates to use for dynamic up- and down-speeding of voice, modem/fax and fax traffic.

4. If necessary, alter the default setting of speech activity detection (SAD). SAD applies to voice and modem/fax traffic on MVP-E FPs.

   ```
   set Vncs/1 Vp/1 silenceSuppression <value>
   ```

**5** If network jitter occurs, modify the settings of the configurable egress buffer. The egress buffer applies to voice, modem/fax, and fax traffic on MVP-E FPs.

```
set Vncs/1 Vp/1 insertedOutputDelay <# msec>
```

**6** To conserve bandwidth during facsimile transmissions, enable fax idle suppression (FIS).

```
set Vncs/1 Vp/1 faxIdleSuppressionG711G726 on
```

**7** If necessary, enable MVP-E FPs to specify that calls involving facsimile machines that operate at 14.4 kbit/s (as defined in ITU-T V.17) use ADPCM modem/fax traffic encoding rates.

```
set Vncs/1 Vp/1 v17EncodedAsG711G726 yes
```

**8** Enable DTMF tone detection and regeneration to prevent the alteration or loss of DTMF tones. DTMF tone detection and regeneration applies to voice traffic on MVP-E FPs only.

```
set Vncs/1 Vp/1 dtmfRegeneration on
```

**9** Repeat step 1 to step 8 to configure each Vncs Vp component on each Passport node in the network that connects to a PBX.

## Variable definitions

| Variable | Definition |
|---|---|
| <# msec> | 5 to 150 milliseconds<br><br>The default value is 22 milliseconds.<br><br>To avoid cell loss, set the insertedOutputDelay attribute higher if high cell delay variations occur in your network. |
| <value> | One of the following settings:<br><br>• off (silence suppression is never performed)<br>• on (the default value; silence suppression is always performed)<br>• congested (silence suppression is applied only when the network is congested)<br>• slow (silence suppression begins when no speech or modem traffic is detected for 10 to 20 seconds)<br>• slowAndCongested (both the slow and congested options are applied)<br><br>End-to-end negotiation determines which setting to support. |
| | |

# Chapter 5
# Voice route and subroute configuration

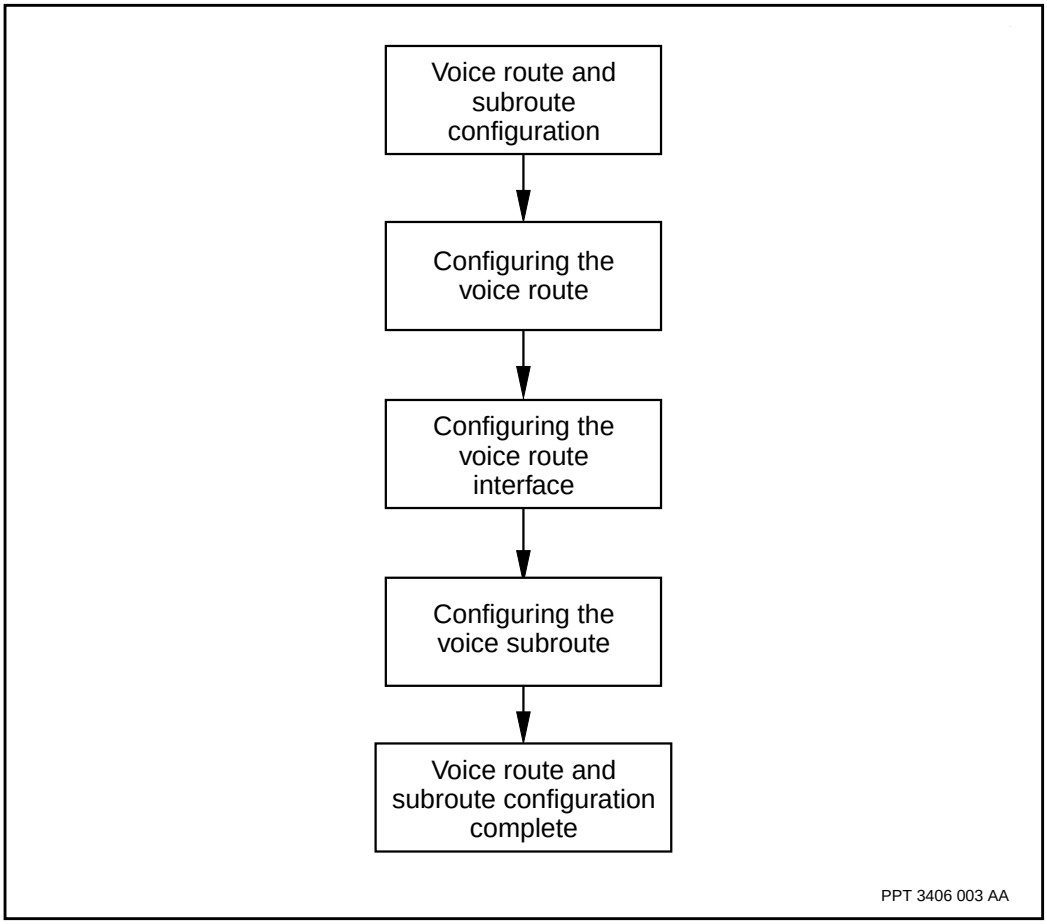
Configure the voice route to define the common characteristics—audio handling capabilities, DNA and dialed number parameters, and accounting options—for a set of timeslots interfacing to a PBX. Configure one VoiceRoute component instance for each connected PBX or customer group.


- "Voice route and subroute configuration task" (page 69)


## Voice route and subroute configuration task

"Voice route and subroute configuration task flow" (page 70) shows you the sequence of tasks and procedures you perform to configure the voice route and subroute. To link to any task or procedure, go to "Task navigation" (page 70).

**Figure 15**
**Voice route and subroute configuration task flow**




## Task navigation

- “Configuring the voice route” (page 71)
- “Configuring the voice route interface” (page 74)
- “Configuring the voice subroute” (page 80)

# Configuring the voice route

Configure voice routes.

## Prerequisites

- Review 241-5701-060 *Passport 7400, 15000, 20000 Components* for attribute descriptions and values related to configuring the voice route (see “Configuring the voice route component hierarchy” (page 73)).

## Procedure steps

1. Add one VoiceRoute component for each customer group or connected PBX.

   **add voiceroute/<vroute_instance>**

2. Verify that the default settings of the provisionable attributes under the VoiceRoute component meet network requirements, and make changes as required.

   **display vroute/<vroute_instance>**

3. If necessary, define values for the diallingPlan attributes.

   **set vroute/<vroute_instance> diallingPlan<0|1|2> <type>**

4. Define the minimum number of digits that this voice route needs to receive before initializing an address resolution request to the VNCS.

   **set vroute/<vroute_instance> minimumDigitsToRoute <minimum_digits>**

5. Link the voice route to the Voice Networking call server.

   **set vroute/<vroute_instance> voiceNetworkingCallServer vncs/<vncs_number>**

6. Specify the private network identifier (PNI) number to enable MCDN connectionless services.

   **set vroute/<vroute_instance> privateNetworkIdentifier <pni_number>**

7. For Voice Networking with DNA support, specify DNA parameters to uniquely identify this voice route.

   **set vroute/<vroute_instance> Dna npi <npi>**

   **set vroute/<vroute_instance> Dna dna <dna_number>**

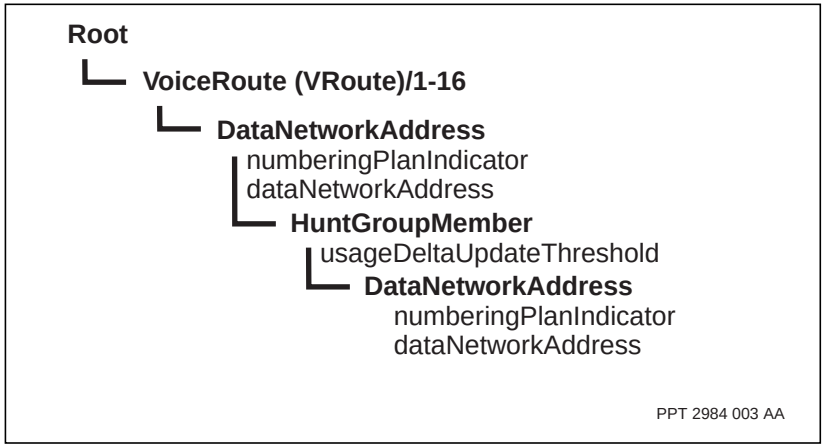
**8** Repeat step 1 to step 7 to configure each VoiceRoute and VoiceRoute Dna component on each Passport node connected to a PBX.

## Variable definitions

| Variable | Definition |
|---|---|
| <dna_number> | A binary coded decimal number between 0 and 15 digits in length (for example 30214111560001). |
| <minimum_digits> | Any decimal number between 1 and 16 (the default is 3).<br><br>To ensure that the voice networking call server (VNCS) is not overwhelmed by requests (the value is set too low) or taking too long to resolve dialed numbers (the value is too high), set the number to equal the average number of digits needed to uniquely identify a remote end-point. |
| <npi> | x121 or e164<br><br>x121 is for X.121, the international numbering plan for public packet switched networks. The default value e164 is for E.164, the international numbering plan for ISDN and the PSTN. |
| <pni_number> | 0 (default) up to 32700. The default value 0 means that this voice route does not support MCDN connectionless services.<br><br>The number you specify must correspond exactly to the particular customer PNI value on the connected PBX.<br><br>***Note:*** Do not use the *privateNetworkIdentifier* attribute under the SignallingChannel McdnSig component. That attribute is not used in the configuration of MCDN connectionless services. |
| <type> | One or more of the following: unknown, international, national, subscriber, p0 up to p7, and casUnknown. |
| <vncs_number> | One of the VoiceNetworkingCallServer component instances. |
| <vroute_instance> | A value from 1 to 153. |
| | |

## Procedure job aid

**Figure 16**
**Configuring the voice route component hierarchy**



# Configuring the voice route interface

Configure the characteristics of the voice route interface.

## Prerequisites

- Review 241-5701-060 *Passport 7400, 15000, 20000 Components* for attribute descriptions and values related to configuring the voice route interface (see “Configuring the voice route interface component hierarchy” (page 79)).
- See “Voice Networking call server configuration” (page 166) for more information on configuring speech activity detection and about enabling fax idle suppression.

## Procedure steps

1. Verify that the default settings of the provisionable attributes under the VoiceRoute Interface component meet network requirements, and make changes as required.

   **`display Vroute/<vroute_instance> Interface`**

2. If necessary (for example, when external echo cancellation devices exist in the network), disable internal echo cancellation on voice FPs,

   **`set Vroute/<vroute_instance> echoCancellation v1 off`**

3. If necessary (for example, when external echo cancellation devices exist in the network), disable internal echo cancellation on MVP-E FPs.

   **`set Vroute/<vroute_instance> echoCancellation v2 off`**

4. On MVP-E FPs when the echoCancellation attribute’s v2 row is set to on, define when to bypass the on-board echo canceller on MVP-E FPs.

   **`set Vroute/<vroute_instance> Interface ecanBypassMode <type>`**

5. On MVP-E FPs when the echoCancellation attribute’s v2 row is set to on, if necessary, adjust the echo delay coverage and signal attenuation parameters on MVP-E FPs.

   **`set Vroute/<vroute_instance> Interface echoTailDelay <echodelay_msec>`**

   **`set Vroute/<vroute_instance> Interface echoReturnLoss <echoloss_dB>`**

**6** On MVP-E FPs when the echoCancellation attribute's v2 row is set to on, set the residual echo control to determine how to treat echo that remains after cancellation.

**set Vroute/<vroute_instance> Interface residualEchoControl <control>**

**7** On MVP-E FPs when the echoCancellation attribute's v2 row is set to on, set the reenableEcan attribute to determine when the echo canceller is going to reenable echo cancellation after being disabled by the presence of a modem tone.

**set Vroute/<vroute_instance> Interface reenableEcan <condition>**

**8** On MVP-E FPs when the echoCancellation attribute's v2 row is set to on, set the length of the echo path to determine the amount of endpath delay the echo canceller can handle.

**set Vroute/<vroute_instance> Interface echoPathLength <path_length>**

**9** On MVP-E FPs when the echoCancellation attribute's v2 row is set to on, set the freezeHRegister attribute for testing or normal operation.

**set Vroute/<vroute_instance> Interface freezeHRegister <freeze_value>**

**10** To assist with network loss planning, apply a gain or a loss to audio data entering or leaving the network, or both.

**set Vroute/<vroute_instance> Interface ingressAudioGain <audiogain_+/-dB>**

**set Vroute/<vroute_instance> Interface egressAudioGain <audiogain_+/-dB>**

**11** Enable tandem pass through (TPT) if the network routes voice traffic through a tandem PBX.

**set Vroute/<vroute_instance> Interface tandemPassThrough enabled**

**12** To prevent the clipping of speech during telephone conversations when using speech activity detection on MVP-E FPs, specify the amount of time that elapses after the end of a speech burst before applying silence suppression.

**set Vroute/<vroute_instance> Interface speechHangoverTime <speechtime_msec>**

**13** Specify a cap for the level of background or comfort noise you want generated when speech activity detection is enabled (that is, when the negotiated value of the silenceSuppression attribute is on).

**`set Vroute/<vroute_instance> Interface comfortNoiseCap <comfortnoise_dBm0>`**

**14** To prevent the clipping of parts of facsimile transmission when using fax idle suppression on MVP-E FPs, specify the amount of time that elapses after the end of a facsimile transmission burst before applying fax idle suppression.

**`set Vroute/<vroute_instance> Interface faxHangoverTimeG711G726 <faxtime_msec>`**

**15** Repeat step 1 to step 14 to configure each VoiceRoute Interface component on each Passport node connected to a PBX.

## Variable definitions

| Variable | Definition |
|---|---|
| <audiogain_+/-dB> | For ingressAudioGain, can be +12 to -12 dB, in 1 dB increments, on MVP-E FPs.<br><br>For egressAudioGain, the value can be +12 to -12 dB, in 1 dB increments, on MVP-E FPs only. |
| <comfortnoise_dBm0> | -40 to -78 dBm0<br><br>The default value -40 dBm0 effectively disables the cap as it represents the maximum level of comfort noise that can be generated. |
| <condition> | modemDataGone or endOfCall<br><br>The default value of modemDataGone value reenables echo cancellation when the energy level is below -36 dBm0 for 300ms. |
| <control> | cancelOnly, suppressResidual, or comfortNoise<br><br>The default for this value is comfortNoise. The comfortNoise value replaces the residual echo with noise at the same level as the ambient noise at the near end.<br><br>The values cancelOnly and suppressResidual should be used only for testing. |

(Sheet 1 of 3)

| Variable | Definition |
|---|---|
| <echodelay_msec> | 32 or 64 milliseconds<br><br>The default value of 64 milliseconds is the recommended setting as it provides the greater amount of coverage. |
| <echoloss_dB> | 0, 3, or 6 decibels<br><br>You can set the echoReturnLoss attribute to one of the higher decibel values when line conditions are poor and echo is a problem. Setting the echoReturnLoss attribute higher can improve echo cancellation performance, but reduces the signal level. |
| <faxtime_msec> | Any value between 300 and 20000 milliseconds (the default value is 1000 milliseconds).<br><br>Fax idle suppression must be enabled (that is, the negotiated value of the faxIdleSuppressionG711G726 attribute must be on) and the negotiated modem/fax encoding choice must be g711G726 or g726 for the faxHangoverTimeG711G726 attribute to take effect. Setting the faxHangoverTimeG711G726 attribute higher reduces clipping but increases bandwidth usage. |
| <freeze_value> | yes or no<br><br>Use the default value of no during normal operation. Use yes during testing. |
| <path_length> | 0, 16, 32, 48, or 64 msec<br><br>The default for this value is 32 msec. |
| <speechtime_msec> | 10 to 500 milliseconds, in 10 millisecond increments<br><br>The default value is 150 milliseconds. Setting the speechHangoverTime attribute higher reduces clipping but increases bandwidth usage. The negotiated value of the silenceSuppression attribute must be the value on for the speechHangoverTime attribute to take effect. |

(Sheet 2 of 3)

| Variable | Definition |
|---|---|
| <type> | One of g165 (default), g164, or never.<br><br>The values g164 and g165 specify that the on-board echo canceller be placed in bypass mode according to ITU-T G.164 (receiving a 2100 Hz tone) and G.165 (receiving a phase-reversed 2100 Hz tone), respectively.<br><br>Use never during debugging procedures only. |
| <vroute_instance> | A value from 1 to 153. |

(Sheet 3 of 3)

## Procedure job aid

**Figure 17**
**VoiceRoute component hierarchy**

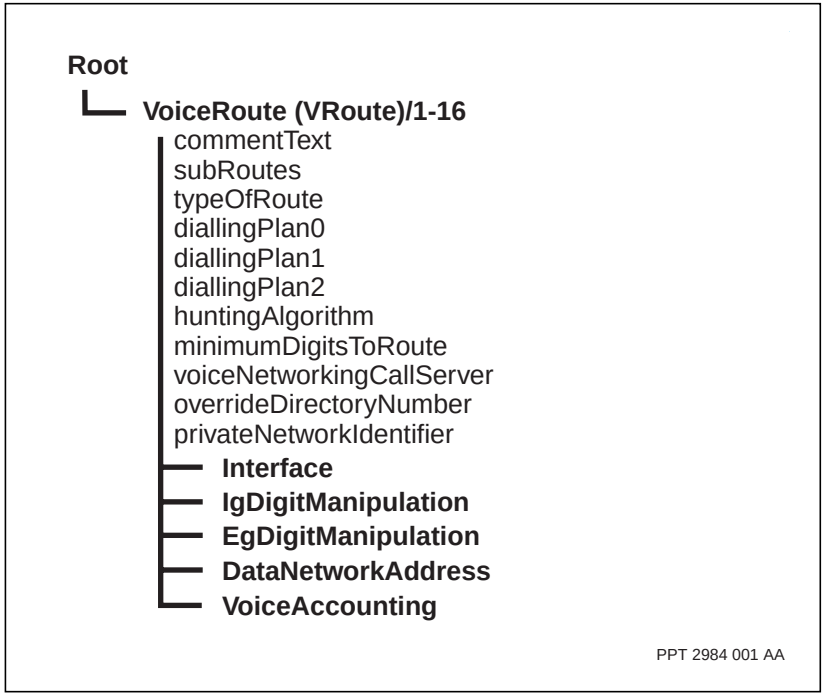
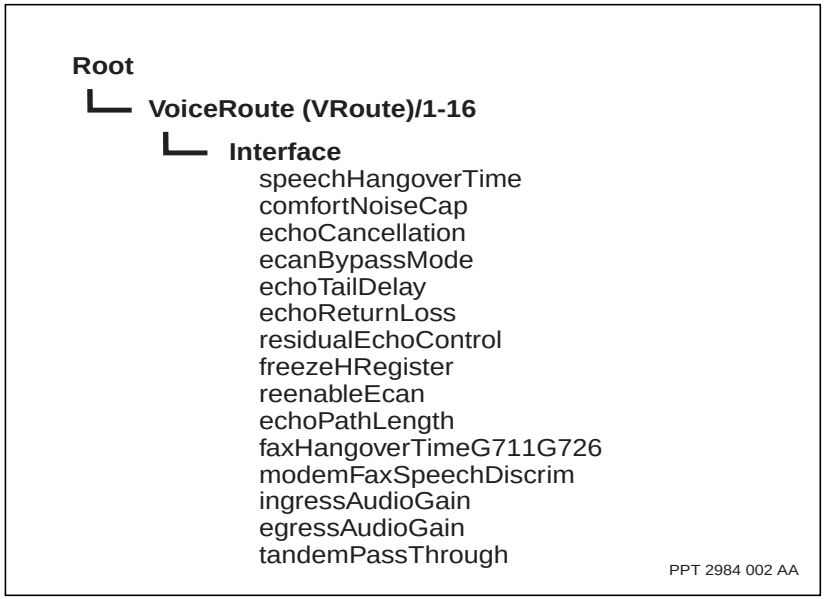


**Figure 18**
**Configuring the voice route interface component hierarchy**



# Configuring the voice subroute

Configure the voice subroute through the VoiceSubroute component and assign each SwitchedVoiceService component to a particular channel and timeslot pair.

## Prerequisites

- See “Logical and function processors” (page 106) for more information about assigning channel and timeslot numbers for CAS and CCS protocols interfacing to E1 or DS1 ports.

## Procedure steps

**1** Add a voice subroute.

```
add VoiceSubroute/<vsr_instance>
```

**2** Add a switched voice service.

```
add VoiceSubroute/<vsr_instance>
SwitchedVoiceService/<svs_number>
```

**3** Link the voice subroute to the signaling channel.

```
set vsr/<vsr_instance> sigchan sigchan/
<sigchan_number>
```

**4** Link the voice subroute to the voice route.

```
set vsr/<vsr_instance> voiceRoute vroute/
<vroute_instance>
```

**5** Link each switched voice service to the bearer services configured under the E1 or DS1 port.

```
set vsr/<vsr_instance> svs/<svs_number> framer
interfacename lp/<lp_number> <port>/0 chan/
<channel_number>
```

**6** Repeat step 5 for each SwitchedVoiceService component instance you provision.

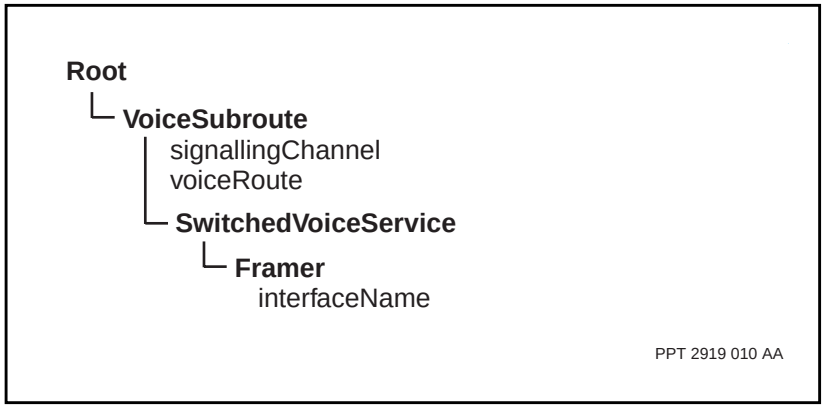
**7** Repeat step 1 to step 6 for each VoiceSubroute component instance on each Passport node connected to a PBX.

## Variable definitions

| Variable | Definition |
|---|---|
| <channel_number> | Depends on the protocol you configure under the SignallingChannel component.<br><br>Although not mandatory, attempt to align SwitchedVoiceService component number instances with the timeslots attribute numbers configured under Channel component instances. By aligning timeslot and SwitchedVoiceService component number instances, it is easier to identify problems and gather performance information. |
| <lp_number> | The instance number of the LP. |
| <port> | Either e1 or ds1. |
| <svs_number> | For a DS1 port, you can add up to 23 SwitchedVoiceService components for bearer services. For an E1 port, you can add up to 30 SwitchedVoiceService components for bearer services. |
| <vsr_instance> | The VoiceSubroute instance values range from 1 to 1534. A maximum of 255 VoiceRoutes can be provisioned on one shelf. |
| | |

## Procedure job aid

**Figure 19**
**Configuring the voice subroute component hierarchy**



# Chapter 6
# Other Voice Networking options configuration

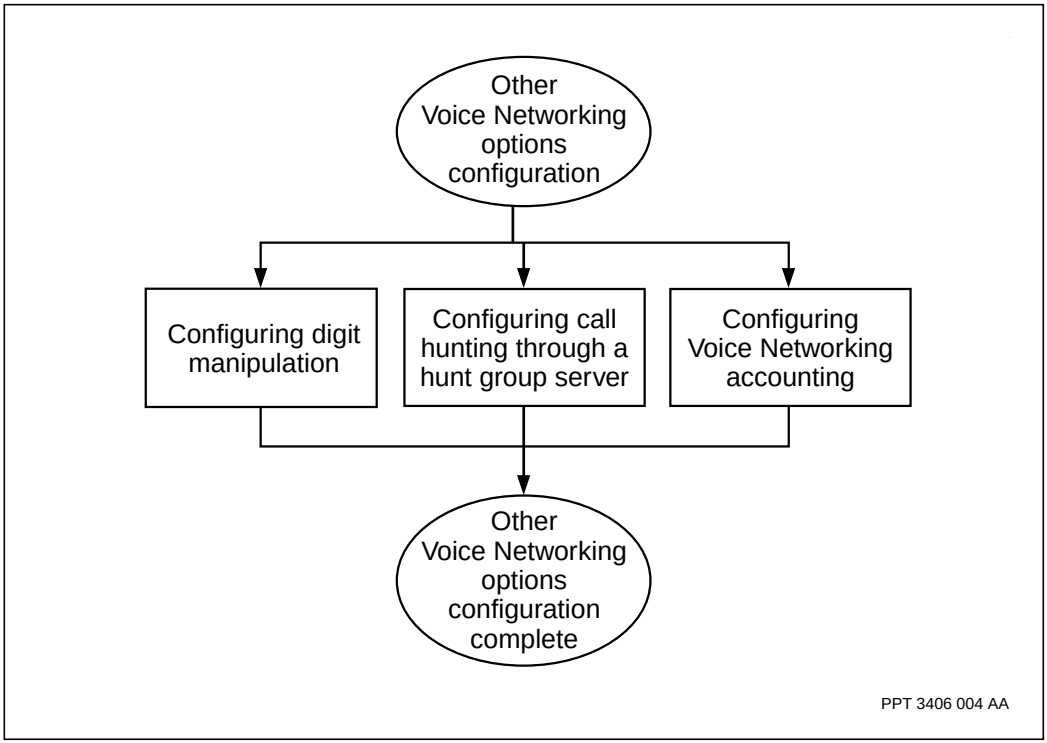
Configure some or all of the Voice Networking options as required by your network.


- “Other Voice Networking options configuration” (page 83)


## Other Voice Networking options configuration

“Other Voice Networking options configuration task flow” (page 84) shows you the sequence of tasks and procedures you perform to configure other Voice Networking options. To link to any task or procedure, go to “Task navigation” (page 84).

**Figure 20**
**Other Voice Networking options configuration task flow**




## Task navigation

- “Configuring digit manipulation” (page 85)
- “Configuring call hunting through a hunt group server” (page 88)
- “Configuring Voice Networking accounting” (page 90)

# Configuring digit manipulation

Configure digit manipulation by adding the ingress, egress, or both digit manipulation components to each corresponding voice route, and then defining one or more of these capabilities:

- digit manipulation variant
- inserted digits
- number of digits to delete
- call routing (ingress only)
- call handling in the case of no digits (egress only)

## Procedure steps

1. Add the digit manipulation component for either ingress or egress.

   **add VoiceRoute/<vroute_instance> <dm_component>**

2. Define the application of digit manipulation.

   **set VRoute/<vroute_instance> <dm_component> variant <var>**

3. Define the digits to insert into the called number.

   **set VRoute/<vroute_instance> <dm_component> digitsToInsert <digits>**

4. Define the number of digits to delete from the called number.

   **set VRoute/<vroute_instance> <dm_component> numDigitsToDelete <digits_delete>**

5. For digit manipulation on ingress, define which number call routing acts on.

   **set VRoute/<vroute_instance> <dm_component> callRouting <cr_point>**

6. For digit manipulation on egress, define how voice networking handles calls for which no digits remain after digit manipulation.

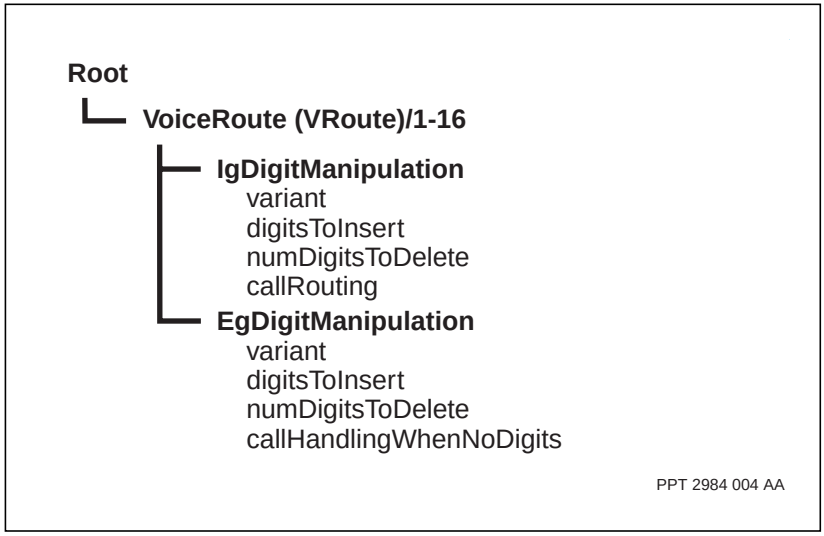
   **set VRoute/<vroute_instance> <dm_component> callHandlingWhenNoDigits <choice>**

## Variable definitions

| Variable | Definition |
|---|---|
| <choice> | Either attemptCompletion or releaseCall.<br><br>The default value is releaseCall. |
| <cr_point> | Either routeOnDialledDigits or routeOnModifiedDigits.<br><br>The value routeOnDialledDigits specifies that calls are routed prior to performing ingress digit manipulation.<br><br>The default value routeOnModifiedDigits specifies that calls are routed after performing ingress digit manipulation. |
| <digits> | A string of up to 16 characters, including the digits 0 through 9, the character # (octothorpe), and the character * (asterisk).<br><br>There is no default value.<br><br>If digit insertion is not a requirement, enter two double quotes <“”> to indicate a blank string. |
| <digits_delete> | A decimal from 0 to 16. The default value is 0. |
| <dm_component> | Either IgDigitManipulation or EgDigitManipulation |
| <var> | Either blindLeading or none.<br><br>The default value is blindLeading<br><br>Selecting the option none disables digit manipulation for that particular component. The system provides a warning if none is selected and the digitsToInsert attribute is not empty or the numDigitsToDelete attribute is not 0. |
| <vroute_instance> | The voice route instance. |
| | |

## Procedure job aid

**Figure 21**
**Configuring digit manipulation component hierarchy**



# Configuring call hunting through a hunt group server

Configure call hunting by setting up a hunt group server and defining the voice routes as hunt group members.

## Prerequisites

- See 241-5701-415 *Passport 7400, 15000, 20000 Hunt Group Server Guide* for descriptions and procedures on setting up a hunt group server.
- See 241-5701-410 *Passport 7400, 15000, 20000 Call Redirection Server Guide* for descriptions and procedures on the CRS.

## Procedure steps

1. Configure the primary hunt group server.
2. Optionally, configure a backup hunt group server.
3. Provision the call redirection server (CRS) to associate the backup server with the primary server.
4. Identify the DNA of the primary hunt group server in the dataNetworkAddress attribute under the VoiceNetworkingCallServer DiallingPlan DirectoryNumber component.
5. Define the voice route as a hunt group member and add the primary VoiceRoute DataNetworkAddress HuntGroupMember HuntGroupAddress/0 component instance.

   **`add Vroute/<vroute_instance> DataNetworkAddress HuntGroupMember`**

6. Optionally, set the number of channels that must be free or occupied before the availability message packet (AMP) is sent to the primary and backup hunt group servers.

   **`set Vroute/<vroute_instance> DataNetworkAddress HuntGroupMember usageDeltaUpdateThreshold <udut>`**

7. If a backup hunt group server is configured, add the component for this backup under the voice route DNA.

   **`add Vroute/<vroute_instance> DataNetworkAddress HuntGroupMember HuntGroupAddress/1`**

8. Set the address of the primary hunt group server to which the VoiceRoute DNA belongs.

```
set Vroute/<vroute_instance> DataNetworkAddress
HuntGroupMember HuntGroupAddress/0
numberingPlanIndicator <npi>
```

```
set Vroute/<vroute_instance> DataNetworkAddress
HuntGroupMember HuntGroupAddress/0 dataNetworkAddress
<dna>
```

9 If a backup hunt group server is available, set the address of this backup.

```
set Vroute/<vroute_instance> DataNetworkAddress
HuntGroupMember HuntGroupAddress/1
numberingPlanIndicator <npi>
```

```
set Vroute/<vroute_instance> DataNetworkAddress
HuntGroupMember HuntGroupAddress/1 dataNetworkAddress
<dna>
```

10 Repeat step 5 to step 9 to configure each voice route that is a member of the hunt group.

## Variable definitions

| Variable | Definition |
|---|---|
| <dna> | An address string of between 1 and 15 characters, which identifies the DNA of the hunt group server. |
| <npi> | Either x121 or e164. The default is x121. |
| <udut> | A value between 1 and 4096. The default is 1. |
| <vroute_instance> | The voice route instance. |
| | |

# Configuring Voice Networking accounting

Set up Passport to collect accounting statistics for calls processed by the VoiceRoute component.

## Prerequisites

- Configure basic Passport accounting using the procedures in 241-5701-650 *Passport 7400, 15000, 20000 Accounting Fundamentals*.

## Procedure steps

**1** Configure DNA parameters for the VoiceRoute component (if you have not already done so).

```
set VRoute/<vroute_instance> Dna dna <dna#>

set VRoute/<vroute_instance> Dna npi <type>
```

**2** Specify the reason for generating accounting records.

```
set VRoute/<vroute_instance> VoiceAccounting
accountCollection <reason>
```

**3** Specify an accounting class.

```
set VRoute/<vroute_instance> VoiceAccounting
accountClass <accountclass#>
```

**4** Specify a data service exchange value.

```
set VRoute/<vroute_instance> VoiceAccounting
serviceExchange <serviceexchange#>
```

**5** Specify the number of trailing digits to be removed from the accounting records.

```
set VRoute/<vroute_instance> VoiceAccounting
digitsSuppressed <digitssuppressed#>
```

**6** Specify accounting options.

```
set VRoute/<vroute_instance> VoiceAccounting
accountingOptions suppressTerminatingEndRecords
```

***Note:*** The value suppressTerminatingEndRecords stops all accounting currently in progress for this VoiceRoute component instance.

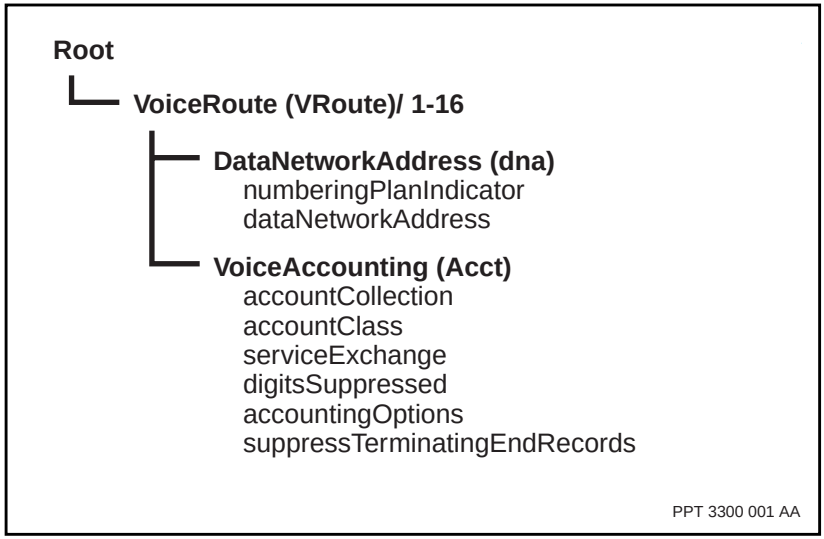
**7** Repeat steps 1 to 6 to configure Voice Networking accounting on each candidate Passport node.

## Variable definitions

| Variable | Definition |
|---|---|
| <accountclass#> | Any decimal number between 0 and 255. The default value is 0. |
| <digitssuppressed#> | Any decimal number between 0 and 8. The default value is 0. Digits are not suppressed if the value is 0. |
| <dna#> | A binary coded decimal number between 0 and 15 digits in length, for example 30214111560001. |
| <reason> | Can be bill, test, study, audit, or force.<br>The accountCollection attribute has no default value. If you do not set a value for the accountCollection attribute, Passport does not collect accounting statistics for this VoiceRoute component. |
| <serviceexchange#> | Any decimal number between 0 and 255. The default value is 0. |
| <type> | x121 or e164<br>x121 is for X.121, the international numbering plan for public packet switched networks.<br>e164 is the default value and is for E.164, the international numbering plan for ISDN and the PSTN. |
| <vroute_instance> | The voice route instance. |
| | |

## Procedure job aid

**Figure 22**
**Configuring Voice Networking accounting component hierarchy**



# Chapter 7
# Voice Networking fundamentals

The following sections provide an overview of the Voice Networking service:


- "What is Voice Networking?" (page 93)
- "What software does Voice Networking require?" (page 96)
- "What hardware does Voice Networking require?" (page 97)
- "How does Voice Networking communicate with PBXs?" (page 97)
- "How does Voice Networking handle audio data?" (page 98)
- "How does Voice Networking route calls?" (page 98)
- "What is the relationship between Voice Networking software components and systems?" (page 99)
- "What configuration parameters impact Voice Networking?" (page 105)
- "Voice Networking accounting" (page 111)


## What is Voice Networking?

The Voice Networking service dynamically routes speech, modem, facsimile, and data calls through a Passport network to addresses (dialed numbers) provided by the calling PBX.

In a Passport network without Voice Networking (see the figure "Passport network without Voice Networking" (page 95)), permanent virtual circuits (PVC) transport calls between the ingress and egress nodes. PVCs, although continuously available,

- do not always provide the most direct or efficient route

- reserve a certain amount of bandwidth even when idle
- require that you define and configure both endpoints of a connection
- can increase the potential for signal delays and distortions. For example, a call from PBX B to PBX E, as shown in the figure “Passport network without Voice Networking” (page 95), requires routing through two tandem PBXs.

In a Passport network with Voice Networking, the best path between the ingress and egress Passport nodes is selected on a call-by-call basis by means of a switched virtual circuit (SVC). As shown in the figure “Passport network with Voice Networking” (page 96), Voice Networking routes a call from PBX B to PBX E over an SVC that represents the most efficient path and accounts for configured parameters and bandwidth availability at the time of the call. When a Voice Networking call clears, the SVC is torn down, freeing up bandwidth in the subnet for subsequent calls. With Voice Networking, you only provision one endpoint of a connection—the dialed number.

**Figure 23**
**Passport network without Voice Networking**

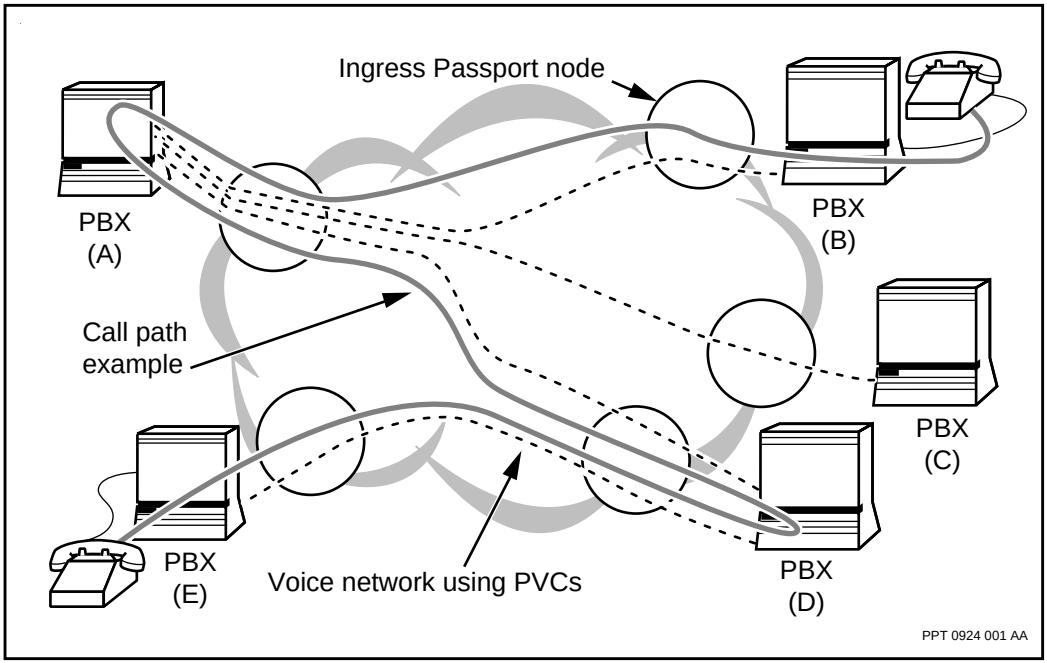
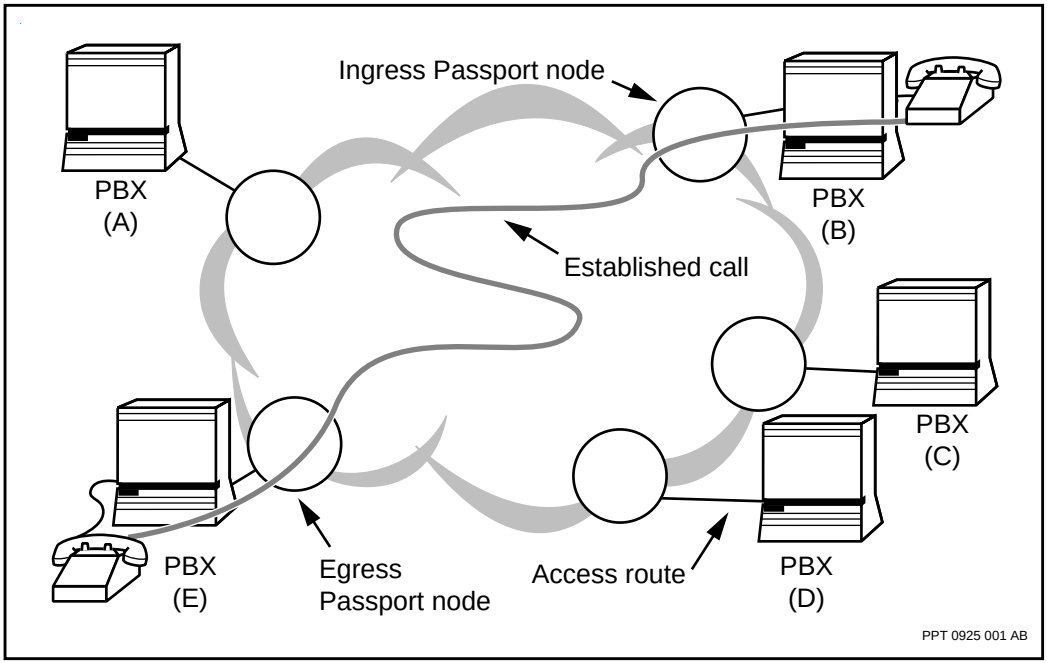


**Figure 24**
**Passport network with Voice Networking**




## What software does Voice Networking require?

To install Voice Networking in a Passport network, you require Voice Networking software on all Passport nodes that directly connect to PBXs. Voice Networking also requires the following software systems to be in operation:

- path-oriented routing system (PORS). The following guides contain information about PORS:
  - 241-7401-750 *Passport 7400 Voice Transport Guide*
  - 241-5701-400 *Passport 7400, 15000, 20000 Networking Overview*
  - 241-5701-435 *Passport 7400, 15000, 20000 Path-Oriented Routing System Guide*
- network clock synchronization (NCS). NCS is described in 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide*.

- Dynamic Packet Routing System (DPRS). You require DPRS only if you employ data network address (DNA) based routing. See “Routing fundamentals” (page 133) for more details. The following guides contain information about DPRS:
  - 241-5701-405 *Passport 7400, 15000, 20000 Call Server Guide*
  - 241-5701-425 *Passport 7400, 15000, 20000 Dynamic Packet Routing System Guide*
  - 241-7401-110 *Passport 7400, DPN-100 Interworking Guide*

## What hardware does Voice Networking require?

The following FPs provide the hardware interface between a PBX and a Passport node running Voice Networking software:

- DS1 MVP-E (1pDS1Mvpe)
- E1 MVP-E (1pE1Mvpe)
- 4-port DS1 MVP-E (4pDS1Mvpe)
- 4-port E1 MVP-E (4pE1Mvpe)
- TTC2M MVP-E (1pTtc2mMvpe)

An E1 port supports up to 31 channels. A DS1 port supports up to 24 channels.

For a description of the hardware elements of these FPs, see 241-7401-200 *Passport 7400 Hardware Description*.

## How does Voice Networking communicate with PBXs?

Signaling protocols allow Passport nodes that operate the Voice Networking service to understand and exchange signaling information with PBXs. Signaling information includes basic call control messages and both call and non-call associated supplementary services. In essence, the Passport node operates as a tandem PBX, transporting signaling information and calls across the network.

You define the signaling protocol to use by configuring the Software and SignallingChannel components on all Passport nodes connected to PBXs. The signaling protocol you define must match the signaling protocol used by the connected PBX.

Voice Networking supports a number of signaling protocols. See “Signaling protocols fundamentals” (page 113) for more information. For compliance statement information, see “Compliance with standards—Voice Networking signaling protocols” (page 181).

## How does Voice Networking handle audio data?

The Voice Networking service supports a number of audio handling capabilities for voice, modem, and facsimile traffic. These capabilities allow you to

- define quality of service parameters based on customer needs
- maximize bandwidth savings by using compression and both silence and idle period suppression techniques on both voice and voice-band data traffic
- provide toll-quality voice signals by using echo cancellation, comfort noise generation, and congestion handling techniques
- engineer the flow of audio data to meet network requirements

You define Voice Networking audio handling capabilities by configuring voice routes and voice profiles. To configure the voice route and voice profile, you provision the attributes under the VoiceRoute Interface and VoiceNetworkingCallServer VoiceProfile components. To configure audio handling capabilities, see “Voice Networking configuration” (page 23).

## How does Voice Networking route calls?

There are three types of routing in Voice Networking.

- Routing can be based on PORS, where calls are placed over PORS SVCs using the VNCS for address resolution.
- Routing can be based on DNA and DPRS, where calls are placed over PORS SVCs using DNAs and DPRS for address resolution to a voice route.

- Routing can be based on DNA and DPRS, in which calls are placed over PORS SVCs using DNA and DPRS for address resolution to a hunt group server. The hunt group server then directs the call to a voice route that is defined as a hunt group member. This feature is sometimes known as Voice Networking call redirection.

Passport can also apply digit manipulation, in which the digits of the called number can be changed, either on ingress or egress.

For a description of these routing features, see “Routing fundamentals” (page 133).

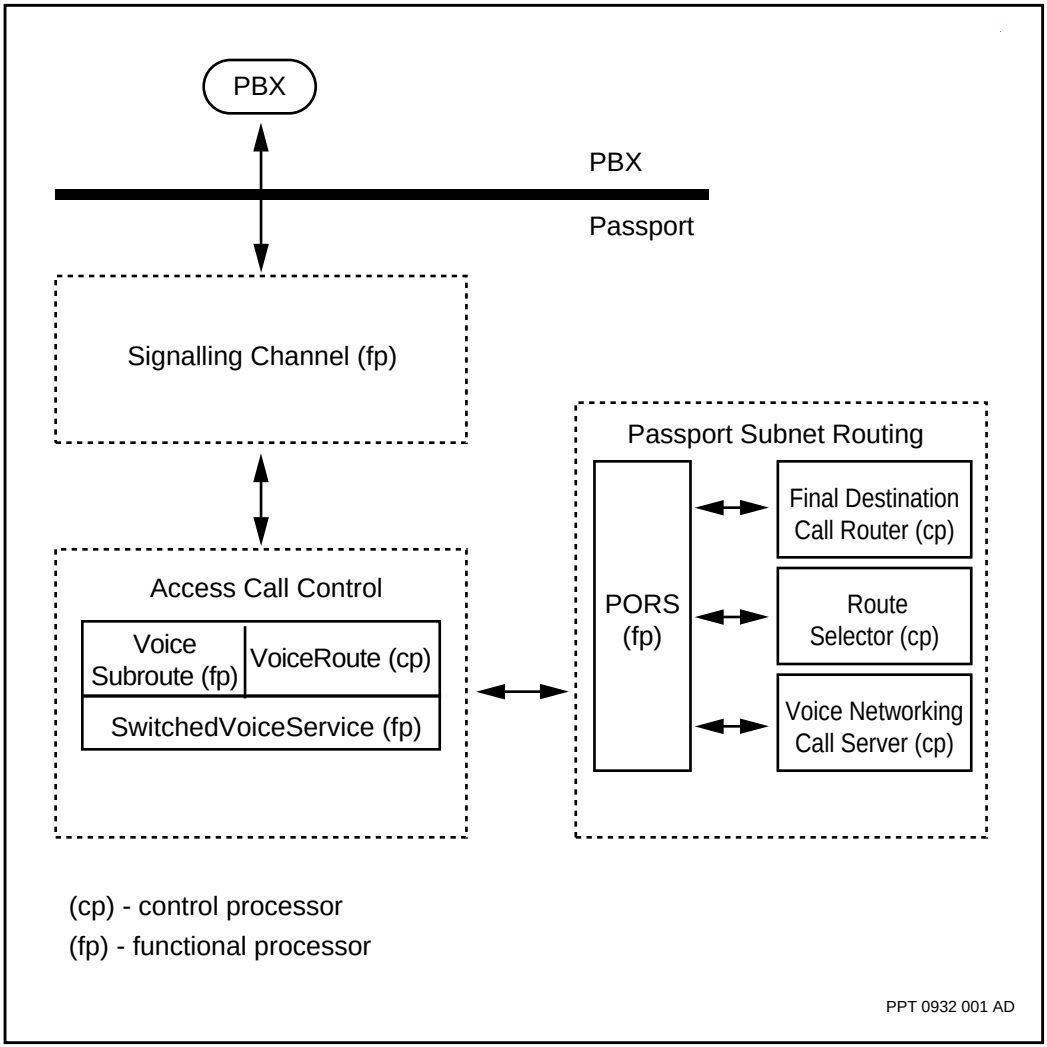
## What is the relationship between Voice Networking software components and systems?

The following groups of components and systems process Voice Networking calls:

- “Access protocol control” (page 103)
- “Access call control” (page 103)
- “Passport subnet routing” (page 104)

The figure “Relationship between Voice Networking components and systems” (page 100) describes the relationship between the three groups of components and systems and indicates whether the component or system resides on the control or function processor.

**Figure 25**
**Relationship between Voice Networking components and systems**

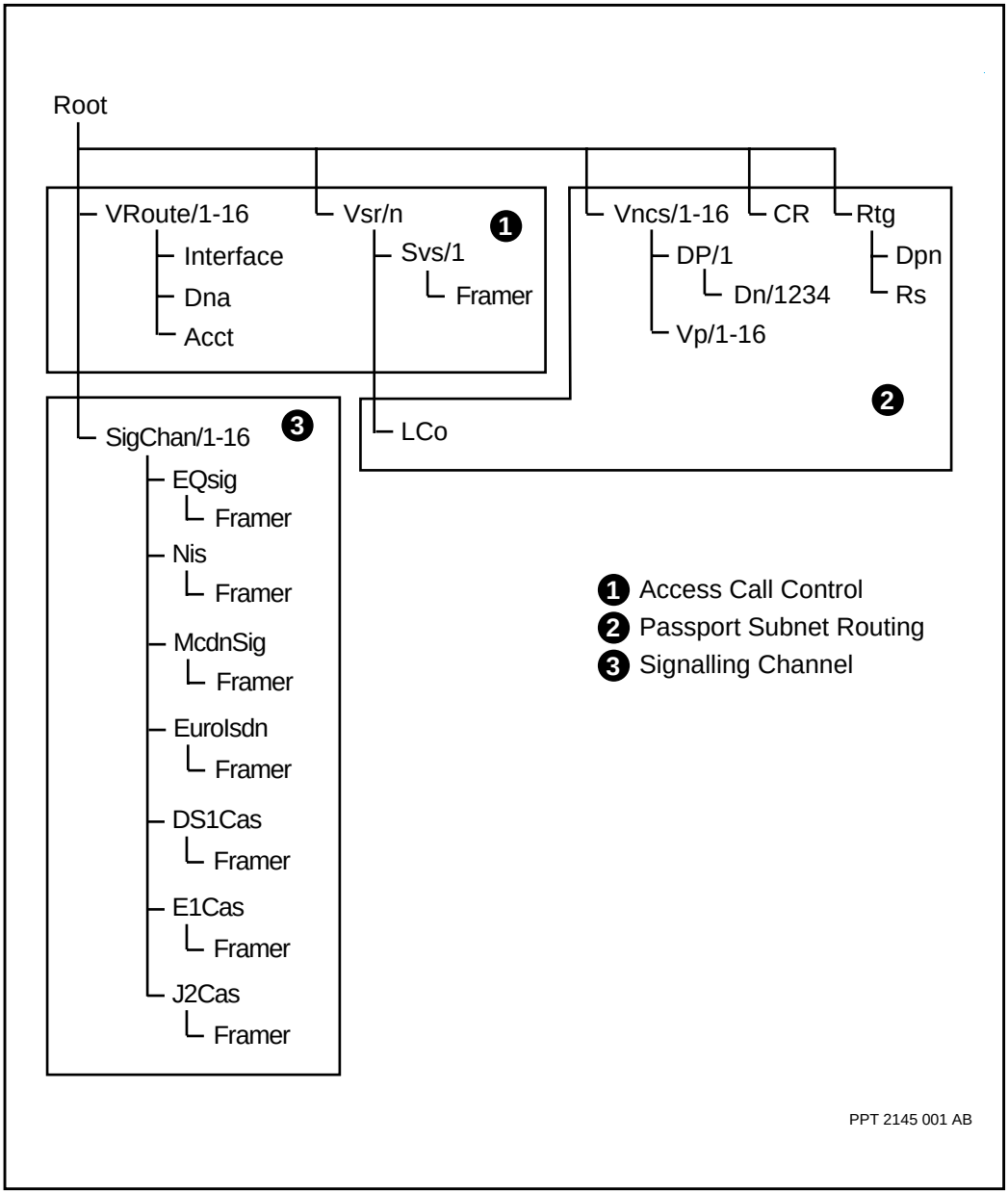


## Component relationships

The component hierarchy in the figure “Hierarchy of Voice Networking and related provisionable components” (page 102) provides a high-level view of the relationship between Voice Networking and related components. Observe the following rules when configuring Voice Networking components:

- There can be up to 16 instances of the VoiceNetworkingCallServer component. Each VoiceNetworkingCallServer component instance represents one numbering plan.
- There can be multiple VoiceRoute components per Passport node.
- A VoiceRoute component can have up to 16 VoiceSubroute components, but a VoiceSubroute component belongs to only one VoiceRoute component.
- A DataNetworkAddress component under the VoiceRoute component can be defined as a hunt group member through the HuntGroupMember and HuntGroupAddress components.
- A VoiceSubroute component belongs to a DS1 or E1 link (that is, it controls a collection of 64 kbit/s channels on a DS1 or E1 link).
- A SignallingChannel component can have multiple VoiceRoute and VoiceSubroute components.
- A SignallingChannel component cannot span multiple DS1 or E1 links.
- A VoiceRoute component can have multiple SignallingChannel components.
- A DS1 or E1 link can have multiple VoiceSubroute components, hence multiple VoiceRoute components.
- A SignallingChannel component can control up to four VoiceSubroute components.

**Figure 26**
**Hierarchy of Voice Networking and related provisionable components**



## Access protocol control

The SignallingChannel component comprises access protocol control. The SignallingChannel component defines the signaling characteristics of the connection to the PBX (the external equipment). Voice Networking supports standards-based signaling protocols for both CCS and CAS signaling formats. The SignallingChannel component responds to signals received from a PBX and communicates with the VoiceSubroute component to initiate call setup and tear-down processes. For details on the signaling protocols supported by Voice Networking, see “Signaling protocols fundamentals” (page 113).

## Access call control

The VoiceRoute, VoiceSubroute, and SwitchedVoiceService components comprise access call control. These 3 components control the progress and state of a call. The table “Access call control components” (page 104) describes the VoiceRoute, VoiceSubroute, and SwitchedVoiceService components and their functions.

**Table 1**
**Access call control components**

| Component | Description |
|---|---|
| VoiceRoute | A collection of channels (or trunks) connecting a Passport node to a destination PBX. A VoiceRoute component is involved in a call during the setup phase at the ingress and egress Passport nodes. A VoiceRoute component identifies a VoiceNetworkingCallServer component instance on the ingress node for address resolution purposes and forwards a call from the Passport subnet to one of its VoiceSubroute components. The VoiceRoute component performs the same function as a route on a PBX. |
| VoiceSubroute | The VoiceSubroute component belongs to a VoiceRoute component, but is associated with a physical DS1 or E1 link. The VoiceSubroute component is involved in a call at both the ingress and egress Passport nodes. A VoiceSubroute component cannot span multiple DS1 or E1 links. |
| SwitchedVoiceService | The SwitchedVoiceService component belongs to a VoiceSubroute component. The SwitchedVoiceService component is created dynamically and associated with a channel (or timeslot) on the DS1 or E1 link. The SwitchedVoiceService component controls the audio call. |
| | |

## Passport subnet routing

The PORS and VoiceNetworkingCallServer, RouteSelector, and FinalDestinationCallRouter components comprise Passport subnet routing (see the table “Passport subnet routing system and components” (page 105) for details). Passport subnet routing elements interact with the VoiceRoute, VoiceSubroute, and SwitchedVoiceService components to find and establish a path across the Passport subnet.

**Table 2**
**Passport subnet routing system and components**

| System/component | Description |
|---|---|
| PORS | PORS manages the connection in the Passport subnet. |
| VoiceNetworkingCall-Server | The VoiceNetworkingCallServer component contains the numbering tree that identifies the far-end Passport node. The VoiceNetworkingCallServer component also contains two sub-components, VoiceProfile and DiallingPlan, that specify quality of service and addressing parameters, respectively. |
| RouteSelector (RS) | The RouteSelector component finds a path across the Passport subnet. |
| FinalDestinationCallRouter (FDCR) | The FinalDestinationCallRouter identifies the VoiceRoute at the far-end Passport node. |
| | |

# What configuration parameters impact Voice Networking?

The following sections describe parameters that impact how you configure the Voice Networking service:

- “Installation” (page 105)
- “Operational” (page 106)
- “Logical and function processors” (page 106)
- “Interworking and gateways” (page 108)

Ensure that you understand and meet the requirements listed in the following sections before you install and configure the Voice Networking service.

## Installation

When installing Voice Networking, consider the following:

- Voice Networking cannot co-exist on an MVP-E FP with Voice Transport or the bit transparent data service (BTDS).

- The TTC2M MVP-E FP only supports channel associated signaling (CAS) and cannot be used for common channel signaling (CCS) as it does not support the high-level data link control (HDLC) protocol.
- All four ports of the 4-port MVP-E must use the same signalling protocol, which can be one of NIS, MCDN, CAS, or ETSI QSIG
- Voice Networking does not support interworking between 4-port MVP-E FPs and Voice FPs
- The 4-port MVP-E FP cannot interwork with the Passport 4400 series
- Interworking between the 4-port MVP-E FPs running PCR 4.2 or higher and the 1-port MVP-E FPs running Release 7.0 is supported.

## Operational

The Voice Networking service operates under the following conditions:

- Numbers which enter the subnet must be unique to allow routing.
- A dialed number can access multiple instances of the VoiceRoute component. All links associated with a particular VoiceRoute component instance must employ either overlap or enbloc dialing. Voice Networking does support connections involving overlap to enbloc dialing.
- Voice Networking does not support non-facility associated signaling (NFAS) or multiple links per D-channel (nB+D).
- The PBX must provide answer supervision.
- A tandem PBX cannot perform any digit manipulation on the digits visible to the Passport subnet. The dialed digits into the PBX must equal the dialed digits out of the PBX.
- Supported modem standards include ITU-T V.34 and earlier.

## Logical and function processors

You define logical processors (LP) and link them to the configured Voice Networking software and specify the type of port and the number of channels for each LP. For each function processor (FP), you define its type and specify its position on the Passport shelf. You then link the LPs to the FPs. Each LP and FP combination represents one physical interface to the PBX.

Voice Networking supports up to 14 LogicalProcessor components (Lp/1 up to Lp/14) and up to 14 LPs. You must reserve Lp/0 for the control processor (CP), which you link to the Shelf Card/0 component instance. You can link Lp/15 to a spare CP or, for example, a trunking FP.

The table "FP-to-capability relationships" (page 107) describes Voice Networking capability-to-card relationships and restrictions.

**Table 3**
**FP-to-capability relationships**

| | **card type** | | | | |
|---|---|---|---|---|---|
| **Capability** | **DS1 MVP-E** | **E1 MVP-E** | **DS1 MVP-E (4-port) (see Note 4:** | **E1 MVP-E (4-port) (see Note 4:** | **TTC2M MVP-E** |
| G.728 (LD-CELP) 16 kbit/s voice compression (see Note 1) | X | X | X | X | X |
| G.729/G.729A (CS-ACELP) 8 kbit/s voice compression (see Note 1:) | X | X | X | X | X |
| Fax relay (see Note 1:) | X | X | X (see Note 2:) | X (see Note 2:) | X |
| G.726 (ADPCM) voice and modem/fax encoding | X | X | X | X | X |
| Dynamic up- and down-speeding (G.711/ G.726) of audio traffic (see Note 4:) | X | X | X | X | X |
| Silence suppression, with provisionable support for capping comfort noise | X | X | X | X | X |
| Fax idle suppression | X | X | X | X | X |
| Provisionable speech and fax hangover time | X | X | X | X | X |
| Tandem pass through (see Note 1:) | X | X | X | X | X |
| Configurable egress buffer | X | X | X | X | X |

(Sheet 1 of 2)

**Table 3 (continued)**
**FP-to-capability relationships**

| | card type | | | | |
|---|---|---|---|---|---|
| **Capability** | **DS1 MVP-E** | **E1 MVP-E** | **DS1 MVP-E (4-port) (see Note 4:** | **E1 MVP-E (4-port) (see Note 4:** | **TTC2M MVP-E** |
| Facsimile/speech discrimination | X | X | X | X | X |
| DTMF tone detection and regeneration | X | X | X | X | X |
| ETSI QSIG segmentation | | X | | X | |
| Echo cancellation | X | X | X | X | X |
| CCS-to-CAS protocol gateway support (see Note 3:) | X | X | X | X | |
| Hunt Group Server-based routing | X | X | X | X | X |

***Note 1:*** To use G.728 and G.729/729A voice encoding, fax relay, and tandem pass through on an MVP-E FP, you must add g728, g729, faxRelay, and tandemPassThrough, respectively, to the featureList attribute under the Software LogicalProcessorType component, along with the voiceNetworking attribute (G.729A voice encoding is only supported on MVP-E FPs). Refer to the configuration procedures in “Voice Networking configuration” (page 23) for more details.

***Note 2:*** Since 4-port MVP-E FPs support fax relay V.17 fax calls, you must provision the v17EncodedAsG711G726 attribute with the value no.

***Note 3:*** CCS-to-CAS protocol gateway support is automatically included when you provision the values MCDN, etsiQsig, nisSig, or casSig for the featureList attribute under the Software LogicalProcessorType component. For a description of the CCS-to-CAS signaling gateways, see “Protocol interworking and gateways” (page 125).

***Note 4:*** Requires PCR 3.0 software.

(Sheet 2 of 2)

## Interworking and gateways

Interworking refers to the interworking of signaling protocols and to connections between MVP-E FPs. For more information on interworking and gateways, see “Protocol parameters” (page 109) for more information.

### Protocol parameters

See the following tables for information about Voice Networking protocol parameters:

- The table “Card types and supported Voice Networking protocols” (page 109) maps Voice Networking function processors (FPs) to the protocols that they support
- The table “Voice Networking protocol interworking” (page 110) describes the protocol interworking supported by Voice Networking. This table only captures interworking scenarios and does not reflect that, by default, native or like protocols communicate with each other (see gray cells in the table).
- The table “Voice Networking CCS-to-CAS protocol gateways” (page 110) describes the CCS-to-CAS protocol gateways supported by Voice Networking. See “Protocol interworking and gateways” (page 125) for details about the functionality included with protocol interworking and gateways.

**Table 4**
**Card types and supported Voice Networking protocols**

| | **Voice networking protocols** | | | | | | |
|---|---|---|---|---|---|---|---|
| **Card type** | **ETSI QSIG** | **NIS A211-1** | **MCDN** | **Euro-ISDN** | **DS1 CAS** | **E1 CAS** | **J2 CAS** |
| 1-port DS1 MVP-E | X | X | X | | X | | |
| 4-port DS1 MVP-E | X | X | X | | X | | |
| 1-port E1 MVP-E | X | | X | X | | X | |
| 4-port E1 MVP-E | X | | X | X | | X | |
| TTC2M MVP-E | | | | | | | X |
| | | | | | | | |

**Table 5**
**Voice Networking protocol interworking**

| | **Protocols** | | | | | |
|---|---|---|---|---|---|---|
| **Protocols** | **E1 CAS** | **DS1 CAS** | **J2 CAS** | **NIS** | **MCDN** | **Euro-ISDN** |
| E1 CAS | | X | X | | | |
| DS1 CAS | X | | X | | | |
| J2 CAS | X | X | | | | |
| NIS | | | | | X | |
| MCDN | | | | X | | |
| Euro-ISDN | | | | | | X (see Note 1:) |

***Note 1:*** Euro-ISDN protocol interworking can occur between the generic ETSI ISDN variant (PBX, or network side) and the Austrian or German variant (PSTN, or user side).

***Note 2:*** ETSI QSIG does not interwork with other protocols.

**Table 6**
**Voice Networking CCS-to-CAS protocol gateways**

| | **CAS protocols** | |
|---|---|---|
| **CCS protocols** | **E1 CAS** | **DS1 CAS** |
| NIS A211-1 (on 1-port and 4-port DS1 MVP-E only) | X | X |
| ETSI QSIG (on 1-port and 4-port DS1 and E1 MVP-E only) | X | X |
| MCDN | X | X |

## Voice Networking accounting

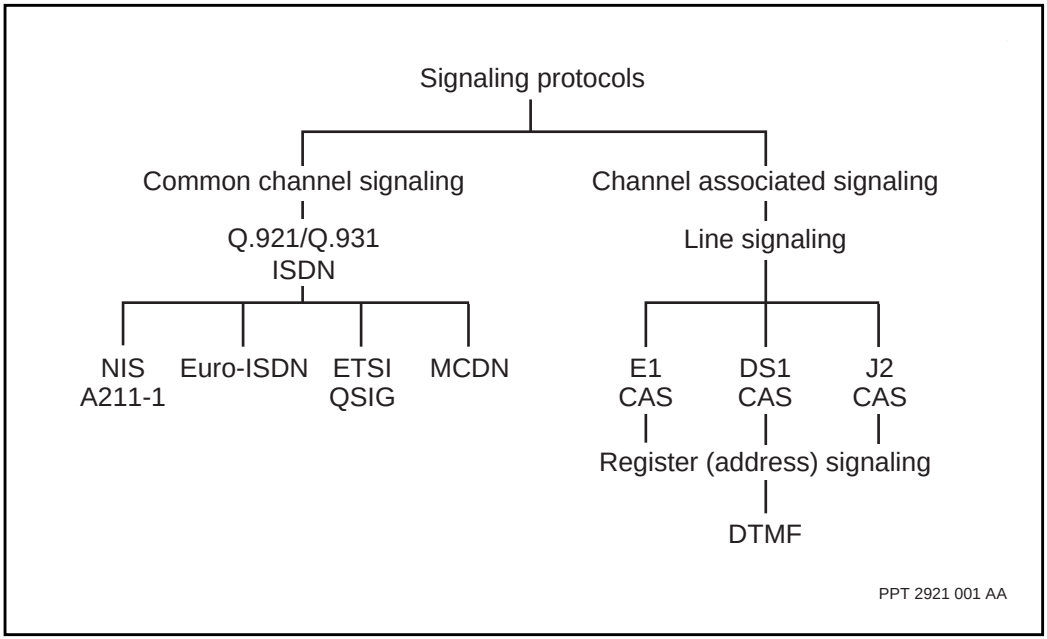
The VoiceAccounting component controls accounting for calls processed by its parent component, the VoiceRoute. Accounting is turned off by default. Any configuration changes made to the default values of the VoiceAccounting component's attributes will take effect on calls that are set up after the changes are committed. Accounting records will not be generated for calls that are already in progress before accounting is turned on. If accounting is turned off and turned on again, the VoiceAccounting component only generates accounting records for those calls set up by the VoiceRoute component after accounting is turned on again.

# Chapter 8
# Signaling protocols fundamentals

See the following sections for information on Voice Networking and signaling protocols:


- “Common channel signaling (CCS)” (page 114)
- “Channel associated signaling (CAS)” (page 123)
- “Protocol interworking and gateways” (page 125)
- “Individual channel busy-back” (page 128)
- “Signaling protocols components” (page 129)
- “Internal cause values” (page 130)

**Figure 27**
**Signaling protocols supported by Voice Networking**




## Common channel signaling (CCS)

The CCS protocols supported by Voice Networking are based on ITU-T Q.921 and Q.931 integrated services digital network (ISDN) signaling standards. Voice Networking CCS protocols include both private-to-private and private-to-public network protocols. See the following sections for information about the CCS protocols supported by Voice Networking:

- “Network interface specification (NIS) A211-1” (page 115)
- “European Telecommunications Standards Institute Q interface signaling (ETSI QSIG)” (page 116)
- “Meridian customer defined networking (MCDN)” (page 118)
- “European Digital Subscriber Signaling System Number One (EDSS1) or Euro-ISDN” (page 121)

CCS protocols use one timeslot or channel—called the delta or D-channel—of a given digital link to carry the signaling information for the calls on the remaining channels—called the bearer or B-channels. For DS1 links using CCS protocols, timeslot 24 carries signaling information. For E1 links using CCS protocols, timeslot 16 carries signaling information. Primary rate interface (PRI) is the name given to the DS1 or E1 digital link between the Passport node and PBX (see the figure “Typical Passport to PBX protocol connection” (page 126)).

In general, CCS protocols support a number of supplementary services, such as name and number display, call forwarding, and voice mail. Supplementary services, as defined in the specific CCS protocol standard, pass transparently (that is, without modification) through a Passport network. For example, MCDN supports calling party name display (CPND). Meridian 1 PBXs use the MCDN signaling protocol. With Voice Networking support of MCDN, Passport nodes connected to Meridian 1 PBXs transport CPND information across the Passport subnet to end-users.

For information on the cause codes Voice Networking supports for each CCS protocol, see “Call release cause codes” (page 219).

## Network interface specification (NIS) A211-1

Voice Networking’s use of the NIS A211-1 signaling protocol is a private application of a public networking protocol. Voice Networking support of NIS A211-1 provides user-to-user interface signaling (that is, between PBXs) over a Passport network.

NIS A211-1 supports basic call and supplementary services as shown in the table “Supported NIS A211-1 basic call and supplementary services” (page 116). For additional information on NIS A211-1 compliance, see “Compliance statement for NIS A211-1” (page 182).

**Table 7**
**Supported NIS A211-1 basic call and supplementary services**

| Category | Details |
|---|---|
| Basic call services | • Call control— setup and tear-down of 64 kbit/s bearer services, including audio (speech, modem, and facsimile) and data calls<br>• 56 kbits/s clear data call transport<br>• Call link and channel maintenance<br>• Called number and calling number transport<br>• Special number services<br>• Federal Communications Commission (FCC) equal access information information transport as a tandem node<br>• Integrated services access information transport as a tandem node |
| Supplementary services | • Calling and connected party number display— presentation and restriction<br>• Network name display— presentation and restriction<br>• Network redirection and reason |
| | |

## European Telecommunications Standards Institute Q interface signaling (ETSI QSIG)

ETSI QSIG is a private networking protocol. Voice Networking's application of the ETSI QSIG signaling protocol provides user-to-user interface signaling (that is, between PBXs) over a Passport network. Voice Networking also supports the handling of large ISDN messages produced by particular PBXs by using a segmentation and reassembly technique. As shown in the table "Supported ETSI QSIG basic call and supplementary services" (page 117), ETSI QSIG supports basic call and supplementary services such as call transfer and name display. For additional ETSI QSIG compliance information, see "Compliance statement for ETSI QSIG" (page 181).

**Table 8**
**Supported ETSI QSIG basic call and supplementary services**

| Category | Details |
|---|---|
| Basic call services | • Call control— setup and tear-down of 64 kbit/s bearer services, including audio (speech, modem, and facsimile) and data calls<br>• Call related application protocol data unit (APDU) transport<br>• Call related notification transport<br>• Connection oriented implementation of call independent APDU transport<br>• Segmentation and reassembly |
| Supplementary services | • Name identification<br>• Transit counter<br>• Call transfer<br>• Call diversion<br>• Path replacement<br>• Call offer<br>• Call completion<br>• Called number and busy number identification<br>• Message waiting indication<br>• Centralized voice mail |
| | |

Only E1 MVP-E FPs support the segmentation and reassembly capability.

Depending on the PBX and the country of origin (for example, Germany), ETSI QSIG ISDN messages can vary in size. The standard-sized ETSI QSIG ISDN message is up to 260 bytes. Large-sized ETSI QSIG ISDN messages are from 261 up to 2013 bytes plus overhead. By setting the msgSegmentation attribute to enabled, you ensure that Voice Networking does not discard the optional message content related to supplementary services that is contained in large-sized messages.

To successfully process large-sized ETSI QSIG ISDN messages, the msgSegmentation attribute must be set to enabled at both ends of a Voice Networking connection.

When you set the e1ChannelNumbers attribute to contiguous, Voice Networking maps channels 1 to 15 to timeslots 1 to 15 and channels 16 to 30 to timeslots 17 to 31. The default value skip16 means that the Passport node expects E1 bearer channels and timeslots—1 to 15 and 17 to 31—to directly align.

## Meridian customer defined networking (MCDN)

MCDN is a proprietary, private networking protocol developed by Nortel Networks for the Meridian 1 PBX. Voice Networking supports the Peer-to-Peer MCDN variant. Voice Networking support of the MCDN Peer-to-Peer signaling protocol provides user-to-user (Meridian 1-to-Meridian 1 PBX) interface signaling over a Passport network. Passport nodes provide tandem node call processing for MCDN basic call and supplementary services. the table “Supported MCDN basic call and supplementary services” (page 119) contains information about the MCDN basic call and supplementary services that Voice Networking support including network attendant services (NAS), network messaging services (voice mail), automatic call distribution (ACD), and anti-tromboning. For additional MCDN compliance information, see “Compliance statement for MCDN” (page 183).

Some of the capabilities and services listed in the table “Supported MCDN basic call and supplementary services” (page 119) can be enabled and disabled through configuration. For more information about configuring MCDN capabilities and services, see “Configuring the MCDN signaling protocol” (page 52).

**Table 9**
**Supported MCDN basic call and supplementary services**

| Category | Details |
|---|---|
| Basic call services | • Call control and link and channel maintenance messaging<br>• Call setup and tear-down for audio traffic (voice, modem, and facsimile); 64 kbit/s (voice and unrestricted) and 56 kbit/s data bearer services<br>• Channel negotiation<br>• Overlap dialing<br>• Called and calling party number transport<br>• Federal Communications Commission (FCC) equal access for transporting the carrier access code (CAC) with the called party number<br>• Flexible numbering plan |

(Sheet 1 of 2)

**Table 9 (continued)**
**Supported MCDN basic call and supplementary services**

| Category | Details |
|---|---|
| Connection-oriented supplementary services | • Attendant and network attendant services (NAS), including break-in, schedule, night service, timed reminder, and call extension<br>• Anti-tromboning (both trunk and NAS); electronic switched network (ESN) and related services, including basic and network alternate route selection (BARS and NARS), coordinated dialing plan, time of day and automatic least cost routing; network and travelling class of service (NCOS and TCOS); off-hook and remote virtual queuing<br>• Call forward, page, park, hunt (internal and external), pickup (network wide), redirection, transfer, and trace (including malicious); drop back busy<br>• Number and name display with privacy options<br>• Network authorization codes (transmitted as DTMF digits)<br>• Trunk route optimization<br>• Access to intercom and radio paging systems |
| Connectionless supplementary services | • Network messaging services (voice mail) and related message waiting signaling, call pickup queries (world wide), and electronic lock<br>• Automatic call distribution (ACD) between pools of agents<br>• Remote call forward<br>• Ring again (busy and no reply)<br>• Network time synchronization<br>• Multisite relocation of wireless phones |
| (Sheet 2 of 2) | |

### MCDN signaling protocol configuration

To enable or disable the call pickup feature, both the connection-oriented *callPickupNetworkWide* (cpnw) attribute and the connectionless *callPickupWorldWide* (cpww) attribute must be set to enabled or disabled. By default, both attributes are set to enabled under the SignallingChannel McdnSig component.

The *privateNetworkIdentifier* attribute under the SignallingChannel McdnSig component is not used in the configuration of the MCDN signalling protocol.

The following information pertains to MCDN anti-tromboning:

- The value antiTromboning includes both NAS and trunk anti-tromboning.
- For anti-tromboning requests involving a dual egress call scenario (that is, a tandem PBX was used to dial out to the destination and out to the user before dropping out), Voice Networking bills the tandem node for accounting purposes.
- To prevent call failures when anti-tromboning is enabled, (1) each configured directory number (DN) must point to the same destination, and (2) each node in the network must have its own DNs configured in the voice networking call server's database. See "Configuring the Voice Networking call server" (page 61) for more information.

## European Digital Subscriber Signaling System Number One (EDSS1) or Euro-ISDN

Euro-ISDN is a common implementation of ISDN signaling standards that includes the different ISDN applications used in separate European countries. Euro-ISDN is defined in ETSI specifications and based on ITU-T recommendations. Voice Networking supports three Euro-ISDN variants: generic (non-country specific or ETSI ISDN), Austrian, and German. Voice Networking transports a number of Euro-ISDN basic call and supplementary services (see the table "Supported Euro-ISDN basic call and supplementary services" (page 122)).

Voice Networking support of Euro-ISDN provides a Passport network with the following two types of interface signaling:

- user-to-network—central office (CO) in a national public switched telephone network (PSTN)-to-private network PBX. Voice Networking's application of Euro-ISDN guarantees the transport of supplementary services between a PBX and a CO. The PBX and CO have, respectively, a service subscriber-to-service provider relationship.
- user-to-user—basic call services between PBXs only, without the guaranteed transport of supplementary services. In a Euro-ISDN network configuration that requires the transport of supplementary services, you can configure, for example, an ETSI QSIG signaling channel connection.

For Euro-ISDN compliance with standards information, see “Compliance statement for Euro-ISDN” (page 184).

**Table 10**
**Supported Euro-ISDN basic call and supplementary services**

| Category | Details |
|---|---|
| Basic call services | • Call control—setup and tear-down of 64 kbit/s bearer services, including audio (speech, modem, and facsimile) and data calls<br>• Channel negotiation<br>• Overlap and enbloc dialing and conversion<br>• Flexible numbering plan |
| Bearer-related supplementary services | • Closed user group (CUG)<br>• Calling line identification presentation and restriction (CLIP and CLIR) and connected line identification and restriction (COLP and COLR)<br>• Direct dial in (DDI)<br>• Subaddressing (SUB), for both called and calling parties<br>• User-to-user signaling (UUS)<br>• Malicious call identification (MCID)<br>• Call diversion services (with partial rerouting), including call forwarding unconditional (CFU), call forwarding busy (CFB), call forwarding no reply (CFNR), and call deflection (CD)<br>• Advice of charge (AOC) for during (AOC-D) and at the end of a call (AOC-E)<br>• Conference call (CONF) |
| Bearer-independent, connection-oriented supplementary services | • Completion of call to busy subscriber (CCSB)<br>• Completion of call on no response (CCNR)<br>• Freephone (FPH)<br>• Message waiting indicator (MWI) |
| | |

### Euro-ISDN signaling protocol configuration

When configuring the Euro-ISDN signaling protocol, you must define the master/slave relationship parameters for channel negotiation according to the protocol variant. The protocol variant corresponds to the type of equipment that the Passport node connects to—either private network PBX (network side) or public switched telephone network CO (user side).

Virtual calls provide bearer-independent, connection-oriented transport of Euro-ISDN supplementary services, such as completion of call on no response (CCNR).

## Channel associated signaling (CAS)

Voice Networking's application of CAS depends on the signaling formats supported by each of the following interfaces:

- "DS1 CAS" (page 124)
- "E1 CAS" (page 125)
- "J2 CAS" (page 125)

CAS is a method of carrying signaling information for all of the voice timeslots of a digital connection. CAS methods vary according to the signaling formats supported by the interface: DS1, E1, or TTC. The interface defines if signaling information is carried in each traffic-carrying timeslot or in a dedicated signaling channel. CAS includes both line and register signaling. Line signaling includes the state change of distinct signaling bits—A, AB, or ABCD. An example of a state change is going from idle to seized. The actual number of signaling bits used (1, 2, or 4) depends on the interface. Register or address signaling always follows line signaling. Voice Networking only supports dual-tone multifrequency (DTMF) register signaling for digit transmission, and does not support pulsed signaling. Both ends of a call must support DTMF signaling. For more information, see "CAS interconnection guidelines" (page 123).

### CAS interconnection guidelines

Voice Networking can support proprietary or standards-based CAS variants. The CAS variant must use DTMF for digit transmission and steady state signaling. The state changes supported by Voice Networking include idle, seize, connect (answer), wink, seize acknowledgment, clear back, and clear

forward. The state changes match the appropriate signaling bit combination (either A, AB, or ABCD). You match the state change indications generated by the connected equipment by configuring the attributes under the AbcdProv and TimerProv groups. You find the AbcdProv and TimerProv groups of attributes under the Ds1Cas, E1Cas, and J2Cas components. The attributes in these two groups allow Voice Networking to support different combinations of signaling bits according to the equipment or signaling type used.

For example, Voice Networking can support analog signaling types originating from the user side of the network. Examples of analog signaling types are 2- or 4-wire ear-and-mouth (E & M) terminal interface equipment (TIE) trunk signaling. The state changes defined in E & M TIE trunk 2-state signaling are like those used over digital CAS trunks.

For call setup signaling sequences, Passport DS1 CAS, E1 CAS, and J2 CAS support both immediate and wink start. E1 CAS also supports seize acknowledgment start. For call disconnect signaling sequences, E1 CAS has clear back defined by default.

Under the J2Cas component, the default value P for the attributes winkTx and winkRx refers to the toggling of the A signaling bit to support wink starts. You define the duration of this pulsed signal by configuring the winkTime attribute. E1 CAS and DS1 CAS also support wink starts, although the default value for the winkTx and winkRx attributes is unused.

## DS1 CAS

Passport DS1 CAS line signaling supports the following two types of DS1 interface framing formats:

- superframe format (either SF or D4)
- extended superframe format (ESF)

DS1 interfaces use robbed bit signaling. The least significant bit is robbed from each traffic timeslot, every six frames. The least significant bits carry signaling information, not traffic. In SF or D4 mode, one set of AB signaling bits is robbed for each superframe (a superframe has 12 frames). In ESF mode, four signaling bits—A,B,C and D—are robbed from each extended superframe (an extended superframe has 24 frames).

### E1 CAS

E1 CAS line signaling allocates timeslot 16 of an E1 multiframe to carry all the signaling information for the 30 remaining timeslots. E1 connections use all four signaling bits—A, B, C, and D—to represent all required line states.

### J2 CAS

Passport J2 CAS line signaling applies to TTC lines. J2 CAS uses timeslot 16 to carry signaling information and uses one signaling bit—A—to represent all required line states.

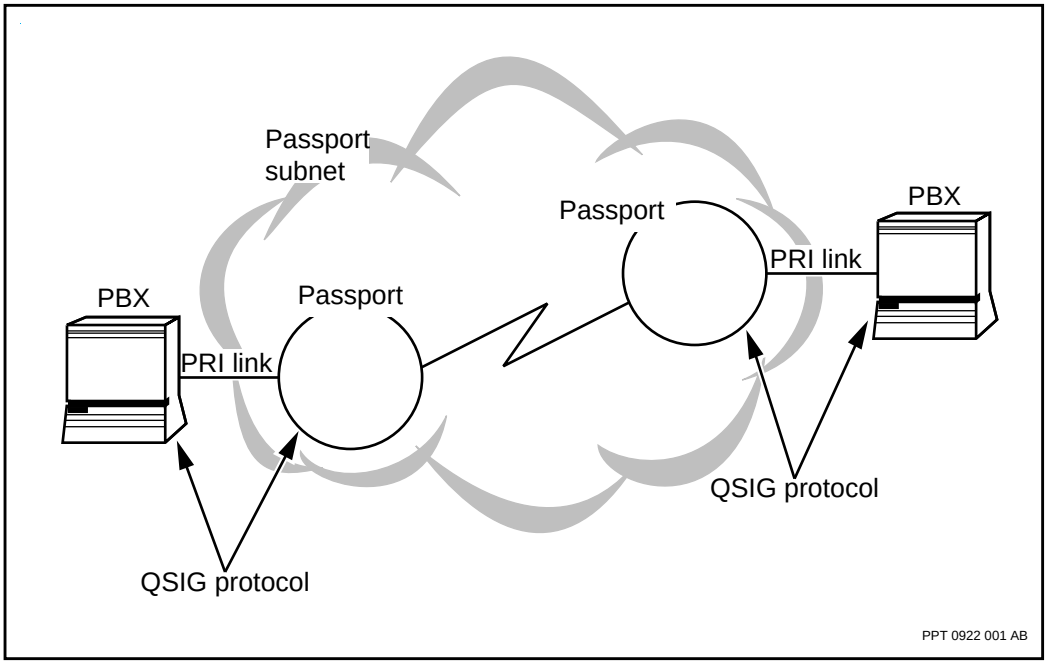
## Protocol interworking and gateways

Typically, the signaling protocol is identical on all Passport nodes that operate the Voice Networking service (see the figure "Typical Passport to PBX protocol connection" (page 126)). For example, there is no gateway function or interworking between the CCS protocols NIS A211-1 and ETSI QSIG. If a network of PBXs contains different CCS protocols that require protocol gateway processing, a tandem PBX and not the Passport node must provide a protocol gateway function.

Protocol interworking and gateways are needed when a call involves two different access protocols. For details about the protocol interworking and signaling gateways supported by Voice Networking, see the following sections:

- "Interworking" (page 126)
- "Gateways" (page 127)

**Figure 28**
**Typical Passport to PBX protocol connection**




## Interworking

With protocol interworking, no conversion takes place between the signaling protocols involved in a call. Voice Networking supports limited CCS-to-CCS interworking and full CAS-to-CAS interworking. The table “Protocol interworking scenarios” (page 127) contains details about the functionality included in each interworking scenario.

**Table 11**
**Protocol interworking scenarios**

| Interworking scenario | Details |
|---|---|
| NIS A211-1-to-MCDN | Functionality includes call setup and tear-down, progress indicators, and supplementary services such as calling and connected line identification presentation and restriction (CLIP and CLIR, respectively). |
| Euro-ISDN private, network-side variant (generic ETSI ISDN)-to-Euro-ISDN PSTN, user-side variant (austria or germany) | As stated in "European Digital Subscriber Signaling System Number One (EDSS1) or Euro-ISDN" (page 121), Voice Networking<br>• supports all Euro-ISDN basic call services<br>• guarantees the transport of Euro-ISDN supplementary services between a PBX (private network) and a CO (public network) |
| CAS variant-to-CAS variant | Interworking is supported between CAS protocols regardless of the signaling format and interface combination involved in a call (for example, a call between DS1 CAS and E1 CAS or J2 CAS). |
| | |

## Gateways

With a protocol gateway, some degree of conversion occurs between two different signaling types. During end-to-end negotiation, calls requiring gateway processing are either accepted or rejected, depending on whether the requested gateway is supported. The egress or destination Passport makes the determination to accept or reject a call requiring gateway processing.

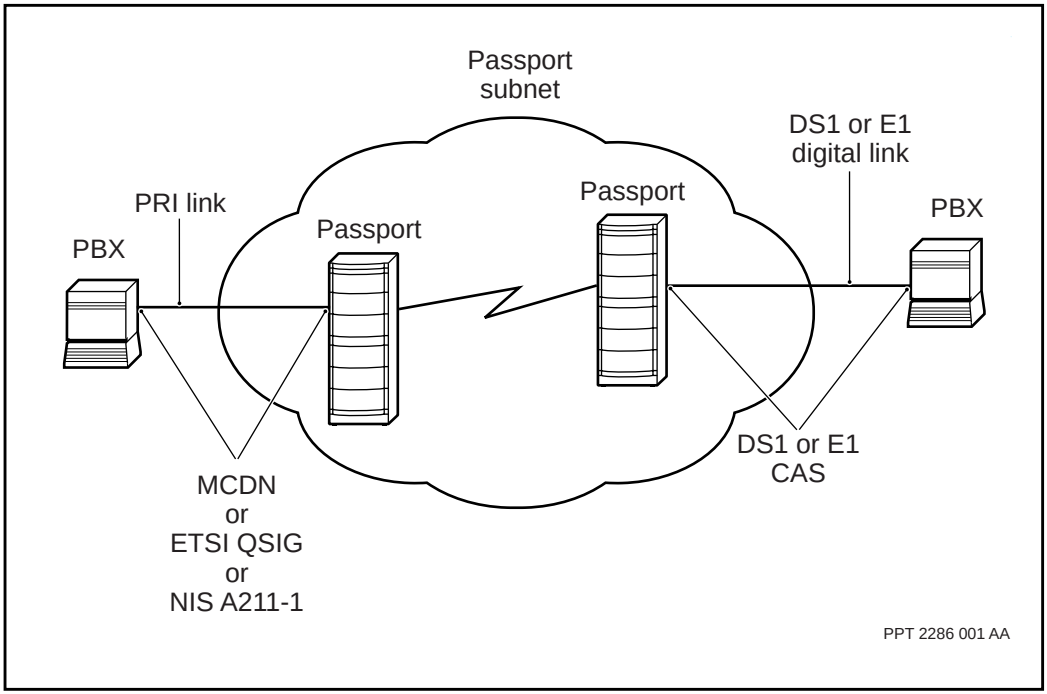
Voice Networking supports the following CCS-to-CAS protocol gateways:

- MCDN, ETSI QSIG or NIS A211-1 to DS1 CAS
- MCDN, ETSI QSIG or NIS A211-1 to E1 CAS

The CCS-to-CAS protocol gateways support both overlap and enbloc dialing, but only deliver basic call features (such as call setup, tear-down, and messaging for unsuccessful call handling). That is, the CCS-to-CAS protocol gateways do not deliver the supplementary call features supported by MCDN,

ETSI QSIG and NIS A211-1. The figure “Voice Networking protocol gateways” (page 128) describes the protocol gateways supported by Voice Networking.

**Figure 29**
**Voice Networking protocol gateways**




## Individual channel busy-back

When a routing problem occurs between a PBX and a Passport node, a yellow or blue alarm is raised on the link. Because Voice Networking supports intelligent signaling on the D-channel, the alarm does not cause the whole link to be taken out of service. Voice Networking allows for effective communication between a PBX and a Passport node, singling out individual channel problems and keeping the rest of the link in service.

# Signaling protocols components

The signaling protocol component you provision under the SignallingChannel component corresponds to the signaling protocol software you provision under the Software component and the signaling format—CCS or CAS—you specify under the E1 or Ds1 port component. You link the signaling protocol component to the LogicalProcessor, port (E1 or DS1) and Channel component combination responsible for carrying the particular signaling protocol's call control messages and information.

Consider the following information when configuring the SignallingChannel component in "Configuring the SignalingChannel component" (page 40).

- Voice Networking supports protocol interworking and protocol gateways.
- It is not necessary at this point to provision the subroutes attribute. When you add VoiceSubroute components, you can link them to a particular SignallingChannel component instance (see "Voice subroute" (page 178)).
- Interworking between different CAS interface signaling types occurs automatically when you configure the casSig protocol software feature on Passport nodes using different interfaces (for example, DS1 and E1)
- The SigChan instance numbers range from 1 to 153.
- EuroISDN to CAS Gateway is not supported for any of the Voice FPs.

The setting of the attributes forceNpiTon and defaultNpiTon impacts call setup for egress calls (that is, for calls leaving the Passport network) and is especially important for calls requiring protocol gateway processing (for example, E1 CAS to ETSI QSIG CCS calls).

The table "Relationship between defaultNpiTon attribute and PBX NPI/TON values" (page 130) describes how the values of defaultNpiTon map to NPI and TON information supplied by the calling PBX.

The values under the defaultNpiTon attribute correspond to the default values available under the VoiceRoute component's diallingPlan0 attribute. For more information on how Voice Networking interprets NPI and TON information supplied by the calling PBX, see "Interaction between configurable components and attributes during call routing" (page 138).

**Table 12**
**Relationship between defaultNpiTon attribute and PBX NPI/TON values**

| Egress Passport node | Calling PBX | |
|---|---|---|
| **defaultNpiTon attribute value** | **NPI** | **TON** |
| casUnknown | unknown | unknown |
| unknown | E.164 | unknown |
| international | | international |
| national | | national |
| subscriber | | subscriber |
| p0 | private | 0 |
| p1 | | 1 |
| p2 | | 2 |
| p3 | | 3 |
| p4 | | 4 |
| p5 | | 5 |
| p6 | | 6 |
| p7 | | 7 |
| | | |

## Internal cause values

The addition of the InternalCauseMap component does not impact Voice Networking's handling of external cause codes. The attributes under the InternalCauseMap component only apply to the NIS A211-1, ETSI QSIG, and Euro-ISDN protocols.

***Note:*** A semantic check error occurs if you attempt to add the InternalCauseMap component to a SignallingChannel component instance configured with a CAS protocol.

Each attribute under the InternalCauseMap component represents one or more call clearing events related to call establishment. For each attribute, the default value autoConfigure represents a different numerical cause value. Each attribute accepts any value between 0 and 127. The number you specify for a given attribute matches the cause value required by the originating PBX to initiate alternate rerouting of a call.

If you do not specify a value for these attributes, or if you do not add the InternalCauseMap component, Voice Networking does not modify or manipulate internally generated cause codes. See “Call release cause codes” (page 219) for more information.

# Chapter 9
# Routing fundamentals

Passport Voice Networking supports routing based on

- path-oriented routing system (PORS) using switched virtual circuits (SVC)
- PORS SVCs with data network address (DNA) support, based on dynamic packet routing system (DPRS)
- hunt group server, through which voice routes are accessible as hunt group members

Routing information is organized into the following sections:


- “Routing parameters” (page 134)
- “Voice Networking using PORS” (page 144)
- “Voice Networking based on DNAs and DPRS” (page 151)
- “Digit manipulation” (page 158)
- “Call redirection: when primary destinations are unavailable” (page 163)
- “Voice Networking call hunting” (page 163)
- “Voice Networking call server configuration” (page 166)
- “Voice, modem, and facsimile encoding choices and rates” (page 168)
- “Voice routes” (page 176)

# Routing parameters

Routing depends on configuration of the following parameters:

- numbering plans
- directory numbers
- dialing mode

Information on these routing parameters, as well as examples of dialed number resolution, are provided in the following sections:


- "Numbering plans" (page 134)
- "Directory numbers" (page 135)
- "Overlap and enbloc dialing" (page 136)
- "Examples of resolving dialed numbers through VNCS" (page 136)


## Numbering plans

A numbering plan consists of the following:

- the organization of digits (dialed numbers) under directory number prefixes
- the association of the dialed numbers to destination Passport nodes by means of destination node and component identification values or DNA parameters
- the application of quality of service parameters according to the dialed number

You define a numbering plan by configuring a Voice Networking call server (VNCS). A VNCS (of which there can be sixteen) represents one way to organize and route dialed numbers.

A numbering plan typically provides two separate internal access codes and an external plan (usually an international numbering plan). Voice Networking supports the following four basic types of numbering plans:

- co-ordinated numbering plan
  - typically fixed length addresses (for each node)

  - always dial all digits
  - typically 4 to 7 digits
  - routing can be based on a minimum set of digits
- flat (transferable) numbering plan
  - variable length addresses
  - typically no pattern to addresses on a node (for example, any number, anywhere)
  - every node has a table with all addresses in it
  - can only route on complete address
- location code (LOC)
  - each node has one or more unique codes or prefixes
  - length of code can vary but typically fixed
  - access code typically gets stripped off before going out on the trunk
- group dialing
  - combination of Co-ordinated Numbering Plan and LOC or a combination of Flat Numbering Plan and LOC

The table "Partial numbering plan of a VoiceNetworkingCallServer component" (page 137) illustrates a partial numbering plan.

## Directory numbers

Directory numbers are numeric prefixes that allow you to organize groups of dialed numbers (excluding the PBX access code). You define directory numbers by configuring the DirectoryNumber component.

When defining directory number prefixes for dialed numbers, you can use one or multiple wildcard characters, represented by the question mark character (?). The wildcard character in a directory number prefix instructs Voice Networking to ignore the digit in that position when making routing decisions. By using wildcards, you can simplify the configuration process by reducing the number of directory numbers needed to properly route calls. For example, by using the directory number prefix 3?1, you can combine dialed numbers beginning with the following digits under one DirectoryNumber

component: 301, 311, 321, 331, 341, 351, 361, 371, 381 and 391. As well, the use of wildcards is important in voice numbering plans that place access codes in arbitrary locations in the dialed number. See the table "Partial numbering plan of a VoiceNetworkingCallServer component" (page 137) for an example of how Voice Networking uses directory numbers to route dialed numbers.

## Overlap and enbloc dialing

The PORS path is established when enough digits are provided to the VoiceNetworkingCallServer component to find a route. Information in the PORS connect message informs the ingress SignallingChannel component if the destination VoiceRoute component is using overlap or enbloc dialing.

With overlap dialing, a connection is attempted before all digits are dialed. Each VoiceRoute component defines a minimum number of dialed digits that uniquely identify a VoiceRoute component on a destination Passport node. Once the minimum number of digits is dialed, a call request is made even though other digits may still be arriving. If the VoiceNetworkingCallServer component cannot resolve the digits to a destination node, then the request is returned specifying that the number is not unique.

When calling from an overlap dialing PBX to an enbloc dialing PBX, all digits must be collected before the setup message can be sent to the egress Passport node. This approach is known as overlap-to-enbloc interworking. This is accomplished either by a SignallingChannel component timer on the Passport node or by an explicit indication from the calling PBX, depending on the protocol.

> ***Note:*** For both the PBX and the Passport, the settings for overlap sending and overlap receiving should be the same. Set both the PBX and the Passport to either overlap dialing or enbloc dialing.

## Examples of resolving dialed numbers through VNCS

Examples of the best match algorithm, as applied to the partial numbering plan shown in the table "Partial numbering plan of a VoiceNetworkingCallServer component" (page 137), are as follows:

- 6137527 maps to nodeId 26
- 61375127 maps to nodeId 25

- 6137755325 maps to nodeId 31
- 61377553251 maps to nodeId 31
- 0 maps to nodeId 20

Based on the organization of digits (directory number prefixes), the best match algorithm would not map the following numbers:

- 613775 does not map (insufficient digits)
- 6137755326 does not map (number not supported)
- 1 does not map (number not supported)

**Table 13**
**Partial numbering plan of a VoiceNetworkingCallServer component**

| Directory number prefix | destination-NodeId attribute | destination-ComponentId attribute | (NPI) and DNA | voiceProfile-Number attribute |
|---|---|---|---|---|
| 6137645 | 10 | VoiceRoute/1 | (X.121) 55554000100001 | 3 |
| 9285 | | VoiceRoute/2 | (X.121) 55554000100002 | 1 |
| 92856 | | | | 1 |
| 9113 | | | | 2 |
| 9432 | | | | 3 |
| 613788 | | VoiceRoute/3 | (X.121) 55554000100003 | 3 |
| 613766<br>613825<br>613543 | 30 | VoiceRoute/2 | (X.121) 55554000300002 | 1 |
| 61377???25 | 31 | VoiceRoute/3 | (X.121) 55554000310003 | 3 |
| 613768 | 20 | VoiceRoute/1 | (X.121) 55554000200001 | 3 |

(Sheet 1 of 2)

**Table 13 (continued)**
**Partial numbering plan of a VoiceNetworkingCallServer component**

| Directory number prefix | destination-NodeId attribute | destination-ComponentId attribute | (NPI) and DNA | voiceProfile-Number attribute |
|---|---|---|---|---|
| 6137512 | 25 | VoiceRoute/2 | (X.121) 55554000250002 | 3 |
| 613752 | 26 | VoiceRoute/1 | (X.121) 55554000260001 | 1 |
| 1800 | 45 | VoiceRoute/1 | (X.121) 55554000450001 | 1 |
| 0 | 20 | VoiceRoute/3 | (X.121) 55554000200003 | 3 |
| ***Note:*** The ? character is a wildcard. | | | | |
| (Sheet 2 of 2) | | | | |

## Interaction between configurable components and attributes during call routing

At the source Passport node, the destination for an incoming call is determined by mapping the dialed number to a DirectoryNumber component instance under VoiceNetworkingCallServer DiallingPlan component instance. The DirectoryNumber component represents a prefix directory number for organizing groups of dialed numbers. The DirectoryNumber component contains attributes that specify the destination of, and parameters associated with (from the VoiceProfile component), the dialed number.

The mapping process occurs in the following order:

1 The PBX numbering plan indicator (NPI) and type of number (TON) that are associated with the dialed number map to a Passport term (for example, international). See “Mapping numbers—PBX to Passport” (page 142).

2 The Passport term maps to a particular VoiceRoute component diallingPlan attribute (for example, dp1). See “Mapping numbers—Passport term to VoiceRoute diallingPlan attribute” (page 143).

3 The VoiceRoute component diallingPlan attribute maps to a VoiceNetworkingCallServer DiallingPlan component instance (for example, DP/1). See “Mapping numbers—VoiceRoute to VoiceNetworkingCallServer DiallingPlan component” (page 143).

The DirectoryNumber component, located under the VoiceNetworkingCallServer DiallingPlan/x component instance, contains the information necessary to route the call to its destination.

### Call routing example

The following steps show how Voice Networking routes a call using PORS SVCs without DNA support. The figure “Conceptual example of Voice Networking call routing (without DNA support)” (page 141) illustrates the progression of these steps.

The following steps describe the concepts illustrated in “Conceptual example of Voice Networking call routing (without DNA support)” (page 141):

1 A subscriber dials 4464.

***Note:*** The parameters used throughout this example are for illustrative purposes only. Each network has unique requirements.

2 The PBX is setup for enbloc dialing. After receiving the dialed number, the PBX maps it to Numbering Plan Indicator (NPI) 1001 and Type of Number (TON) 001.

3 There are two VoiceSubroute components on the Primary Rate Interface (PRI): Vsr/1 contains timeslots 1-15 and Vsr/2 contains timeslots 17 -30. Both of these voice subroutes are associated with SignalingChannel/1. On an E1 link, timeslot 16 is reserved for the SignalingChannel component. Timeslot 16 is used to exchange signaling information between the PBX and Passport node.

4 Signaling information is sent, by way of timeslot 16, to the Passport node.

5 At the ingress Passport node, SigChan/1 identifies which subroute will handle the call.

6 Based on the requested voice path timeslot, the call is processed by Vsr/1.

7 Vsr/1 maps to VoiceRoute/1.

8 The combination of NPI 1001 and TON 001 (from the PBX) maps to Passport term P1 by way of a lookup table. See the table “Mapping PBX terms to Passport terms” (page 142).

9 Passport term P1 maps to the diallingPlan1 attribute under VoiceRoute/1. See the table “Mapping example: Passport terms to VoiceRoute component diallingPlan attribute values” (page 143).

10 VoiceRoute/1 maps to VoiceNetworkingCallServer/1.

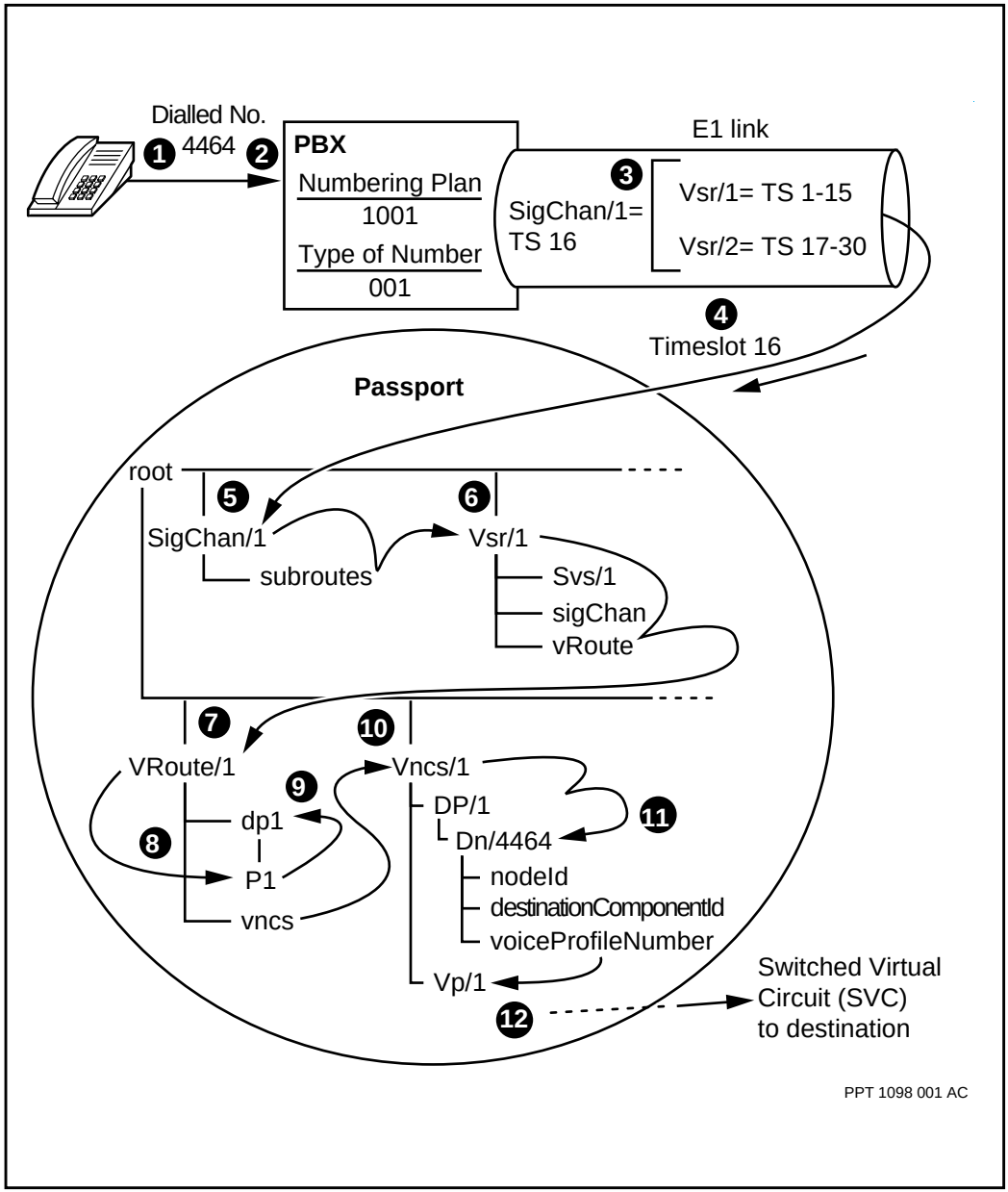
11 VoiceNetworkingCallServer/1 maps the dialed number to DP/1 (from step 9) and Dn/4464.

Dn/4464 maps the dialed number to a destination node and component ID (under the destinationNodeId and destinationComponentId attributes, respectively) and a voice profile (under the voiceProfileNumber attribute). See the table “Partial numbering plan of a VoiceNetworkingCallServer component” (page 137).

12 The value configured under the voiceProfileNumber attribute maps the dialed number to Vp/1.

VoiceProfile/1 contains all of the configurable attributes necessary for the VoiceSubroute SwitchedVoiceService LogicalConnection and Framer components to set up an SVC to the destination Passport node.

**Figure 30**
**Conceptual example of Voice Networking call routing (without DNA support)**



### Mapping numbers—PBX to Passport

The table "Mapping PBX terms to Passport terms" (page 142) illustrates how a PBX NPI and TON combination map to a Passport term for use within the Passport subnet.

**Table 14**
**Mapping PBX terms to Passport terms**

| PBX NPI | PBX TON | Passport term |
|---|---|---|
| 0000 (Unknown) | XXX | Unknown |
| 0001 (E.164) | 000 | Unknown (see Note 1) |
| | 001 | International |
| | 010 | National |
| | 100 | Subscriber |
| 1001 (Private) | 000 | P0 |
| | 001 | P1 |
| | 010 | P2 |
| | 011 | P3 |
| | 100 | P4 |
| | 101 | P5 |
| | 110 | P6 |
| | 111 | P7 |
| XXXX | XXX | casUnknown (see Note 2) |

***Note 1:*** PBX NPI 0001 (E.164) and TON 000 combination is identical to the PBX NPI 0000 (Unknown) and TON XXX combination. Voice Networking maps these combinations in the same way.

***Note 2:*** The value casUnknown applies to any number received from channel associated signaling (CAS) trunks. The value casUnknown does not correspond to a particular PBX NPI and TON combination.

### Mapping numbers—Passport term to VoiceRoute diallingPlan attribute

The table “Mapping example: Passport terms to VoiceRoute component diallingPlan attribute values” (page 143) provides an example of how a Passport term can be mapped to a VoiceRoute component diallingPlan attribute. In each VoiceRoute component, the TON can be mapped to a specific diallingPlan attribute (for example, international can be mapped to one of dp0, dp1, or dp2) through configuration. Alternately, a TON (for example, national) can be excluded from a diallingPlan attribute.

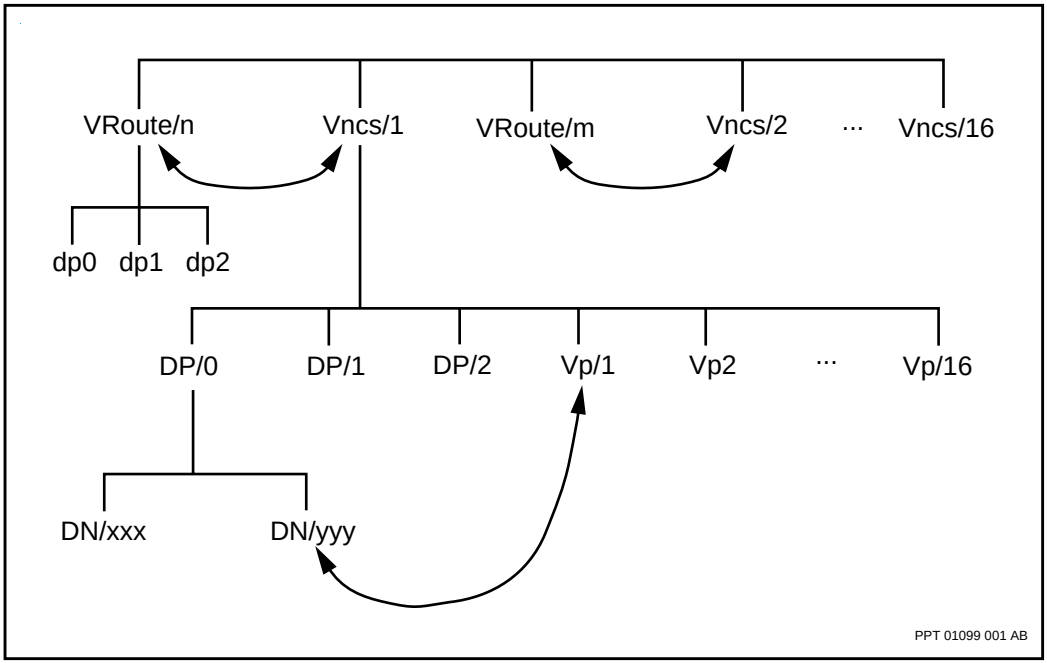
**Table 15**
**Mapping example: Passport terms to VoiceRoute component diallingPlan attribute values**

| **Passport term** | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| **Unknown** | **International** | **National** | **Subscriber** | **P0** | **P1** | **...** | **P7** | **casUnknown** | **dialling-Plan attribute** |
| X | | | | | | | | | dp0 |
| | | | | X | X | | | | dp1 |
| | X | | | | | | | | dp2 |
| | | | | | | | | | |

### Mapping numbers—VoiceRoute to VoiceNetworkingCallServer DiallingPlan component

A VoiceRoute component instance maps to a VoiceNetworkingCallServer component instance. The figure *“VoiceRoute to VoiceNetworkingCallServer component association” (page 144)* provides an example. There can be up to sixteen VoiceNetworkingCallServer instances on each Passport node, with each VoiceNetworkingCallServer component containing three DiallingPlan component instances (DP/0, DP/1, DP/2). The DirectoryNumber component, under a particular VoiceNetworkingCallServer DiallingPlan component instance, contains attributes which specify the destination and calling parameters for a dialed number.

**Figure 31**
**VoiceRoute to VoiceNetworkingCallServer component association**




## Voice Networking using PORS

PORS establishes and maintains a transmission path to transmit voice and data traffic across a Passport network. PORS offers two services to maintain transmission paths: failure recovery and, optionally, periodic optimization of voice paths. For more information on PORS, refer to 241-5701-435 *Passport 7400, 15000, 20000 Path-Oriented Routing System Guide*.

Voice Networking uses PORS SVCs to transmit data and voice traffic. PORS SVCs are established and torn down on a per-call basis. An SVC communicates with the VoiceNetworkingCallServer component to determine the destination of a call, and with the route selector to find a route to that destination. The values configured for the destinationNodeID and destinationComponentID attributes under the VoiceNetworkingCallServer component determine the destination of a call.

PORS negotiates and reserves bandwidth and establishes the communication path. When a call setup message reaches and is accepted by a destination node, the bandwidth is guaranteed and the call can proceed.

> ***Note:*** The use of PORS SVCs in Voice Networking means that PORS setup time is also a factor in voice call setup timing.

Information on Voice Networking through PORS is provided in the following sections:

- "PORS and switched voice calls" (page 145)
- "Numbering plans" (page 134)
- "Directory numbers" (page 135)
- "Overlap and enbloc dialing" (page 136)
- "Examples of resolving dialed numbers through VNCS" (page 136)

## PORS and switched voice calls

The PORS application service interface supports the following:

- switched logical connection (LC) VoiceProfile components
- RouteSelector component modifications for node identification instead of the string based node name
- Dynamic path adjustment. Dynamic path adjustment allows applications using PORS to request changes to their bandwidth requirement and holding priority. Applications can adjust their bandwidth reservation downwards based on the estimation of their bandwidth after call establishment.

PORS logical connections maintain a virtual connection of path-oriented data between two endpoints. PORS includes the following components:

- a dynamic switched logical connection component that is derived from the configured VoiceProfile component and the application.

- an operational LogicalConnection component. The switched logical connection defines all attributes of an LogicalConnection component for both ends of a connection and is therefore virtually identical to the LogicalConnection component. The only exception is that some attributes from the application can be adjusted after call establishment.

## Description of PORS call setup and tear-down

The call routing scenarios described in the following sections are based on routing using PORS without data network address (DNA) support. The following scenarios are described:


- "Call setup—ingress Passport node" (page 146)
- "Call setup—egress Passport node" (page 149)
- "Call tear-down" (page 151)


### Call setup—ingress Passport node

Call setup begins at the ingress Passport node as described in the following steps. The figure "Message flow during call establishment—at the ingress Passport node" (page 148) illustrates these concepts.

1 The SignalingChannel component on the ingress Passport node receives the incoming dialed number (for example, 55543897) from PBX A.

2 The SignalingChannel component sends the formatted message (55543897) to the VoiceSubroute component, which in this case acts as a signal multiplexor/de-multiplexor.

3 The VoiceSubroute component passes the message (55543897) to the appropriate SwitchedVoiceService component.

4 The SwitchedVoiceService component initiates an address resolution request to the VoiceNetworkingCallServer component. In this example, the VoiceNetworkingCallServer component contains the following configuration information:

VoiceNetworkingCallServer/4 DiallingPlan/1 DirectoryNumber/5554

where destinationNodeId = 526, destinationComponentId = VoiceRoute/9, and voiceProfileNumber = 1.

5 In this example, a unique match is found for the prefix-DN 5554. The VoiceNetworkingCallServer component returns the node identification number of the destination Passport and the parameters from the VoiceProfile component associated with the dialed number.

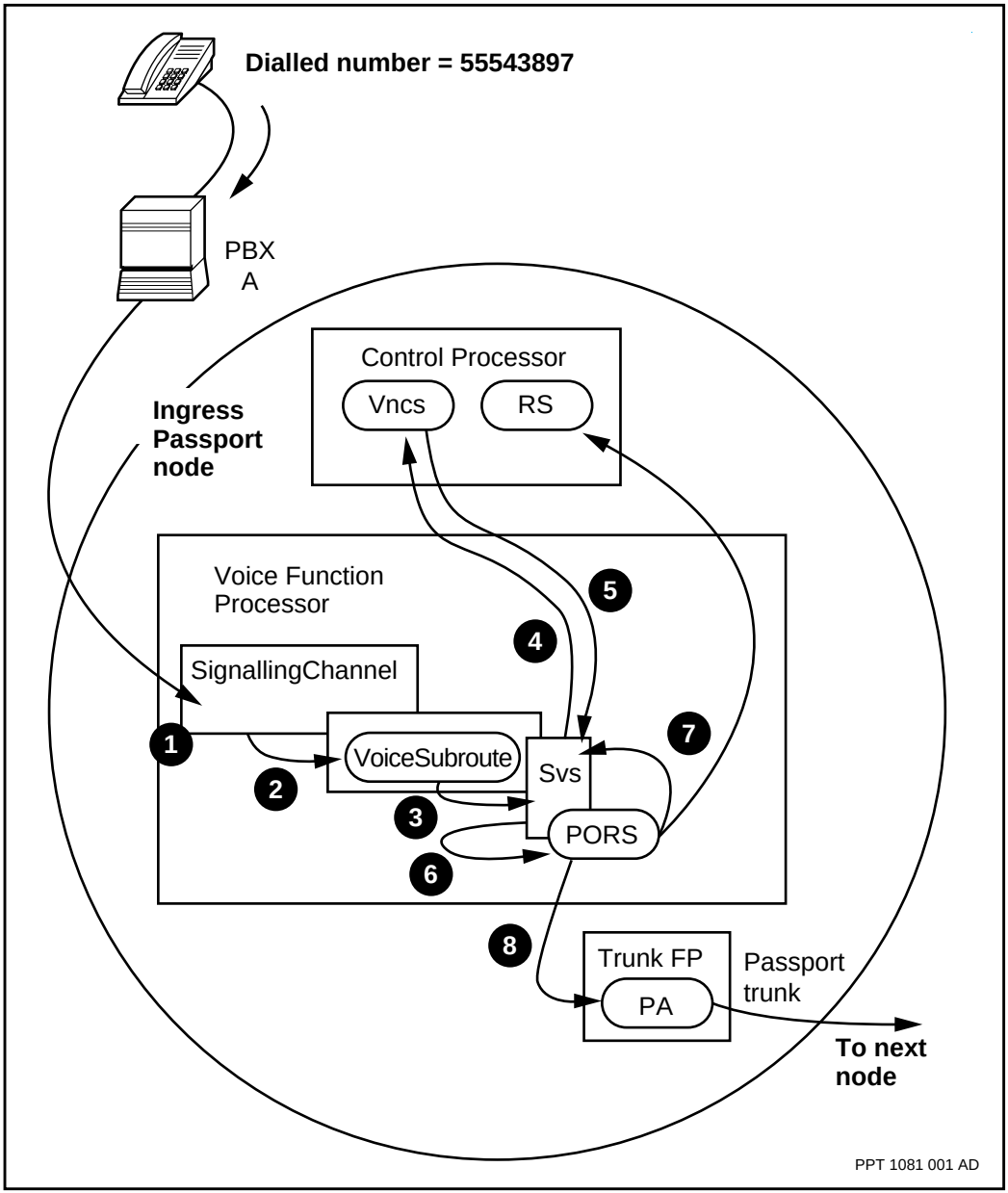
6 The SwitchedVoiceService component passes the information to the PORS.

7 PORS, by way of the RouteSelector component, attempts to establish a SVC connection with the egress Passport, using the parameters from the VoiceProfile component. PORS also informs the SwitchedVoiceService component of the quality of service that is required.

8 The Trunk PathAdministrator component sends a path setup packet, containing information on all the Passport trunks to be used, to the egress Passport node by way of the PathAdministrator component on each node along the data path.

At each intermediate node, the PathAdministrator component sets up the connection across the Passport trunk (that is, assigns logical channel numbers, verifies bandwidth availability, and reserves bandwidth).

**Figure 32**
**Message flow during call establishment—at the ingress Passport node**



### Call setup—egress Passport node

Call setup continues at the egress Passport node as described in the following steps. The figure “Message flow during call establishment—at the egress Passport node” (page 150) illustrates these concepts.

1 At the destination Passport node (nodeId = 526), the Final Destination Call Router (FDCR) hands-off the path setup packet to the appropriate VoiceRoute component instance (in this example, VoiceRoute/9).

2 The VoiceRoute component selects one of its VoiceSubroute component instances to receive the path setup message. For VoiceRoute components that have more than one VoiceSubroute component, the huntingAlgorithm attribute specifies how to search for and select an available VoiceSubroute component instance.

3 The VoiceSubroute component selects a SwitchedVoiceService component to take the call and processes the message.

4 The SwitchedVoiceService component requests that PORS establish the SVC with the ingress Passport node. A path setup confirmation packet is returned to the ingress node to enable the path for data transfer. Once the path is established, the ingress Passport node sends the call setup message across the network to the egress Passport node.

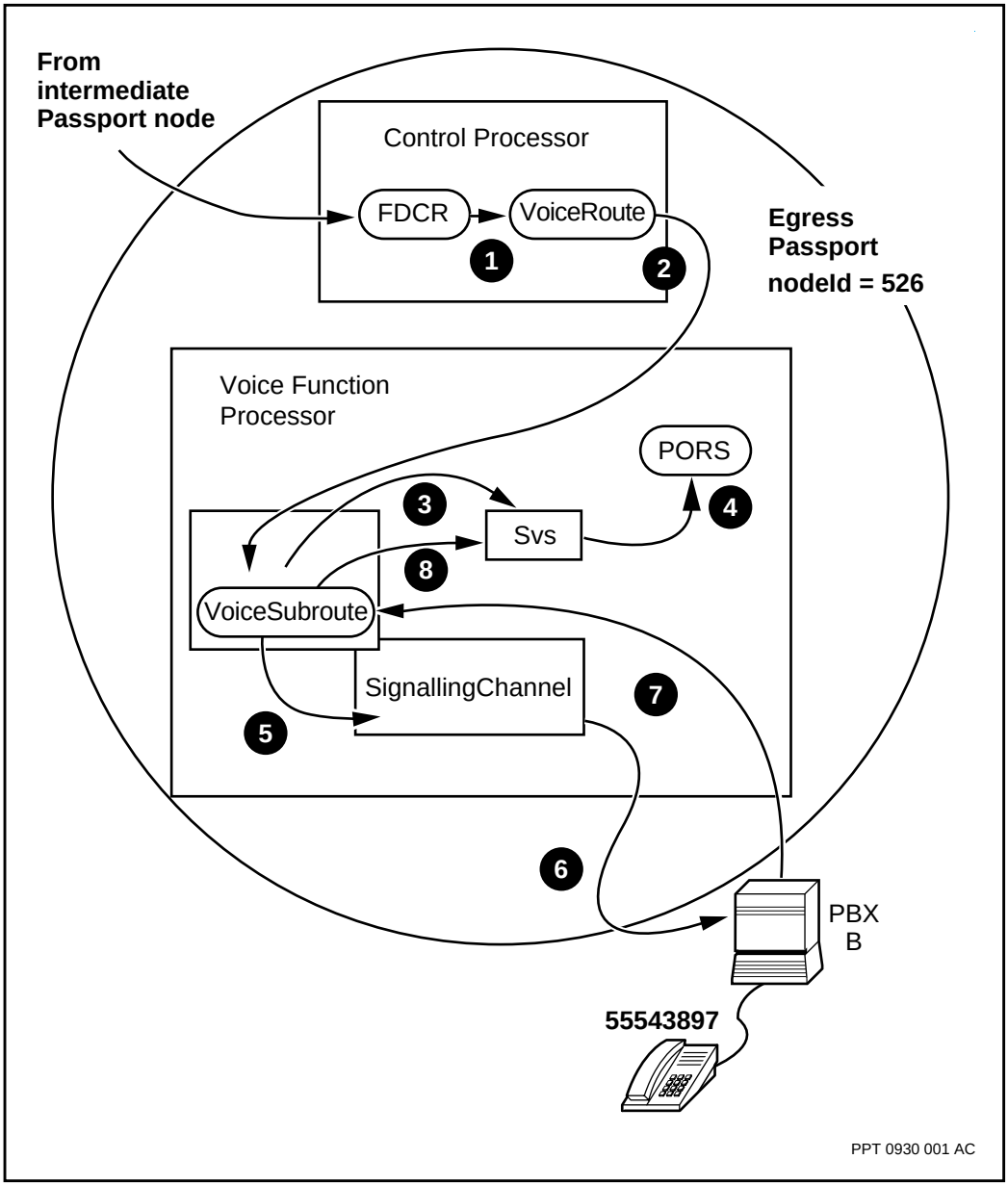
5 The VoiceSubroute component passes the call setup message to the SignalingChannel component.

6 The SignalingChannel component formats and sends the call setup message to the destination PBX (in this example, PBX B).

7 PBX B informs the VoiceSubroute component that it is ready to accept the call.

8 The VoiceSubroute component informs the appropriate SwitchedVoiceService component that a new outgoing call is established.

**Figure 33**
**Message flow during call establishment—at the egress Passport node**



### Call tear-down

An established call terminates when the ingress or egress PBX sends a valid disconnect message into the Passport network. The following steps show how the network completes the call termination request.

1 The SignalingChannel component forwards the disconnect message to the VoiceSubroute component.

2 The VoiceSubroute component forwards the disconnect message to the appropriate SwitchedVoiceService component.

3 The SwitchedVoiceService component forwards the disconnect message through the Passport subnet.

4 The SwitchedVoiceService component informs the SignalingChannel component to acknowledge the disconnect message to the local PBX.

5 The SwitchedVoiceService component initiates a call clear by way of the Passport Subnet Routing components and systems.

6 The far-end SwitchedVoiceService component forwards the disconnect message to its SignalingChannel component.

## Voice Networking based on DNAs and DPRS

Through DNAs, Voice Networking is compatible with DPRS and aligns with the Passport Frame Relay service.

For more information on DPRS, see 241-5701-425 *Passport 7400, 15000, 20000 Dynamic Packet Routing System Guide*.

Information on Voice Networking based on DNAs and DPRS is provided in the following sections:


- “DNAs and PORS SVCs” (page 152)
- “Call setup steps with DNA support” (page 152)
- “Prerequisites to implementing DNA support” (page 154)
- “How to implement DNA support—new installation” (page 155)
- “How to migrate to DNA support—existing network” (page 156)

## DNAs and PORS SVCs

With DNA support, a PORS SVC determines the destination of a voice connection by looking up the digits configured for the dataNetworkAddress attribute under the VoiceNetworkingCallServer DiallingPlan DirectoryNumber component. These digits form the destination's unique identifier.

Address resolution for PORS SVCs with DNA support involves mapping a received dialed number to an internal DNA under a DirectoryNumber component. DNAs are associated with a numbering plan indicator (NPI) which is either X.121 or E.164 (see 241-5701-405 *Passport 7400, 15000, 20000 Call Server Guide* for more information on DNAs and numbering plans).

***Note:*** PORS SVCs use DNAs when the destinationNodeID and destinationComponentID attributes are not present in the VoiceNetworkingCallServer DiallingPlan DirectoryNumber component's database.

## Call setup steps with DNA support

The following steps describe a typical Voice Networking call setup using PORS SVCs with DNA support. The figure "Conceptual example of call establishment using PORS SVCs with DNA support" (page 154) illustrates this example.

1 Call setup request from Access Call Control (ACC) to PORS VC of the source node. The call setup request message contains the DNA of the destination node, and network and PORS parameters.

***Note:*** The node ID and destination component ID of the destination node are used instead of the DNA if they are present in the VoiceNetworkingCallServer DiallingPlan DirectoryNumber component's database.

2 DPRS call request from PORS VC of the source node to the Final Destination Call Router (FDCR) of the destination node. The FDCR passes the call request to the destination VoiceRoute component.

3 DPRS abort from VoiceRoute component of the destination node to PORS VC of the source node.

In response to the query, the VoiceRoute component at the destination node returns its node ID and process ID (PID) by means of the DPRS abort message. The node ID and PID of the VoiceRoute component at the destination node are required by the source node before PORS call establishment can begin.

4 PORS call request from PORS VC of the source node to the VoiceRoute component of the destination node.

PORS call establishment procedures begin. The node ID determines the path and the destination PID determines the VoiceRoute component.

5 PORS call request forwarded from the VoiceRoute component to PORS VC of the destination node.

The VoiceRoute component at the destination node selects a PORS VC. The PORS call request message describes the source node.

6 End-to-end compatibility check by the ACC at the destination node.

Based on the description of the source node and capability of the destination node, the connection parameters are selected and sent to the source node. If the connection is rejected, the PORS call request is aborted.

7 PORS call connected from PORS VC of the destination node to the PORS VC of the source node.

The PORS call is now established.

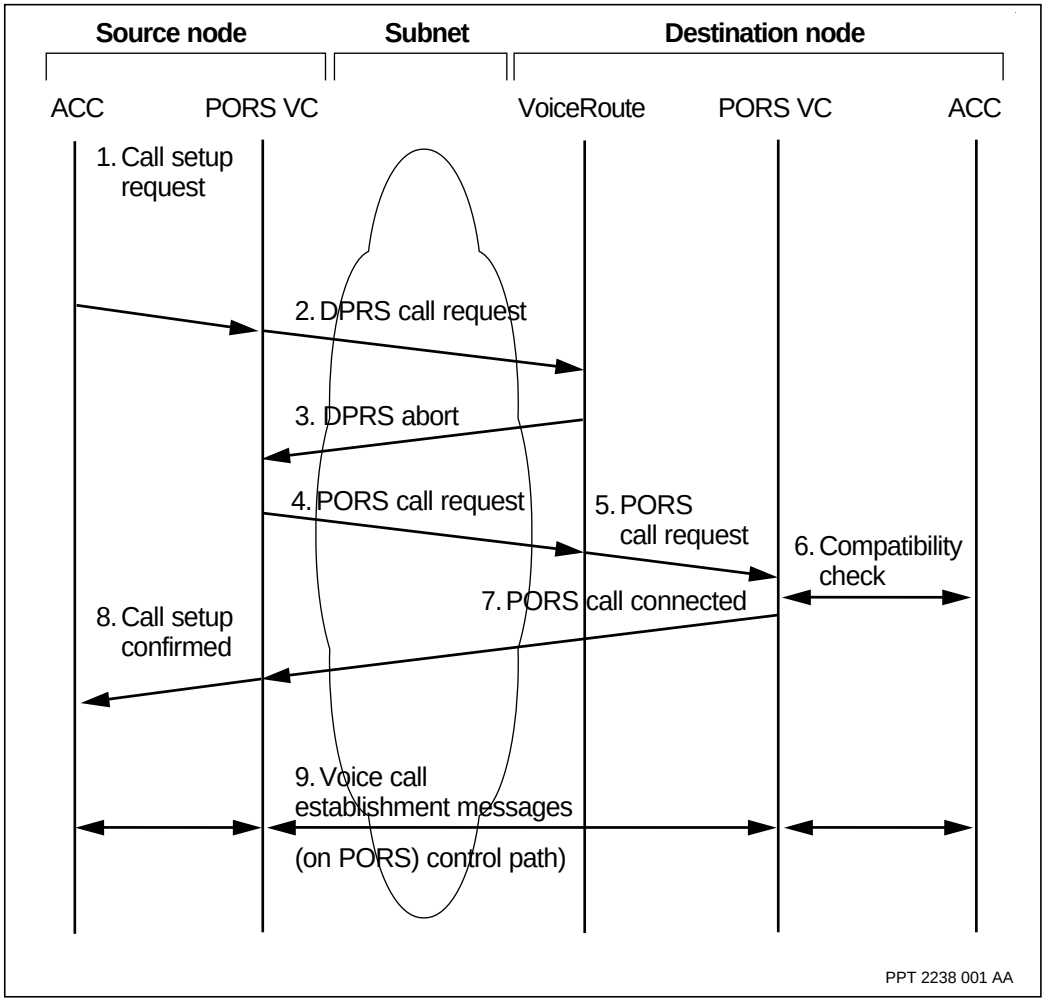
8 Call setup confirmed message from PORS VC to ACC of the source node.

The ACC at the source node is notified that call establishment succeeded and is provided with the connection parameters.

9 Voice Networking call establishment.

Upon establishment of a Voice Networking call, messages are exchanged directly between the two applications involved in the call.

**Figure 34**
**Conceptual example of call establishment using PORS SVCs with DNA support**




## Prerequisites to implementing DNA support

Each node in a network that supports Voice Networking calls must have some call routing capabilities by way of either of the following:

- CallRouter component on Passport
- Call Server Resource Manager (CSRM) through a DPN gateway

If a CSRM is available through a DPN gateway, it will service all Passport nodes which are in the same routing module or RID. The NPI (E.164 or X.121) and the DNA must be configured in the CSRM or the CallRouter component to allow routing between nodes in the network.

## How to implement DNA support—new installation

To implement DNA support for Voice Networking, assign each VoiceRoute component a DataNetworkAddress subcomponent (see “Considerations for attributes under the DataNetworkAddress component” (page 155) for more information). This makes each VoiceRoute component visible to DPRS. The DataNetworkAddress subcomponent has the following two configurable attributes:

- dataNetworkAddress
- numberingPlanIndicator

### Considerations for attributes under the DataNetworkAddress component

When defining the attributes of a DataNetworkAddress component, observe the following rules:

- The NPI/DNA pair has to be unique in the entire network. Each VoiceRoute component registers its NPI/DNA pair at its node’s FDCR.
- The VoiceRoute component’s NPI/DNA pair is used to configure VoiceNetworkingCallServer DiallingPlan DirectoryNumber component entries which terminate on the VoiceRoute component.
- Configure DPRS to support NPI/DNA pairs for all VoiceRoute components in the network by configuring DNA prefixes on the CallRouter component. This implies the following:
  - — To keep the number of entries in the CallRouter component’s table low, configure all VoiceRoute components on a single node with the same NPI and DNA prefix. This creates only one entry for each Passport node in the CallRouter component’s table.
  - — To keep DNA prefixes in the CallRouter component’s table short, the VoiceRoute components on different nodes require different DNA prefixes or NPIs, or both.

— If a VoiceRoute component is added with a DataNetworkAddress component that does not follow the NPI or DNA prefix, or both, of the existing VoiceRoute components on the node, ensure that a DNA prefix exists in the CallRouter component's table to properly route calls to the added VoiceRoute component's node.

For more information on the call router, refer to 241-5701-405 *Passport 7400, 15000, 20000 Call Server Guide*.

## How to migrate to DNA support—existing network

The following steps describe how to migrate a Voice Networking service which uses PORS node IDs to establish SVCs and route traffic to one that supports the use of DNAs to establish SVCs and route traffic:

**CAUTION**
**Service disruption**
During the migration to DNA support, service disruptions are possible if your DPRS configuration is not set up properly (see points 7 and 8 below). To avoid a major outage, DO NOT REMOVE the destination node and component IDs all at once.

1 Prepare each Passport node that supports Voice Networking without DNA capability for a software upgrade.

Ensure that the Routing DpnAddressPlan component exists and is configured properly. If the Routing DpnAddressPlan component does not exist, add the dpnRouting feature to the featureList attribute of the Software LogicalProcessorType component instance for the control processor (CP). For more information on configuring the Routing DpnAddressPlan component, refer to 241-7401-110 *Passport 7400, DPN-100 Interworking Guide*.

2 Upgrade each Passport node that supports Voice Networking without DNA capability with, at a minimum, R4.2 software.

3 Set the VoiceRoute DataNetworkAddress component's attributes.

Ensure that the ModuleData VirtualCircuitSystem component exists and is configured properly. If using a call router, ensure that all of the DNA prefixes that you require are in the CallRouter component's table and are correctly configured. If using a CSRM, ensure that it is visible from the node and that the DNAs are correctly configured under the appropriate NPIs.

4 The nodes now support both PORS node ID-based routing and DNA-based routing. The nodes are ready to accept a DPRS call (using DNAs), but cannot initiate a call using DPRS.

For all calls originating from these nodes, the destination node and destination component IDs are passed to PORS in the call setup request message and the VoiceNetworkingCallServer component's translation reply includes destination node ID and destination component ID Information Elements (IEs).

5 Add DNAs to the VoiceNetworkingCallServer component's database. However, do not remove the destination node IDs and destination component IDs at this time (see "Considerations for attributes under the DataNetworkAddress component" (page 155)).

Prepare a central VoiceNetworkingCallServer component database file with all configured DNA entries and distribute the file to all nodes within the network. If the VoiceNetworkingCallServer component's database is centrally managed, use Preside Multiservice Data Manager (PMDM) Global Data Manager (GDM) to distribute the dialing plans.

6 Activate the configuration changes. The nodes still support both types of routing (PORS node ID and DNA). The calls, however, continue to be set up using the destination node ID and destination component ID values. DNAs are not yet used for routing, but they can be validated by the PMDM Network Reporting System (NRS)-based Service Integrity Checks (NSIC) tool. This allows the user to correct errors in the VoiceNetworkingCallServer component's database without incurring operational problems. For more information on the NSIC tool, refer to 241-6001-022 *Preside MDM Network Reporting System User Guide*.

7 Correct any DNA configuration problems in the central VoiceNetworkingCallServer component's database.

8 Remove the destination node IDs and destination component IDs from the VoiceNetworkingCallServer component's database, one set at a time.

9 Distribute the VoiceNetworkingCallServer component's database file to all nodes within the network. Activate the configuration. DNAs are now used for routing. DPRS call setup messages can now be processed by the VoiceRoute components at the destination nodes. Every node upgraded in this manner makes use of the DNA-based PORS call setup procedure.

# Digit manipulation

Digit manipulation permits the service provider to delete and insert digits at the ingress and egress switches on a Passport network. This capability allows the service provider to continue using PBXs that do not support digit manipulation in networks where this capability is required. Further, digit manipulation provides increased flexibility when building the Passport VNCS tables since the service provider can now route calls based on either dialed digits or manipulated digits. This flexibility also has the potential for reducing the size of the VNCS tables. Digit manipulation is supported for the NIS A211-1, ETSI QSIG, Euro- ISDN, DS1 CAS and E1 CAS protocols. In addition, digit manipulation supports the following gateways: ETSI QSIG to DS1 CAS, ETSI QSIG to E1 CAS, NIS A211-1 to DS1 CAS and NIS A211-1 to E1 CAS.

Information on digit manipulation is organized into the following sections:


- "Summary of features" (page 158)
- "Supported digit types" (page 159)
- "Empty digit strings at ingress" (page 160)
- "Empty digit strings at egress" (page 160)
- "Accounting impacts of digit manipulation" (page 162)


## Summary of features

Passport digit manipulation supports the following capabilities:

- deletion at the ingress Passport switch of up to 16 leading digits from the dialed number digit string received from the originating PBX

- insertion at the ingress Passport switch of a digit sequence of up to 16 digits at the beginning of the dialed number digit string received from the originating PBX
- deletion at the egress Passport of up to 16 leading digits from the ingress called number digit string to be sent to the destination PBX
- insertion at the egress Passport of a digit sequence of length up to 16 digits at the beginning of the ingress called number digit string to be sent to the destination PBX
- configuration at the voice route level, which allows the service provider to
  - — apply the same digit manipulation rules to all subroutes
  - — turn digit manipulation off for any voice route without affecting other voice routes
- configuration of call routing to use either dialed digits or manipulated digits
- logging for ingress accounting records, to record the dialed number (as received from the PBX) as the called number
- logging for egress accounting records, to record the called number as the called number received from the ingress side prior to any egress digit manipulation
- support for both en bloc and overlap dialing types

## Supported digit types

Passport digit manipulation supports the following digits types:

- 0 through 9
- octothorpe (#)
- asterisk (*)
- A through D

Through digit manipulation, the Passport switch can delete any of these digit types from the called number. However, the switch can insert only digit types 0 through 9, octothorpe, and asterisk into the called number. If digit manipulation is configured to insert an octothorpe or asterisk, then routing is

based on the dialed digits (not the modified digits) since only the digit type 0 through 9 is supported by VNCS. For example, if the string that digit manipulation inserts is

613*

and the minimum digits required to route is 4, then calls associated with that voice route always fail since the fourth digit sent with a routing request is always an asterisk.

## Empty digit strings at ingress

In a configuration without digit manipulation, the ingress Passport switch never forwards an empty called number string since network protocol requires a minimum of one digit at the ingress side to route the call. If a called number contains no digits, the call fails.

With digit manipulation, an empty digit string occurs at ingress when the number of digits to delete is greater than or equal to the number of digits received and there is no insertion digit string. In this configuration, an empty digit string is forwarded to the egress switch if routing is configured to use dialed digits. If routing is configured to use modified digits, the call fails.

An empty string is acceptable to support the following call handling requirements:

- an overlap dialing call has not yet received all of the digits from the originator and more digits may follow, which allows support of normal non-auto-terminate trunks
- the egress side can be configured to support digit manipulation
- the egress side can interface with a PBX that supports empty digit strings through call interception or auto-terminate trunks

Passport digit manipulation options permits the service provider to configure the ingress switch to route on either the dialed digits or the manipulated digits.

## Empty digit strings at egress

Empty digit strings are created by delete operations at the egress side of the Passport network. These empty digit strings have an impact on call SETUP/SEIZE procedures.

In networks that do not have digit manipulation at egress, the Passport switch transmits the SETUP/SEIZE message to the egress PBX as soon as it receives digits. The PBX then starts an inter-digit timer. When egress digit deletion is configured, the Passport switch holds back the SETUP/SEIZE message to the egress PBX until either the number of digits received at the egress side is equal to the number of digits to be deleted or the switch detects an “end of dial”. This hold back allows the Passport switch to process received digits. When the required number of digits are deleted, the Passport switch sends the SETUP/SEIZE message to the PBX; all remaining digits are now governed by the ingress PBX link inter-digit timer. Without the hold back, the timer on the PBX can expire while waiting for the first post-deleted digit.

If the switch detects an “end of dial” digit type before the number of digits received is equal to the number of digits to be deleted, the call is not further delayed. The switch completes the deletion (that is, all received digits are deleted) and then completes any configured egress insertion.

For digit manipulation with no insertion string configured, there are three cases where an empty digit string can result:

- the Passport switch detects an “end of dial” digit type when the number of digits to delete is greater than the number of digits received
- the number of digits received is equal to the number of digits to delete but more digits are forthcoming (that is, overlap dialing)
- an empty digit string was send from the origination side

In all cases, there must be no insertion string provisioned for in the digit manipulation. If an insertion string is provisioned, then the insertion string will make up the digit string.

Empty digit string handling is configurable either to attempt call completion or to release the call.

### Attempt call completion

When attempting call completion, the egress switch forwards the empty digit string to the PBX. The PBX then makes the final determination for the call.

### Release the call

If configure to release the call, the egress switch records cause code 62, “Inconsistency in designated outgoing access information and subscribe class” in accounting records against the rejected call. However, the egress switch translates cause code 62 to cause code 3, “No route to destination” and sends it back to the originating PBX. Cause code 3 permits many PBXs to respond by retrying the call over an alternate route. As with any rejected call, the switch increments the egress VoiceRoute operational attribute *callsRejected* and the ingress *VoiceSubroute* operational attribute *callsRejectedByFarEnd*.

If the digit string has only an “end of dial” indicator, call clearing is returned to the subnet as usual. However, the Passport switch must also clear toward the destination with appropriate call clearing messages. For ISDN, the switch sends cause code 31, “Normal, unspecified”.

## Accounting impacts of digit manipulation

With digit manipulation, accounting on the ingress switch logs the dialed number as the called number. At the egress side of the network, accounting logs the ingress called number as the called number received by the egress switch prior to any egress digit manipulation.

By choosing to log these numbers rather than the post-manipulated numbers alone, there is no loss of information. If the switch logs only the post-manipulated ingress and egress numbers, the service provider cannot reconstruct the dialed and egress numbers if a digit deletion is applied since all deleted digits are lost. By logging the dialed and egress received numbers, the service provider can reconstruct the numbers transmitted by the ingress and egress sides of the network but only if the configured ingress and egress digit manipulations are known at the time the call was placed.

With digit manipulation configured, the called number accounting record at the ingress side of the network may not match the called number accounting record at the egress side. Without digit manipulation, the ingress and egress called numbers are always the same, barring any network loss.

Accounting records a clear cause code of 62 for any empty string call that is released.

# Call redirection: when primary destinations are unavailable

In a typical network, calls fail to complete if the primary destination node or route is unavailable. Passport has a call redirection feature that permits the service provider to route calls through a hunt group server, which directs the call to one of a group of eligible routes. This feature is not to be confused with the call redirection server, which redirects a call based on one of address, routing identifier/module identifier (RID/MID), or RID alone.

Call redirection is formally known as Voice Networking call hunting. For information on this capability, see “Voice Networking call hunting” (page 163).

# Voice Networking call hunting

The Passport network can be configured so that voice routes are accessible through a hunt group server. This configuration is known as Voice Networking call hunting. The call hunting feature provides the ability to load share calls across multiple Passports and voice routes in the network, thereby improving network utilization, performance, and reliability.

To configure call hunting, a primary hunt group server with voice routes as hunt group members is configured on a Passport node. For redundancy, a backup hunt group server can also be configured on another Passport in the network. The backup hunt group server supports call hunting when the primary hunt group server is out of service. See 241-5701-415 *Passport 7400, 15000, 20000 Hunt Group Server Guide*.

## Overview of call hunting routing process

Voice Networking is more robust when directing calls to a hunt group server, since the hunt group server can offer a choice of multiple voice routes.

Calls destined for a Passport hunt group server DNA are routed to the hunt group server, which in turn chooses an eligible hunt group member DNA from the list of members. In this configuration, the members are voice route DNAs. If the voice route is available, the network forwards the call. Otherwise, the hunt group server selects the next eligible member in the list. The choice of hunt group member depends on the search mode that you

configure for the server and the availability of the voice route. Availability of each voice route is signaled to the hunt group server through the availability message packet (AMP).

This process continues until a voice route can carry the call or the member list is exhausted. If the member list is exhausted and the call is not completed, the network rejects the call if a backup hunt group server is not available. See 241-5701-415 *Passport 7400, 15000, 20000 Hunt Group Server Guide* for further explanation of search modes and using the backup hunt group server.

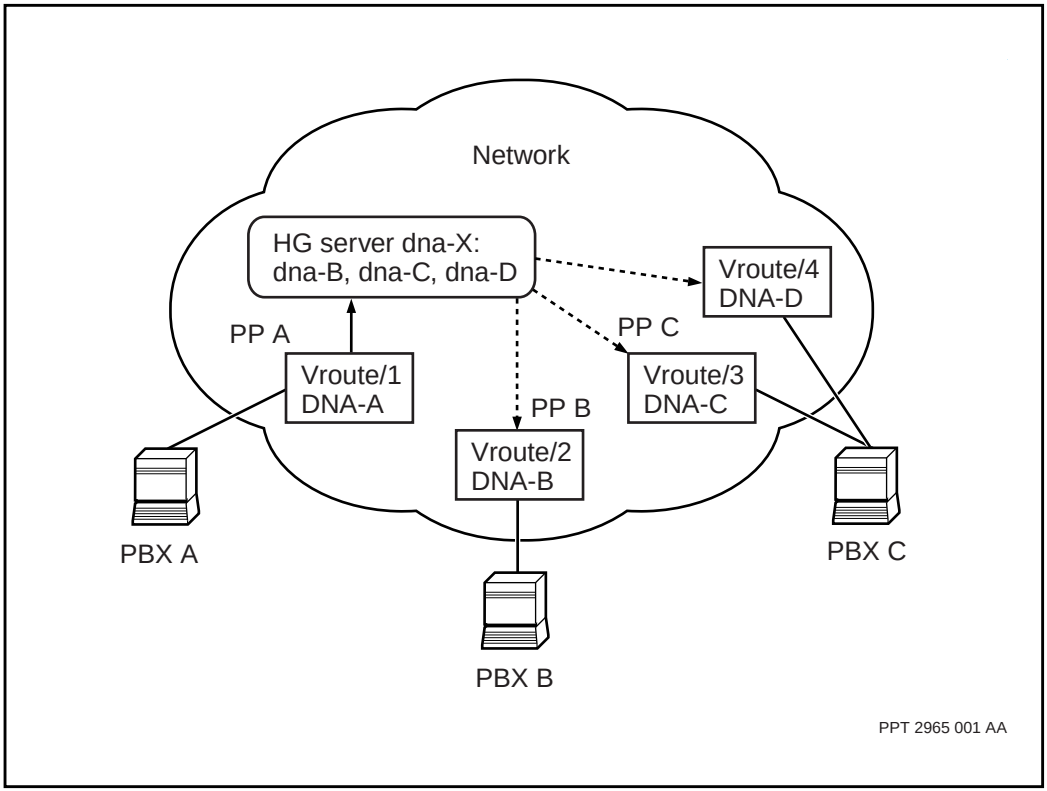
Call hunting supports call routing under the CCS and CAS protocols.

## Example of call setup

The figure “Example of a Voice Networking call hunting” (page 165) provides an example of call hunting through a hunt group server.

A call request comes from PBX A to Passport A (PP A). The VNCS for PP A resolves the destination address which, in this example, is the hunt group server DNA X. A call request is sent to the hunt group server based on the DNA. The hunt group server, upon receiving the call request, will select one of it's members (dna-B, dna-C, dna-D), based on the server's search mode (see 241-5701-415 *Passport 7400, 15000, 20000 Hunt Group Server Guide*), and forward the call request to that member. If the selected member's availability is sufficient to handle the call, the call will proceed as a normal Voice Networking call. If the selected member's availability is not sufficient to handle the call, the call will be redirected back to the hunt group server. On a redirected call request, the hunt group server selects the next hunt group member.

**Figure 35**
**Example of a Voice Networking call hunting**




## Implementation and configuration considerations

At least one Passport hunt group server must be present in the network to support call hunting. See 241-5701-415 *Passport 7400, 15000, 20000 Hunt Group Server Guide* for descriptions and configuration information for hunt group servers.

If the primary hunt group server cannot be reached or is out of service, the call can route through backup hunt group server. The backup hunt group server must be configured in the call redirection server (CRS). See 241-5701-410 *Passport 7400, 15000, 20000 Call Redirection Server Guide* for descriptions and configuration information for CRS.

The following points provide an overview to implementing call hunting:

1 Set up a primary hunt group server, with the DNAs of voice routes as hunt group members. This configuration enables call hunting among a set of destination voice routes to complete a given call. Also, specify the type of hunting algorithm to use: start from zero, rotary, or most available.

2 Optionally, set up a backup hunt group server with a member set that is identical to that on the primary server.

3 Specify the DNA address of the primary hunt group server in the dataNetworkAddress attribute under the Vncs DiallingPlan DirectoryNumber component.

4 Add a HuntGroupMember component under each VoiceRoute Dna component. Optionally, change the value of the usageDeltaUpdateThreshold attribute, which defines the threshold for availability of the member to both the primary and backup hunt group servers. Specify the DNS of the primary hunt group server; optionally, specify the DNS of the backup hunt group server. This DNS entry defines the location of the servers to which the voice route sends the AMP.

For configuration details and procedures, see “Configuring call hunting through a hunt group server” (page 88).

## Voice Networking call server configuration

By default, dynamic up- and down-speeding is enabled on MVP-E FPs when the negotiated encoding choice value is g711G726 or g726.

To prevent the clipping of voice traffic on MVP-E FPs when the negotiated value of the silenceSuppression attribute is on, ensure that the VoiceRoute Interface component’s speechHangoverTime attribute is set appropriately. You can also specify a cap on the amount of background or comfort noise to generate when silence suppression is operational. See “Voice routes” (page 176) for more information.

End-to-end negotiation determines whether to support FIS. FIS operates on MVP-E FPs when the negotiated modem/fax encoding choice is g711G726 or g726. To prevent the clipping of portions of facsimile transmissions on MVP-E FPs when the negotiated value of the faxIdleSuppressionG711G726

attribute is on, ensure that the faxHangoverTimeG711G726 attribute under the VoiceRoute Interface component is set appropriately. See “Voice routes” (page 176) for more information.

To operate, the negotiated modem/fax encoding choice value for modem/fax traffic must be g711, g711G726 or g726, the negotiated encoding choice value for fax traffic must be v29V27Relay or v17V29V27Relay, and the negotiated value of the v17EncodedAsG711G726 attribute must be yes.

The DTMF tones that can be lost are those produced after call establishment (for example, when accessing voice mail). The value on is the recommended setting for voice traffic using G.728 and G.729 encoding. For DTMF tone detection and regeneration to operate, end-to-end negotiation must result in a value of on.

## Voice networking call server components

You configure the voice networking call server by configuring the VoiceNetworkingCallServer, VoiceProfile, DiallingPlan, and DirectoryNumber components. The VoiceNetworkingCallServer component provides address resolution information to the PORS during switched voice call establishment by means of the DiallingPlan and DirectoryNumber components. The DirectoryNumber component contains attributes to define PORS-based routing using destination node and component identification or PORS-based routing with DNA support. The VoiceNetworkingCallServer component gathers audio handling parameters from the VoiceProfile component associated with particular DiallingPlan and DirectoryNumber component instance combinations. Before proceeding with configuration the attributes under the VoiceProfile component, see “Voice, modem, and facsimile encoding choices and rates” (page 168).

Each VoiceNetworkingCallServer component instance represents one numbering plan. That is, one way to organize and group dialed numbers to define routing (DiallingPlan component) and quality of service (VoiceProfile component) parameters.

A DN is a numeric prefix for one or more dialed numbers (excluding the PBX access code) that have the same destination. DNs allow you to efficiently organize groups of dialed numbers for particular destinations or geographic regions.

***Note:*** To prevent call failures on an MCDN-configured signaling channel that has anti-tromboning enabled, ensure that you specify the same destination for each Dn component instance and provision each node's own DNs in the VNCS database. That is, all Dn component instances on all Passport nodes in the network should be configured with identical data.

# Voice, modem, and facsimile encoding choices and rates

To successfully establish a call, the source and destination FPs must agree on how to encode audio traffic. Part of Voice Networking end-to-end negotiation involves the source and destination FPs exchanging voice profile configuration information. During the exchange, the FPs determine a mutually preferred and supported voice, modem/fax, and fax choice. A voice, modem/fax, and fax choice is the combination of the negotiated encoding choice and the encoding choice's negotiated minimum and maximum rates. Each choice corresponds to a traffic type—voice, modem/fax, and fax, respectively. You must, therefore, know how to properly provision the voiceEncodingChoice, voiceRates, modemFaxEncodingChoice, and modemFaxRates attributes.

The following sections provide information on how to provision the VoiceProfile component's encoding choice and rate attributes for voice, modem/fax, and fax traffic:


- "Considerations for voice, modem, and facsimile traffic configuration" (page 168)
- "VoiceEncodingChoice and voiceRates attributes" (page 170)
- "ModemFaxEncodingChoice and modemFaxRates attributes" (page 171)
- "Defining traffic handling options" (page 175)


## Considerations for voice, modem, and facsimile traffic configuration

To successfully establish a call,

- each FP must support a configured encoding choice and certain rates between the minimum and maximum inclusive

- a corresponding encoding feature, for certain encoding types, must be configured under the featureList attribute

In some cases, an MVP-E FP supports an encoding choice but modifies certain configured rate values. For example, MVP-E FPs support dynamic up- and down-speeding of audio traffic for dealing with congestion in the network. However, MVP-E FPs do not support all of the available ITU-T G.726 ADPCM voice and modem/fax encoding rates. The table "Relationship between FPs, encoding choices, and supported rates" (page 169) lists the available voice and modem/fax encoding choice and FP combinations, and the rates each combination supports.

For operational information about encoding types and compression rates for all traffic types, view the operational attributes under the VoiceSubroute SwitchedVoiceService Framer component's Negotiation group. The attributes in the Negotiation group indicate the results of end-to-end negotiation.

**Table 16**
**Relationship between FPs, encoding choices, and supported rates**

| **Voice encoding choice** | **FP type** | **Supported rate(s) in kbit/s** |
|---|---|---|
| g711Only | MVP-E | 64 |
| g711G726 | MVP-E | 24, 32, 64 |
| g728Only | MVP-E | 16 |
| g729Only | MVP-E | 8 |
| **Modem/fax encoding choice** | **FP type** | **Supported rate(s) in kbit/s** |
| g711Only | MVP-E | 64 |
| g711G726 | MVP-E | 32, 64 |
| | | |
| relay | MVP-E | 2.4, 4.8, 7.2, 9.6, 12.0, 14.4 |
| | | |

## VoiceEncodingChoice and voiceRates attributes

By configuring the voiceEncodingChoice attribute, you can set encoding preferences for voice traffic which uses a particular VoiceProfile component. When configuring the voiceEncodingChoice attribute, you determine the order and priority—first, second, and third—of the voice encoding choices to be used during end-to-end negotiation. During end-to-end negotiation, the source and destination FPs select a voice choice for use on voice traffic from the configured encoding choices and rates. In selecting a voice choice, the FPs remove

- unsupported encoding types, including those with unsupported rates between the minimum and maximum values
- an encoding choice whose corresponding feature is not present in the featureList attribute

  ***Note:*** It is possible to provision the voiceEncodingChoice attribute with an encoding choice which requires a corresponding feature and not provision the Software component's featureList attribute. Further, the system does not notify you of this omission when you check your configuration. Only during end-to-end negotiation are encoding feature omissions discovered. Therefore, to avoid potential call setup problems, you must provision the encoding choice's corresponding feature. See the table "Default and provisionable voice encoding rates for each voice encoding choice" (page 171).

The table "Default and provisionable voice encoding rates for each voice encoding choice" (page 171) contains the default and provisionable values for the minimum and maximum rates of the voiceRates attribute corresponding to each of the voiceEncodingChoice attribute's values. This table also lists whether an encoding choice requires you to provision a corresponding feature. As the table shows, only g711G726 encoding allows you to set the minimum and maximum rates to a value other than the default value. Thus, setting encoding choices for voice traffic requires very little configuration.

**Table 17**
**Default and provisionable voice encoding rates for each voice encoding choice**

| voiceEncoding-Choice attribute values | Feature required | Default voiceRates attribute rates (in kbit/s) | Provisionable voiceRates attribute rates (in kbit/s) |
|---|---|---|---|
| g711Only | N/A | min = 64<br>max = 64 | min = 64<br>max = 64 |
| g711G726 (see Note 1) | N/A | min = 24<br>max = 64 | min = 16, 24, 32, 64<br>max = 16, 24, 32, 64 |
| g728Only | g728 | min = 16<br>max = 16 | min = 16<br>max = 16 |
| g729Only | g729 | min = 8<br>max = 8 | min = 8<br>max = 8 |
| none (see Note 2) | N/A | N/A | N/A |
| ***Note 1:*** MVP-E FPs support g711G726 as a voice encoding choice, but only at the following rates: 24, 32, and 64 kbit/s.<br>***Note 2:*** The encoding choice none specifies the end of the encoding list. As well, none cannot appear as the first encoding choice, and no encoding choice other than none can appear after none. | | | |
| | | | |

## ModemFaxEncodingChoice and modemFaxRates attributes

You determine the order and priority of encoding choices for modem/fax and fax traffic that use a particular VoiceProfile component by setting the three elements—first, second, and third—of the modemFaxEncodingChoice attribute. Along with selecting a voice choice for use on voice traffic during end-to-end negotiation, the FPs select two modem/fax choices: one for use on modem/fax traffic and one for use on fax traffic. In selecting the modem/fax choices, the FPs remove

- unsupported encoding types, including those with unsupported rates between the minimum and maximum values

- an encoding choice whose corresponding feature is not present in the featureList attribute

  ***Note 1:*** It is possible to provision the modemFaxEncodingChoice attribute with an encoding choice which requires a corresponding feature and not provision the Software component's featureList attribute. Further, the system does not notify you of this omission when you check your configuration. Only during end-to-end negotiation are encoding feature omissions discovered. Therefore, to avoid potential call setup problems, you must provision the encoding choice's corresponding feature. See the table "Default and provisionable modem/fax encoding rates for each modem/fax encoding choice" (page 173).

  ***Note 2:*** Even if you do not plan to make modem and facsimile calls, you must still provision the modemFaxEncodingChoice and modemFaxRates attributes properly. If these attributes are not set properly, all audio calls fail during end-to-end negotiation.

The table "Default and provisionable modem/fax encoding rates for each modem/fax encoding choice" (page 173) contains the default and provisionable values for the minimum and maximum rates of the modemFaxRates attribute corresponding to each of the modemFaxEncodingChoice attribute's values. The table also lists whether an encoding choice requires you to provision a corresponding feature. As the table shows, only the value g711G726 allows you to set both the minimum and maximum rates to a value other than the default value. The value relay only allows you to set the maximum rate to a value other than the default value.

See the following sections for more information on configuring modem/fax encoding choices:

- "Encoding modem/fax and fax traffic" (page 173)
- "Considerations for fax relay configuration" (page 174)
- "Using the value none" (page 175)

**Table 18**
**Default and provisionable modem/fax encoding rates for each modem/fax encoding choice**

| modemFax-EncodingChoice attribute values | Feature required | Default modemFaxRates attribute rates (in kbit/s) | Provisionable modemFaxRates attribute rates (in kbit/s) |
|---|---|---|---|
| g711Only | N/A | min = 64<br>max = 64 | min = 64<br>max = 64 |
| g711G726 (see Note 1) | N/A | min = 32<br>max = 64 | min = 16, 24, 32, 64<br>max = 16, 24, 32, 64 |
| relay (see Note 2) | faxRelay | min = 0.0<br>max = 14.4 | min = 0.0<br>max = 2.4, 4.8, 7.2, 9.6, 12.0, 14.4 |
| none (see Note 3) | N/A | N/A | N/A |
| ***Note 1:*** MVP-E FPs support g711G726 as a modem/fax encoding choice, but only at the following rates: 32 and 64 kbit/s.<br>***Note 2:*** The minimum fax relay rate of 0.0 kbit/s is fixed.<br>***Note 3:*** The encoding choice none signifies the end of the encoding list. See "Using the value none" (page 175) for more information on setting the three elements of the modemFaxEncodingChoice attribute to none, none, and none. | | | |
| | | | |

## Encoding modem/fax and fax traffic

Once end-to-end negotiation determines the preferred voice choice to use on voice traffic, the two highest remaining preferred modem/fax choices combine to determine the encoding of modem/fax and fax traffic. The table "Modem/fax encoding choice combinations and traffic type matrix" (page 174) describes all possible modem/fax choice combinations and the negotiated encoding choice for each traffic type.

**Table 19**
**Modem/fax encoding choice combinations and traffic type matrix**

| Modem/fax choices remaining (after end-to-end negotiation) | | Traffic type and possible negotiated encoding values | |
|---|---|---|---|
| **First choice** | **Second choice** | **Modem/fax traffic** | **Fax traffic** |
| relay | none | v29V27Relay or v17V29V27Relay (see Note 1) | v29V27Relay or v17V29V27Relay |
| relay | g711Only | g711 | v29V27Relay or v17V29V27Relay |
| relay | g711G726 | g711G726, g726 | v29V27Relay or v17V29V27Relay |
| g711Only | --- (see Note 2) | g711 | g711 |
| g711G726 | --- (see Note 2) | g711G726, g726 | g711G726, g726 |
| none | none (see Note 3) | See “Using the value none” (page 175) | |

***Note 1:*** The use of relay for modem/fax traffic means that modem calls are not supported (see “Considerations for fax relay configuration” (page 174)).

***Note 2:*** If either g711Only or g711G726 is the first modem/fax encoding choice, then it is used for both modem/fax and fax traffic. Voice Networking ignores the second modem/fax choice.

***Note 3:*** End-to-end negotiation can result in the modification of the remaining modem/fax choices to none and none (different from setting the modemFaxEncodingChoice attribute to none and none). In this situation—having no remaining supported modem/fax encoding choices—the call is rejected.

## Considerations for fax relay configuration

Fax relay encoding applies only to fax traffic. If you need to support modem calls and want to use fax relay encoding, you must provision the modemFaxEncodingChoice attribute with the value relay and a modem/fax encoding choice for modem/fax traffic. Use the information in the table “Modem/fax encoding choice combinations and traffic type matrix” (page 174) as a guideline.

## Using the value none

By setting all three elements of a VoiceProfile component's modemFaxEncodingChoice attribute to the value none,

- all audio traffic is treated as voice traffic. In other words, Voice Networking ignores modem/fax and fax traffic
- bandwidth measurements are simplified as all traffic uses a fixed rate
- voice, modem, and facsimile calls proceed as long as the FPs negotiate g711G726 as the voice choice for voice traffic

However, setting all three elements of the modemFaxEncodingChoice attribute to none is not the same as having no supported modem/fax encoding choices during end-to-end negotiation. The latter instance—no supported modem/fax encoding choices—results in call failure.

## Defining traffic handling options

When configuring voice and modem/fax encoding choices, you can simplify the configuration process by categorizing and defining how to handle voice, modem/fax, and fax traffic. The table "Available Voice Networking traffic handling options" (page 176) describes the categorization options available and the FPs that support each option. If necessary, see also the table "Modem/fax encoding choice combinations and traffic type matrix" (page 174) for more information.

The difference between each of the four possible traffic handling options depends on the setting of the modemFaxEncodingChoice attribute. Since all audio calls begin as voice traffic, end-to-end negotiation must result in the selection of a mutually preferred and supported voice encoding choice. Otherwise, any audio call—voice, modem, or facsimile—fails.

**Table 20**
**Available Voice Networking traffic handling options**

| Option | FP type | Configuration details |
|---|---|---|
| Support all call types and encode fax traffic using fax relay | MVP-E only | set Vncs/# Vp/# modemFaxEncodingChoice first relay<br>set Vncs/# Vp/# modemFaxEncodingChoice second <g711Only or g711G726> |
| Support all call types and encode modem/fax and fax traffic using g711G726 or g711Only | MVP-E | set Vncs/# Vp/# modemFaxEncodingChoice first <g711Only or g711G726> |
| Support only voice and facsimile calls and encode modem/fax and fax traffic using relay | MVP-E only | set Vncs/# Vp/# modemFaxEncodingChoice first relay<br>set Vncs/# Vp/# modemFaxEncodingChoice second none<br>set Vncs/# Vp/# modemFaxEncodingChoice third none |
| Treat all traffic as voice traffic (see “Using the value none” (page 175) for more information) | MVP-E | set Vncs/# Vp/# modemFaxEncodingChoice first none<br>set Vncs/# Vp/# modemFaxEncodingChoice second none<br>set Vncs/# Vp/# modemFaxEncodingChoice third none |
| | | |

# Voice routes

The VoiceRoute component controls from 1 to 16 VoiceSubroute components.

## Configuring the voice route and DNA

By default, the diallingPlan0 attribute contains all of the available Passport values corresponding to the various supported PBX type of number (TON) and numbering plan indicator (NPI) combinations.

Each diallingPlan attribute (dp0, dp1, and dp2) corresponds to a like-numbered DiallingPlan component (DP/0, DP/1, and DP/2) under the VoiceNetworkingCallServer component. You can thus organize numbers

according to type or customer group using the diallingPlan attributes and define how to handle each group using the DirectoryNumber subcomponents under each DiallingPlan component.

In some cases, the signaling protocol defines how to configure the diallingPlan attributes. For example, the MCDN signaling protocol supports specific types of numbers according to the 3 supported Meridian 1 numbering plans: Unknown, E.164, and Private. See the table “Supported MCDN numbering plan and type of number combinations” (page 177)). For MCDN, you map a particular NPI and TON combination—corresponding to a particular Passport value—under a specific diallingPlan attribute instance: dp0, dp1, or dp2.

As shown in the table “Supported MCDN numbering plan and type of number combinations” (page 177), coordinated dialling plan (CDP) numbers map to the diallingPlan0 attribute. Therefore, you assign and organize dialed numbers that belong to the CDP type to DirectoryNumber subcomponents under the DiallingPlan/0 component of the appropriate VoiceNetworkingCallServer component instance.

**Table 21**
**Supported MCDN numbering plan and type of number combinations**

| Numbering plan | Type of number | Passport value | diallingPlan attribute |
|---|---|---|---|
| Unknown (0000) | Unknown (000) | unknown | diallingPlan0 |
| Private (1001) | Coordinated dialling plan or CDP (110) | p6 | |
| E.164 (0001) | Local/subscriber | subscriber | diallingPlan1 |
| E.164 (0001) | International | international | diallingPlan2 |
| E.164 (0001) | National | national | |
| Private (1001) | ESN special purpose number or SPN (011) | p3 | |
| Private (1001) | ESN location code or LOC (101) | p5 | |
| | | | |

## Configuring the voice route interface

By default, echo cancellation is enabled on MVP-E FPs.

### Tandem Pass Through

Tandem pass through applies to voice traffic on MVP-E FPs.

For TPT to operate,

- you must add tandemPassThrough to the Software component's featureList attribute on all candidate nodes
- you must set the tandemPassThrough attribute to enabled on both tandem nodes and the value of the negotiatedTandemPassThrough operational attribute must also be enabled
- the voiceEncoding attribute on the tandem nodes must be configured with the same compression algorithm. When the negotiated voice encoding algorithm is g726 or g711G726 and the encoding rate is 24 or 32 kbit/s, the value of the operational attribute currentEncodingRate can be different on the tandem Passport nodes, if you configured dynamic up- and down-speeding.
- the intermediate PBX and the tandem nodes cannot alter the compressed voice data. Therefore, the PBX and the tandem nodes cannot apply audio gain or loss pads, use external echo cancellers, perform compander law conversion, or use bit 7 stuffing for line coding (when setting the zeroCoding attribute).

TPT also introduces a certain amount of delay into the end-to-end connection. You can compensate for the delay by adjusting the jitter buffer on the egress node. You adjust the jitter buffer by configuring the insertedOutputDelay attribute under the VoiceProfile component (see "Voice Networking call server configuration" (page 166)).

## Voice subroute

Each VoiceSubroute component controls a group of channel and timeslot pairs, associated with a particular logical processor and E1 or DS1 port combination, by means of SwitchedVoiceService components.

Each SwitchedVoiceService component has a dynamic LogicalConnection operational component that uses the PORS parameters you provision under the VoiceNetworkingCallServer VoiceProfile component to set up and control a Voice Networking connection.

Each VoiceSubroute component interfaces to one SignallingChannel component and one VoiceRoute component instance.

The VoiceSubroute component manages all aspects of the call establishment and tear-down processes by communicating with the SignallingChannel component and using the parameters you specify under the VoiceRoute and VoiceNetworkingCallServer components and subcomponents.

# Appendix A
# Compliance with standards—Voice Networking signaling protocols

The following sections contain compliance information for both the CCS- and CAS-based signaling protocols supported by Voice Networking:


- "Compliance statement for ETSI QSIG" (page 181)
- "Compliance statement for NIS A211-1" (page 182)
- "Compliance statement for MCDN" (page 183)
- "Compliance statement for Euro-ISDN" (page 184)
- "Compliance statements for CAS" (page 186)


## Compliance statement for ETSI QSIG

Voice Networking's implementation of the ETSI QSIG signaling protocol complies, in general, with the ETSI QSIG standards listed in the table "Voice Networking's ETSI QSIG compliance with standards" (page 182).

***Note:*** The name identification, transit counter, call transfer, call diversion, path replacement, call offer, call completion, called/busy number identification, message waiting indication, and centralized voice mail pertain to transit Private Integrated Services Network Exchanges (PINX) only.

**Table 22**
**Voice Networking's ETSI QSIG compliance with standards**

| Standard | Details |
|---|---|
| ETS 300 172, 3rd Edition | Basic call (call related APDU transport, call related notification transport, connection oriented call independent APDU transport) |
| ETS 300 239, 3rd Edition | Generic functional procedures |
| ETS 300 172, Annex ZA (1995) | Segmentation and reassembly |
| ETS 300 239, 2nd Edition | Transport of manufacturer specific information |
| ETS 300 238, 2nd Edition | Name identification |
| EN 301 048, V1.1.1 | Transit counter |
| ETS 300 261, 2nd Edition | Call transfer |
| ETS 300 257, 2nd Edition | Call diversion |
| ETS 300 259, 2nd Edition | Path replacement |
| ETS 300 362, 2nd Edition | Call offer |
| ETS 300 366, 2nd Edition | Call completion |
| ETS 300 239, 2nd Edition | Called/Busy number identification |
| ETS 301 360/255, 2nd Edition | Message waiting indication |
| ETS 300 257, ETS 300 239, ETS 301 255 (2nd Edition) | Centralized voice mail |
| | |

# Compliance statement for NIS A211-1

Voice Networking's implementation of the NIS A211-1 signaling protocol complies, in general, with the standards listed in the table "Voice Networking's NIS A211-1 compliance with standards" (page 183). This includes support for

- basic call using 64 kbit/s bearer channels
- facility associated signaling (23 B + D)

- listed services applying to call associated signaling
- Layer 1, 2 and 3: maintenance messages, signaling and line requirements for the preceding items

**Table 23**
**Voice Networking's NIS A211-1 compliance with standards**

| Standard | Details |
|---|---|
| NIS A211-1, Standard Release 6, (1994-03) | Basic call control, link and channel maintenance messaging, called and calling number transport |
| NIS A211-1, Standard Release 6, (1994-03) | 56 kbit/s clear data call transport |
| NIS A211-1, Standard Release 6, (1994-03) | Calling, connected party number display (presentation and restriction) |
| NIS A211-1, Standard Release 6, (1994-03) | Network name display (presentation and restriction) |
| NIS A211-1, Standard Release 6, (1994-03) | Network redirection and reason |
| NIS A211-1, Standard Release 6, (1994-03) | Special number services |
| NIS A211-1, Standard Release 6, (1994-03) | Federal Communications Commission (FCC) Equal Access |
| NIS A211-1, Standard Release 6, (1994-03) | Integrated services access |
| | |

## Compliance statement for MCDN

Voice Networking support of the Meridian Customer Defined Networking (MCDN) Peer-to-Peer signaling protocol conforms in general to the capabilities available as of Release 23 for Meridian 1. Voice Networking does not support all of the capabilities supported by MCDN and the Meridian 1 PBXs. Voice Networking's implementation of MCDN includes the following:

- basic call support, as defined in MCDN Release 1 (1988), TR-88-5003-R/M/S, TR-87-0041-R/M/S, TR-88-5505-S

- called, calling, and connected number support, as defined in MCDN Release 1 (1988), TR-87-0045-R/S, TR-88-0047-R/M/S
- subaddressing support
- channel negotiation and overlap dialing support
- maintenance messaging support
- connection-oriented feature transport support (part of NAS development)
- connectionless feature transport support, as defined in MCDN Release 1 (1988), PS-87-0010,and including network ring again, as defined in TR-87-0041-R/M/S
- redirection information support, as defined in TR-88-5004-R/M/S

***Note:*** Because of the large number of feature specifications and development enhancements for Meridian 1 PBXs, it is not possible or feasible to list them all.

See the table “Supported MCDN basic call and supplementary services” (page 119) for the MCDN basic call and supplementary capabilities supported by Voice Networking.

## Compliance statement for Euro-ISDN

Voice NetworkSing’s implementation of the Euro-ISDN signaling protocol complies, in general, with the ETSI and ITU-T standards described in the table “Voice Networking’s Euro-ISDN compliance with standards” (page 185).

**Table 24**
**Voice Networking's Euro-ISDN compliance with standards**

| Standard | Details |
|---|---|
| ITU-T Q.931, (1993-03) | Digital subscriber system number one (DSS1); ISDN UNI Layer 3 specification for basic call control |
| ETSI EN 300 403-1, V.1.2.2 (1998-04)<br>***Note:*** Includes support of ETS 300 102-1 (1990). | ISDN; DSS1; Signaling network layer for circuit-mode basic call control |
| ETSI EN 300 196-1, V.1.2.2 (1998-04) | ISDN; DSS1; Generic functional protocol for the support of supplementary services |
| ETSI ETS 300 138-1, Second edition (1997-05) | ISDN; DSS1; Closed user group (CUG) supplementary service |
| ETSI ETS 300 092-1, (1992-03) | ISDN; DSS1; Calling line identification presentation (CLIP) supplementary service |
| ETSI ETS 300 093-1, (1992-03) | ISDN; DSS1; Calling line identification restriction (CLIR) supplementary service |
| ETSI ETS 300 064-1, Second edition (1996-09) | ISDN; DSS1; Direct dialing in (DDI) supplementary service |
| ETSI ETS 300 061-1, (1991-10) | ISDN; DSS1; Subaddress (SUB) supplementary service |
| ETSI ETS 300 097-1, (1992-05) | ISDN; DSS1; Connected line identification presentation (COLP) supplementary service |
| ETSI ETS 300 098-1, (1992-05) | ISDN; DSS1; Connected line identification restriction (COLR) supplementary service |
| ETSI ETS 300 286-1, (1996-02) | ISDN; DSS1; User-to-user signaling (UUS) supplementary service |
| ETSI ETS 300 130-1, (1992-05) | ISDN; DSS1; Malicious call identification (MCID) supplementary service |
| ETSI ETS 300 207-1, (1994-12) | ISDN; DSS1; Diversion supplementary services, including call forwarding busy (CFB), call forwarding no reply (CFNR), call forwarding unconditional (CFU), and call deflection (CD) |
| (Sheet 1 of 2) | |

**Table 24 (continued)**
**Voice Networking's Euro-ISDN compliance with standards**

| Standard | Details |
|---|---|
| ETSI ETS 300 182-1, (1993-04) | ISDN; DSS1; Advice of charge (includes AOC-D and AOC-E) supplementary service |
| ETSI ETS 300 185-1, (1993-04) | ISDN; DSS1; Conference call, add-on (CONF) supplementary service |
| ETSI ETS 300 359-1, (1995-11) | ISDN; DSS1; Completion of calls to busy subscriber (CCBS) supplementary service |
| ETSI EN 301 065-1, V.1.1.1 (1997-12) | ISDN; DSS1; Completion of calls on no reply (CCNR) supplementary service |
| ETSI ETS 300 210-1, (1996-02) | ISDN; DSS1; Freephone (FPH) supplementary service |
| ETSI ETS 300 745-1, (1997-07) | ISDN; DSS1; Message waiting indicator (MWI) supplementary service |
| ETSI EG 201 189-1, V.1.2.1 (1997-07) | ISDN; DSS1; Master list of codepoints and operation values |
| (Sheet 2 of 2) | |

# Compliance statements for CAS

The following describe how Passport Voice Networking conforms to specific channel associated signaling (CAS) standards as they relate to each interface:

- DS1 CAS—Passport Voice Networking supports A and B bit line signaling based on the ANSI/EIA/TIA-464A standard.
- E1 CAS—Passport Voice Networking supports A,B,C, and D bit line signaling based on ITU Recommendations Q.421 and Q.422.
- J2 CAS (for TTC2M links)—Passport Voice Networking supports A bit line signaling based on TTC Standards, specifically sections JJ-20-10, JJ-20-11 and JJ-20-12.

# Appendix B
# Network migration considerations

The following sections contain information which describe how to migrate, or upgrade, a Passport Voice Transport permanent logical connection or PLC-based network to a Passport Voice Networking switched virtual circuit or SVC-based network:


- "Upgrade options" (page 187)
- "Preliminary considerations" (page 188)
- "Mixing PLCs and VoiceNetworking" (page 191)
- "Upgrading to Voice Networking" (page 194)
- "Inserting Voice Networking into a PBX network" (page 205)


The advantage of this upgrade is that the Voice Networking service more closely maps the reserved bandwidth to the actual bandwidth used.

## Upgrade options

A Voice Transport network can be upgraded in the following ways:

- upgrade all Passport nodes to Voice Networking with full capability. This is the preferred option, especially where the PBX protocol is supported by Voice Networking.

  ***Note:*** In cases where the PBX protocol is not supported by Voice Networking, CAS signaling (the lowest common denominator in most PBXs) can be used—although the extended features that only CCS signaling can provide, such as Call Display, may be lost.

- upgrade all Passport 7400 nodes to Voice Networking but only use the fixed end point mode. This option smooths the migration to full Voice Networking capability.
- only load Voice Networking onto Passport nodes that are added to the network. This option leaves the original PLC-based network intact.

## Preliminary considerations

During a call, the PBX to PBX link through the Passport subnet has fixed end points. This applies regardless of the form of voice or data call. In addition, the Passport nodes only create the path when the connection is needed. The path may change during the call based on path bumping and optimization; although not common, both operations do occur to make the best use of system resources.

For a PLC, the destination never changes. Whereas for an SVC connection, the destination changes on a call by call basis.

> ***Note:*** Throughout this chapter, the term "PLC" describes a group of PLC connections (rather than a single PLC connection) between two PBXs across a Passport subnet.

The difference between a Voice Transport PLC-based call and a Voice Networking switched-call only shows up when the call completes, as follows:

- A PLC behaves as a point to point connection, where the end points cannot change the path of their interconnecting link (unless they are manually reconfigured).

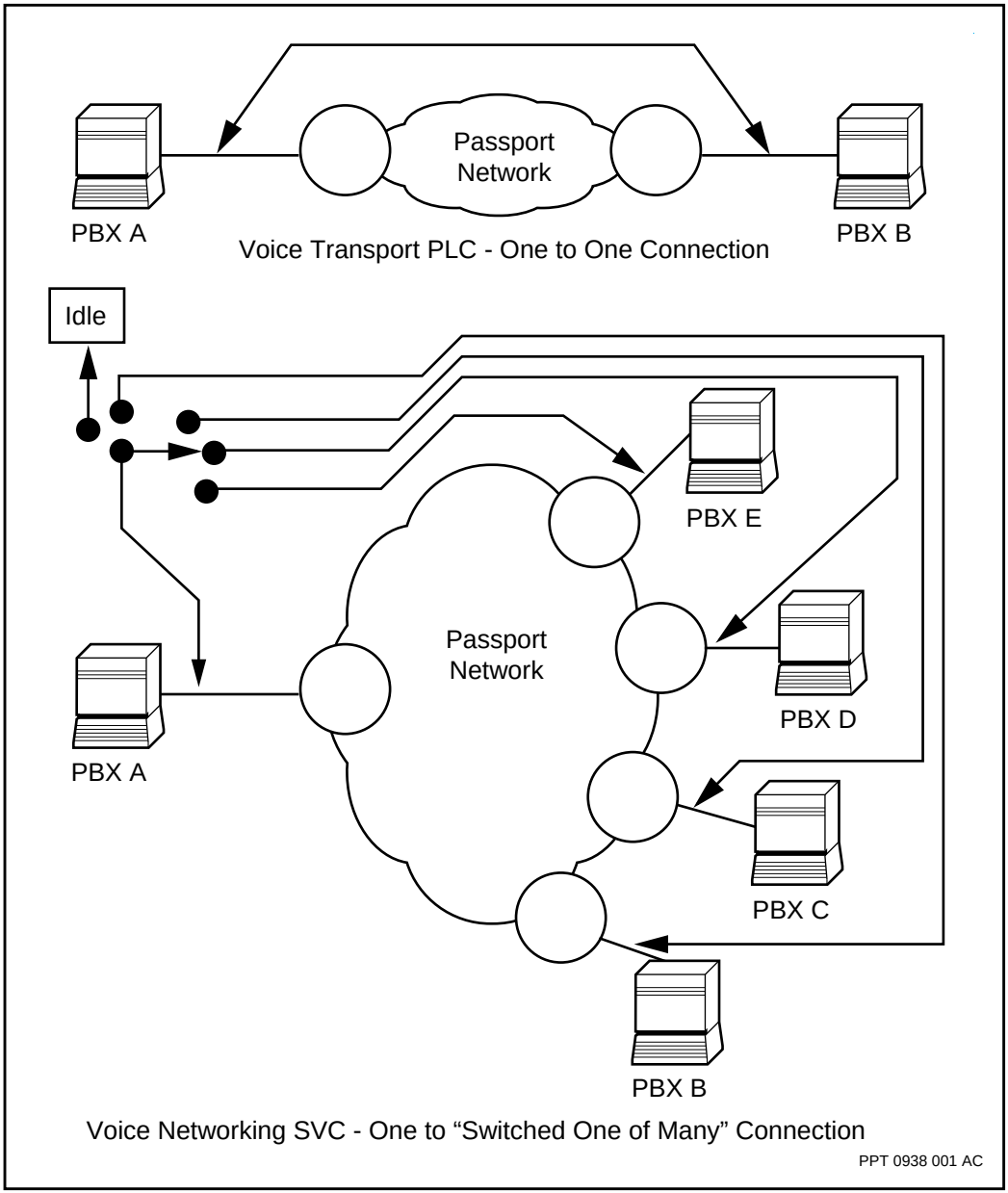
  When PBX A on Passport node A terminates a call to PBX B on Passport node B, clearing the call has no effect on the interconnecting link; the link remains in place and its bandwidth reservation remains in effect.

- A Voice Networking SVC behaves as a point to multipoint connection, where the end points are dynamically linked at the start of a call.

  When PBX A on Passport node A terminates a call to PBX B on Passport node B, clearing the call clears the interconnecting link; the link is set to idle and its 'reserved' bandwidth is made available for active links.

The figure “Voice Transport PLC versus Voice Networking SVC” (page 190) illustrates the difference between a Voice Transport PLC-based call and a Voice Networking switched-call. A Voice Transport PLC is a fixed point to point connection. Whereas a Voice Networking SVC is a point to multipoint connection; the end points are selected as part of the setup, and disassociated after clearing.

**Figure 36**
**Voice Transport PLC versus Voice Networking SVC**



## Mixing PLCs and VoiceNetworking

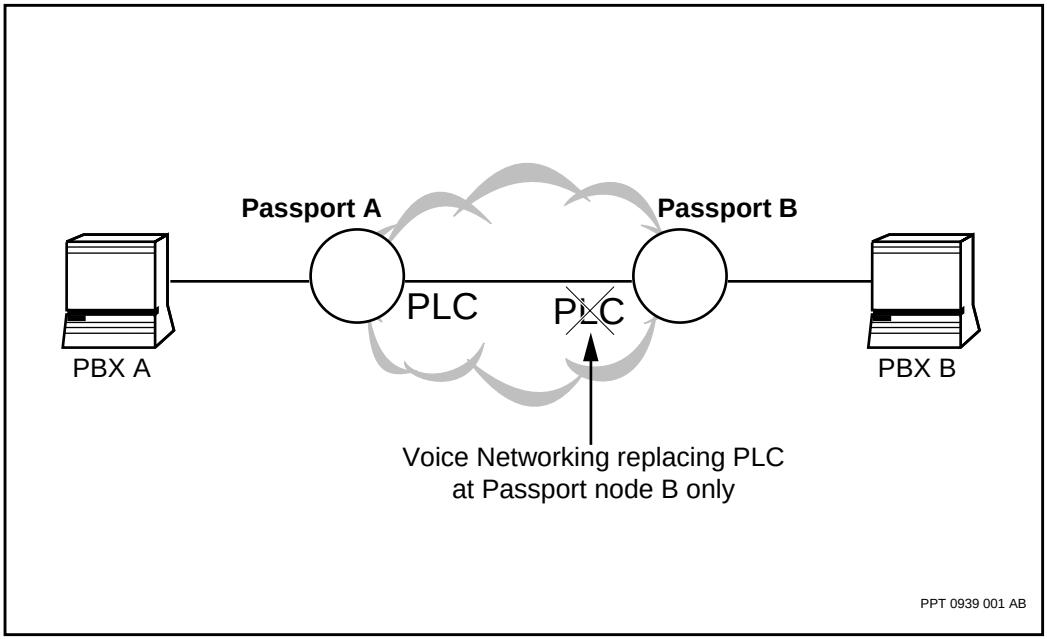
Voice Networking is incompatible with other Passport voice services on a PBX link by PBX link basis.

> ***Note:*** In the figures that follow, the connection between two Passport nodes by means of PLCs or SVCs are shown as a single path. This is a simplification of the “end point view”.

If a PLC-based connection exists between Passport node A and Passport node B, the PLC connection cannot be replaced at Passport node B alone. See the figure “Example of incompatible configurations” (page 192). The PLC connection at Passport node A must also be replaced or no calls can traverse the subnet, since the PLC at Passport node A neither analyses nor provides digits.

In the figure “Example of incompatible configurations” (page 192), Voice Networking at Passport node B requires digit information to complete connections for the B-channel.

**Figure 37**
**Example of incompatible configurations**

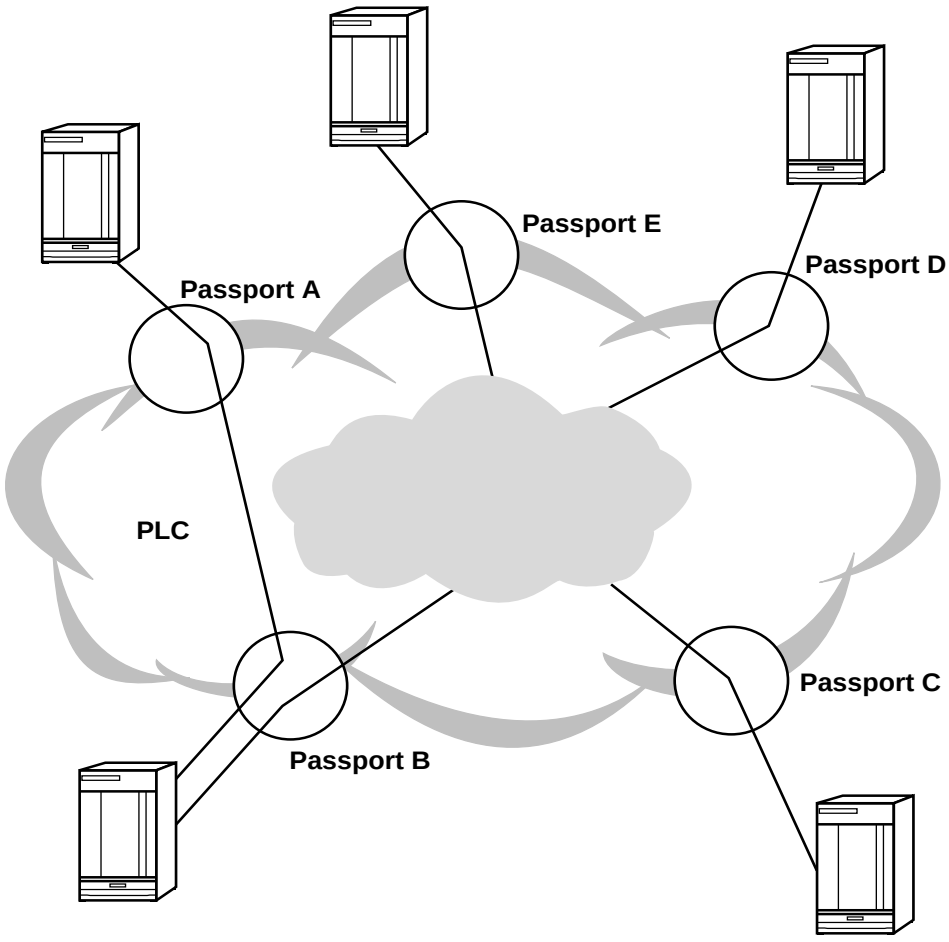



However, if Passport nodes A and B are interconnected by a PLC, as shown in the figure “Example of mixed configurations” (page 193), and Passport nodes B and C are interconnected by Voice Networking, calls can pass from Passport node C to A by way of B.

In the figure “Example of mixed configurations” (page 193), Passport nodes B, C, D and E are connected by Voice Networking. Regardless of the route through the subnet, a single call only involves two PBXs–the originating and the terminating ones.

Passport nodes A and B are connected by PLCs. Therefore, a call from Passport node C that is destined for Passport node A, must tandem through Passport node B to reach A. In this case, three PBXs are involved.

**Figure 38**
**Example of mixed configurations**




(The cloud illustrates that the path of the call is independent of the originating and terminating nodes. Between the end points, the path can vary widely from call to call.)

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# Upgrading to Voice Networking

This section describes how Voice Networking can be inserted into an existing Voice Transport network that is based on point-to-point, PLC connections. The following two scenarios are described:

- upgrading a Passport network on a basis of easy changes first
- dropping a Passport network into a private network

In each scenario the method is identical but the deciding criteria are different. See "Terminology" (page 194) for information on the terms used in the following section. The process of inserting Voice Networking is described in "Methodology" (page 196). See "Migration: example one" (page 197) and "Migration: example two" (page 206) for details.

## Terminology

The following terms are used throughout this section:

- access path (stub) – interconnects two PBX/Passport nodes. Where one Passport node/PBX can have multiple PLCs to other destinations in the subnet and the other Passport node/PBX has only one PLC to the subnet.

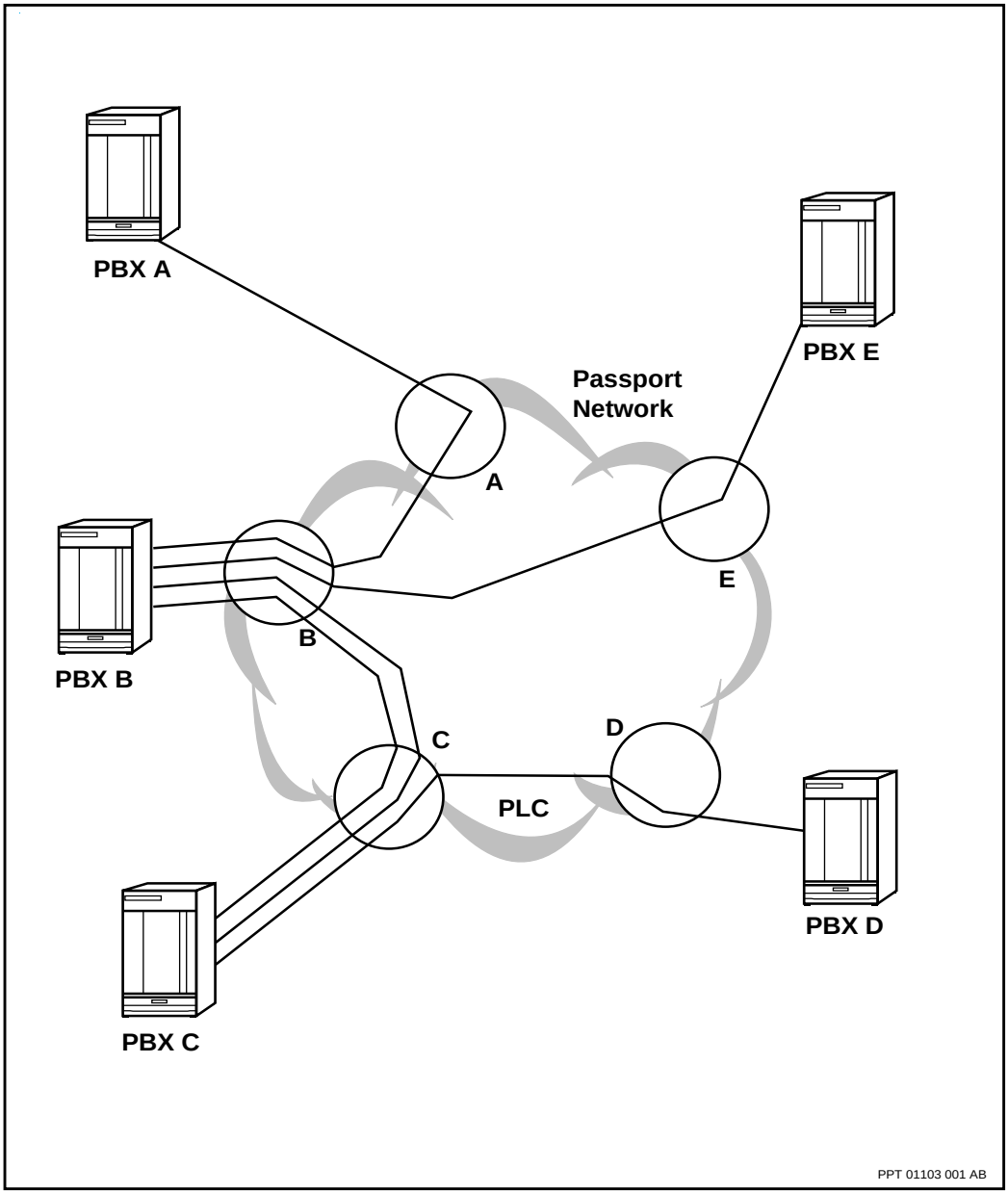
  An access path is shown in the figure "Example of access paths and high volume path" (page 195), interconnecting PBX/Passport node D and PBX/Passport node C.

- high volume path – provides tandem call capability for several PLCs. That is, two adjacent PBXs share PLCs for communication. Other adjacent PBXs can also use it by calling through the tandem PBX.

  A high volume path is shown in the figure "Example of access paths and high volume path" (page 195), interconnecting PBX/Passport node B and PBX/Passport node C.

In the figure "Example of access paths and high volume path" (page 195), PBXs A, D and E are single point of entry stubs; they have a single connection to the Passport subnet. PBXs B and C provide a high volume path, as D and C reach all other PBXs through the C to B link. Similarly, they are reached by the other three PBXs through the B to C link.

**Figure 39**
**Example of access paths and high volume path**



## Methodology

Voice Networking can be inserted into an existing Voice Transport network that is based on point to point PLC connections.

**1** Create the initial switched voice PBX-to-Passport node-to-PBX link(s). This can be either a single link or a small number of links replacing PLCs at carefully selected and non-adjoining nodes.

- Preferred links are either the highest volume path through the network, or single point of entry stubs connecting an isolated PBX to the network. They can be addressed in series (end to end path installation) or parallel (do all discrete paths first).
  Group the discrete links into a VoiceRoute. For example, in the figure "Example of access paths and high volume path" (page 195), group the two links between PBXs B and C under a single VoiceRoute component.
- At the selected Passport node to PBX connections add an FP link as a Voice Networking SVC.
- Provision the Passport node for Voice Networking on the new logical link. Set up the SVC using the overrideDirectoryNumber attribute. All calls to this VoiceRoute for the local PBX are then routed to a specific endpoint regardless of the dialed number. The endpoint is defined in the voice networking call server (VNCS) database.
- Optionally, configure the VNCS numbering plan to destination PBX data. This step can be left until the network is configured more completely. For an example of a completed list of paths, see the table "Passport node VNCS database programming for migration example one" (page 204).
- Provision the PBX to access the Voice Networking link (single E1 or DS1 on the PBX shelves).

**2** Prove-in the SVC.

**3** Disable and remove the PLCs and the E1 / DS1 link between the PBX and the Passport node.

**4** Repeat steps 1 to 3 with other PLC links until the network has been re-engineered.

**5** If not yet created (see bullet in step 1), create the VNCS data and download it to all Passport nodes.

**6** Monitor the results of the VNCS mapping to confirm that calls terminate correctly. See the table "Passport node VNCS database programming for

migration example one" (page 204) for an example of a completed list of paths.

**7** On a node by node basis, remove values from the overrideDirectoryNumber attribute for the selected PBX-to-Passport node links. This allows calls originating on the selected PBX-to-Passport node links to switch throughout the subnet.

**8** Optionally, optimize the Passport node-to-PBX links to match traffic volumes by removing redundant DS1 or E1 links. That is, if a PBX to Passport node link has four E1 cards (providing a total of 120 time slots) but has a peak volume of 75 calls, reduce the E1 card count to three (for a total of 90 time slots).

## Migration: example one

The figure "Initial Passport ISDN network - using PLCs" (page 198) illustrates a network of five PBXs. One Passport node has three PLC links, another has two. The remaining Passport nodes have only one PLC link.

Logically, a Voice Networking connection to any single link PBX (such as London) behaves like a PLC at the stub end. Changing one or more of the Ottawa / London / Melbourne links is simple, with the stub end programming identical at all phases of the change over.

**Figure 40**
**Initial Passport ISDN network - using PLCs**

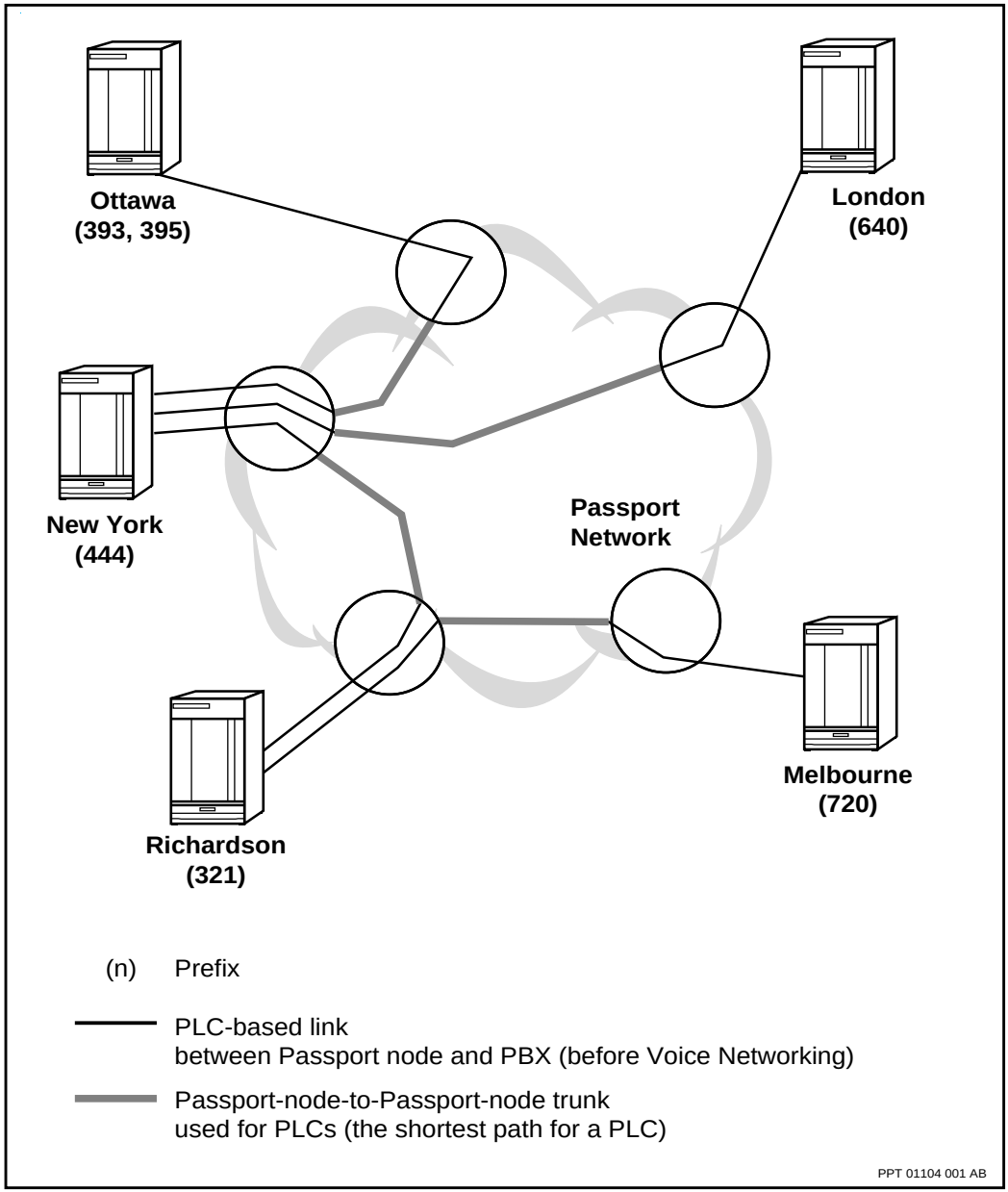
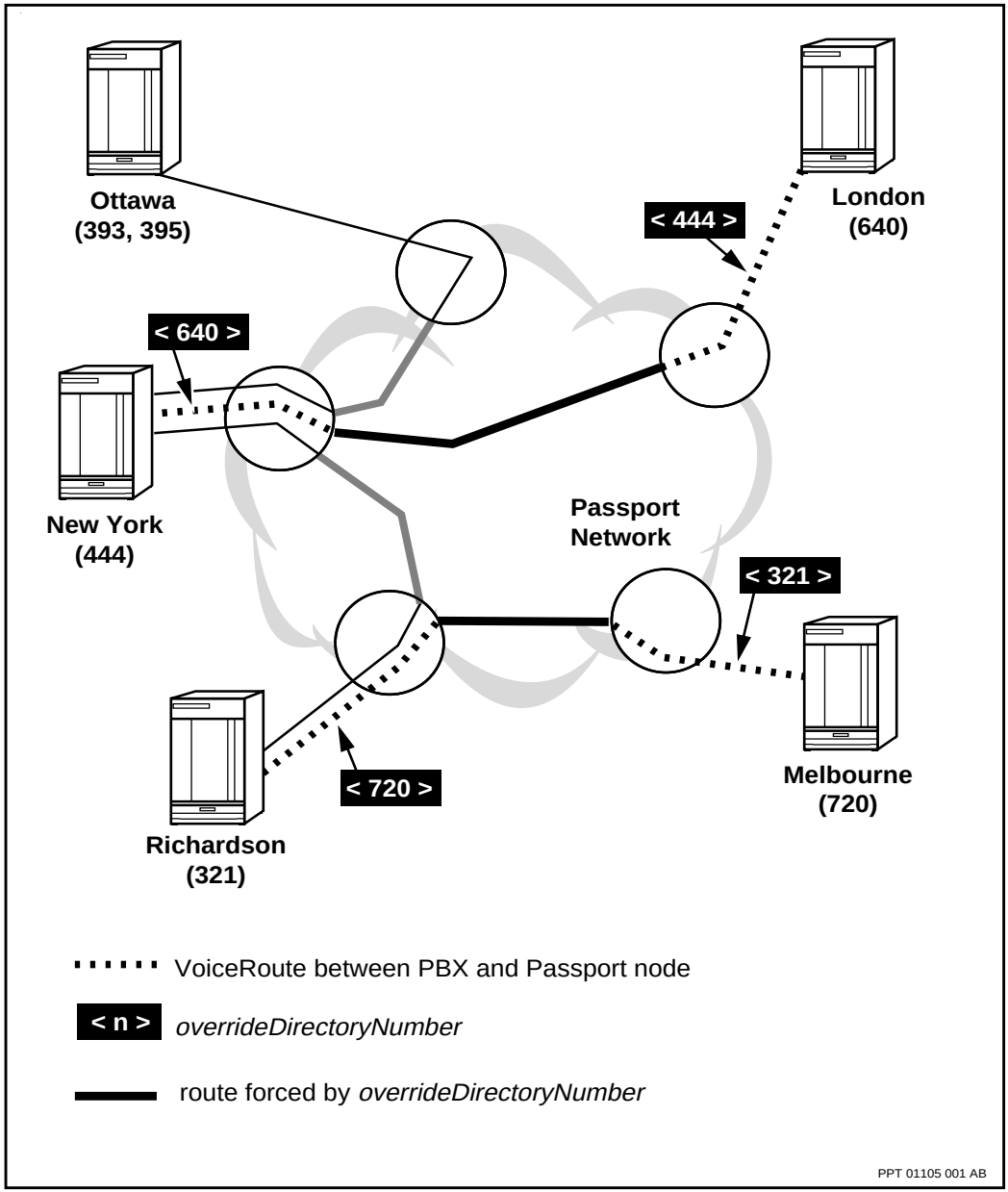


**Figure 41**
**Conversion of Passport ISDN network**



The following is an example of upgrading a network on a basis of "easy changes" first:

1. Initially, London and Melbourne will be migrated to Voice Networking (see the figure "Stage 2 of conversion of Passport ISDN network" (page 201)). Add Voice Networking switched links–parallel to the PLCs that already connect each Passport node with their associated PBX.

2. In New York, London, Richardson and Melbourne create the VNCS database shown in the table "VNCS database" (page 200).

**Table 25**
**VNCS database**

| Prefix | Destination | NodeId | Route | VoiceProfile |
|---|---|---|---|---|
| 444 | New York | 1021 | VRoute/x | 2 |
| 321 | Richardson | 1019 | VRoute/x | 2 |
| 720 | Melbourne | 32 | VRoute/x | 1 |
| 640 | London | 97 | VRoute/x | 1 |
| | | | | |

3. Using the overrideDirectoryNumber attribute in the VoiceRoute component, force all calls originating in Melbourne to be routed to Richardson and all calls originating in London to be routed to New York.

4. Prove-in the links and de-commission the PLCs (New York-London and Richardson-Melbourne) at the Voice Networking sites.

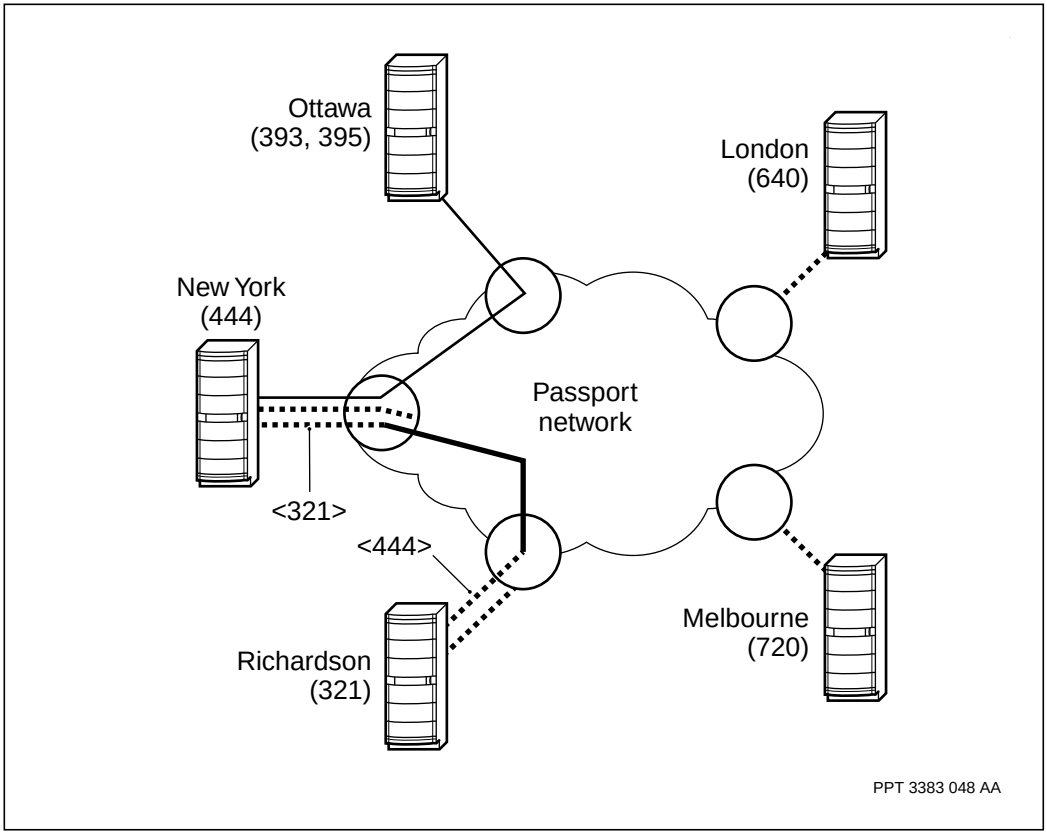
   There is no physical change to the network topology. However, the "Richmond-Melbourne" and "New York-London" voice connections are now dynamic–lasting only as long as a call is active.

   The initial links are now in. Since all other links involve New York, it is probable that due to shelf space constraints only one link can be converted at a time. In addition, should the provisioner be updating VNCS data during the installation of links, the risks of error are higher when configuring two links at the New York site at once.

5. Assume that the New York to Richardson link is desired next, to allow maximum intercontinental savings. See the figure "Stage 2 of conversion of Passport ISDN network" (page 201). Duplicate the New York to Richardson PLC link and program the nodes to use the new switched links as though they were PLC connections (using overrideDirectoryNumber). Note that the VNCS database in the table "VNCS database" (page 200)

does not need to be updated as it already contains the prefixes for New York and Richardson.

**Figure 42**
**Stage 2 of conversion of Passport ISDN network**




**6** Prove-in the switched voice links.

**7** Decommission and remove the PLC links.

**8** Duplicate the New York-Ottawa link (the only remaining link). Update the VNCS database and set the overrideDirectoryNumber accordingly. See the table “VNCS database” (page 200).

**9** Prove-in the switched voice links.

**10** Decommission and remove the New York-Ottawa PLC link.

**11** Reprogram the subnet.

At this time, Voice Networking has been configured on all nodes using the overrideDirectoryNumber. This means that Voice Networking is acting like an “intelligent PLC”. In other words, connections between nodes are only present when a call is active, but the end-points of the connection are independent of the dialed number because overrideDirectoryNumber is configured.

**12** If not already done, ensure that the VNCS tables are identical on every node.

**13** Start setting the overrideDirectoryNumber to “nothing”. For example, if overrideDirectoryNumber is set to “nothing” on Melbourne, then a Melbourne to Ottawa call would go directly instead of tandeming through Richardson and New York.

**14** Provision the Passport nodes identically. If not already done, provision the VNCS tables, and confirm that calls route correctly.

The table “Passport node VNCS database programming for migration example one” (page 204) shows all of the configuration data for VNCS tables, although sites do not use their own node names. That is, a call from Melbourne to Melbourne does not enter the subnet so it never accesses that entry.

**Figure 43**
**Final stage of conversion of Passport ISDN network**

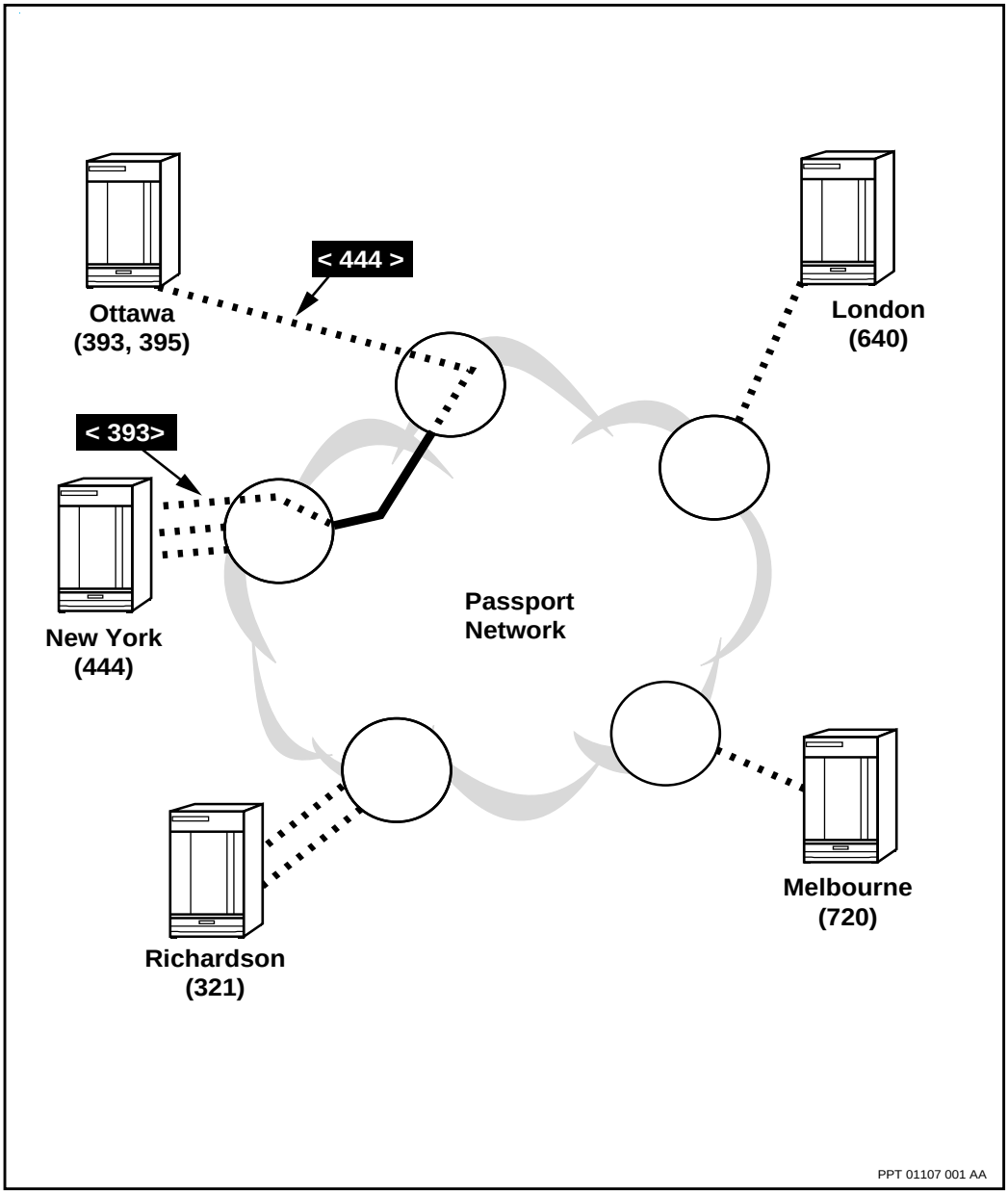


**Figure 44**
**Passport ISDN network after completion of VNCS table entry**

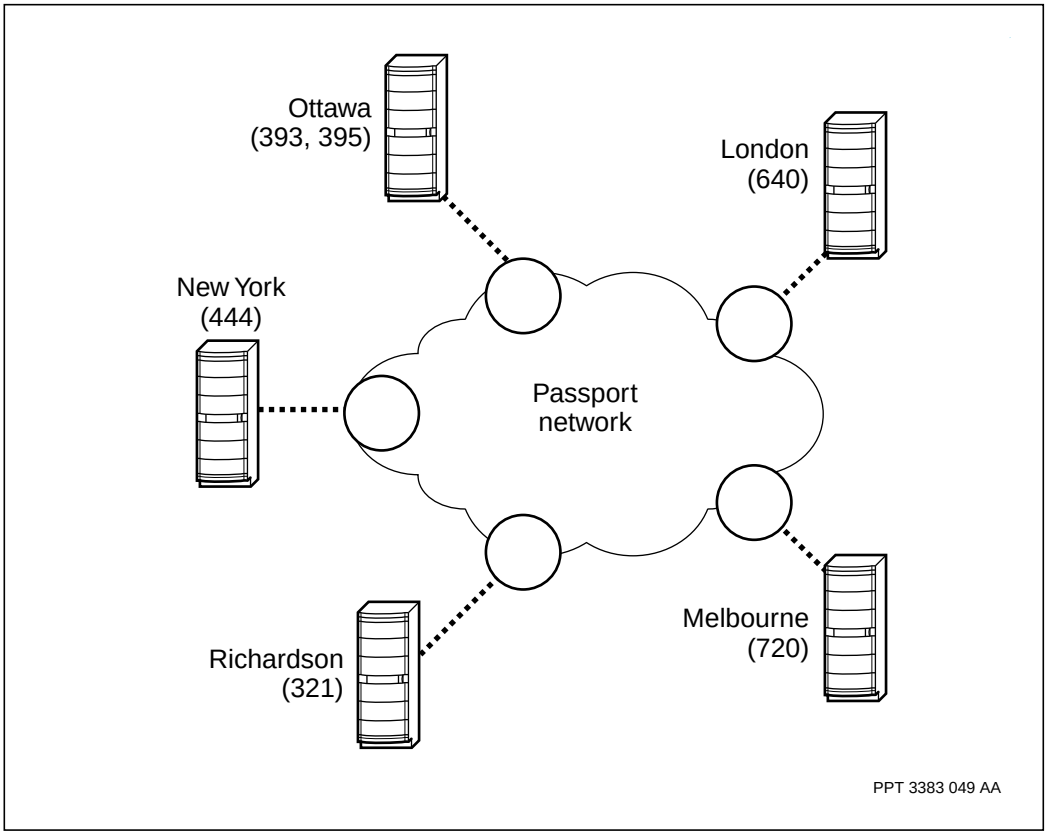



**Table 26**
**Passport node VNCS database programming for migration example one**

| Prefix | Destination | NodeId | Route | VoiceProfile |
|---|---|---|---|---|
| 393, 395 | Ottawa | 632 | VRoute/x | 2 |
| 444 | New York | 1021 | VRoute/x | 2 |
| 321 | Richardson | 1019 | VRoute/x | 2 |
| (Sheet 1 of 2) | | | | |

**Table 26 (continued)**
**Passport node VNCS database programming for migration example one**

| Prefix | Destination | NodeId | Route | VoiceProfile |
|---|---|---|---|---|
| 720 | Melbourne | 32 | VRoute/x | 1 |
| 640 | London | 97 | VRoute/x | 1 |
| (Sheet 2 of 2) | | | | |

Having replaced all PLC services, it is advisable but not necessary to optimize. If all routes to a single PBX created during the switchover could be grouped in one single route on the PBX, the hardware and software can be optimized. If, for example, only two links worth of traffic now uses the three links at New York (since no tandem calls now use the PBX), one link can be removed.

# Inserting Voice Networking into a PBX network

Voice Networking can be inserted into an existing network of PBXs based on point to point TIE (leased line PRI or CAS) connections. However, doing so requires hardware and data operations on both the PBXs and Passport nodes involved. See “Migration: example two” (page 206).

1. Create the initial switched voice PBX-to-Passport node-to-PBX link(s). This can be either a single link or a small number of links replacing PLCs at carefully selected and non-adjoining nodes.
   - Preferred links are either the highest volume path through the network, or single point of entry stubs connecting an isolated PBX to the network. They can be addressed in series (end to end path installation) or parallel (do all discrete paths first).
   - At the selected Passport node to PBX connections add a Voice Networking SVC.
   - Provision the Passport node for Voice Networking on the new logical link. Set up the SVC using an overrideDirectoryNumber to mimic PLC connections.
   - Provision the PBX to access the Voice Networking link (single E1 or DS1 on the PBX shelves).
2. Prove-in the PBX to Passport node link.
3. Disable and remove the PBX to PBX TIE trunk.

4 Repeat steps 1 to 3 with other PLC links until the network has been re-engineered.

5 If not yet created (see bullet in step 1), create the VNCS data and download it to all Passport nodes.

6 Monitor the results of the VNCS mapping to confirm that calls terminate correctly. See the table “Passport node VNCS database programming for migration example two” (page 214) for an example of a completed list of paths.

7 On a node by node basis, turn off the overrideDirectoryNumber for the selected PBX-to-Passport node links. This allows calls originating on the selected PBX-to-Passport node links to switch throughout the subnet.

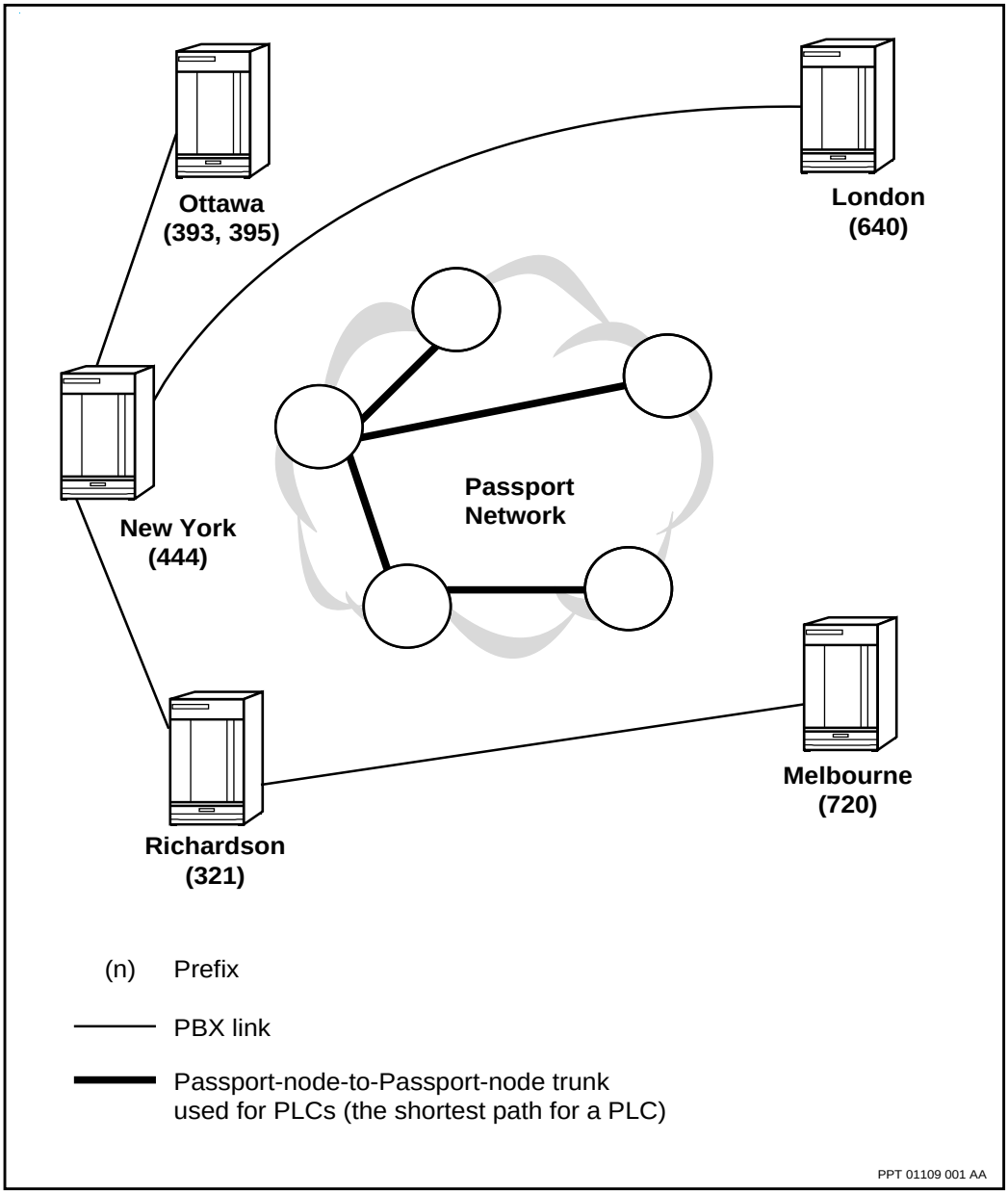
8 Optionally, optimize the Passport node-to-PBX links to match traffic volumes by removing redundant DS1 or E1 links. That is, if a PBX to Passport node link has four E1 cards (providing a total of 120 time slots) but has a peak volume of 75 calls, reduce the E1 card count to three (for a total of 90 time slots).

## Migration: example two

The figure “Initial Passport ISDN network - using PLCs” (page 198) illustrates a network of five PBXs. One Passport node has three PLC links, another has two. The remaining Passport nodes have only one PLC link.

Logically, a Voice Networking connection to any single link PBX (such as London) behaves like a PLC at the stub end. Changing one or more of the Ottawa / London / Melbourne links is simple, with the stub end programming identical at all phases of the change over.

**Figure 45**
**Initial Passport ISDN network—using PLCs**

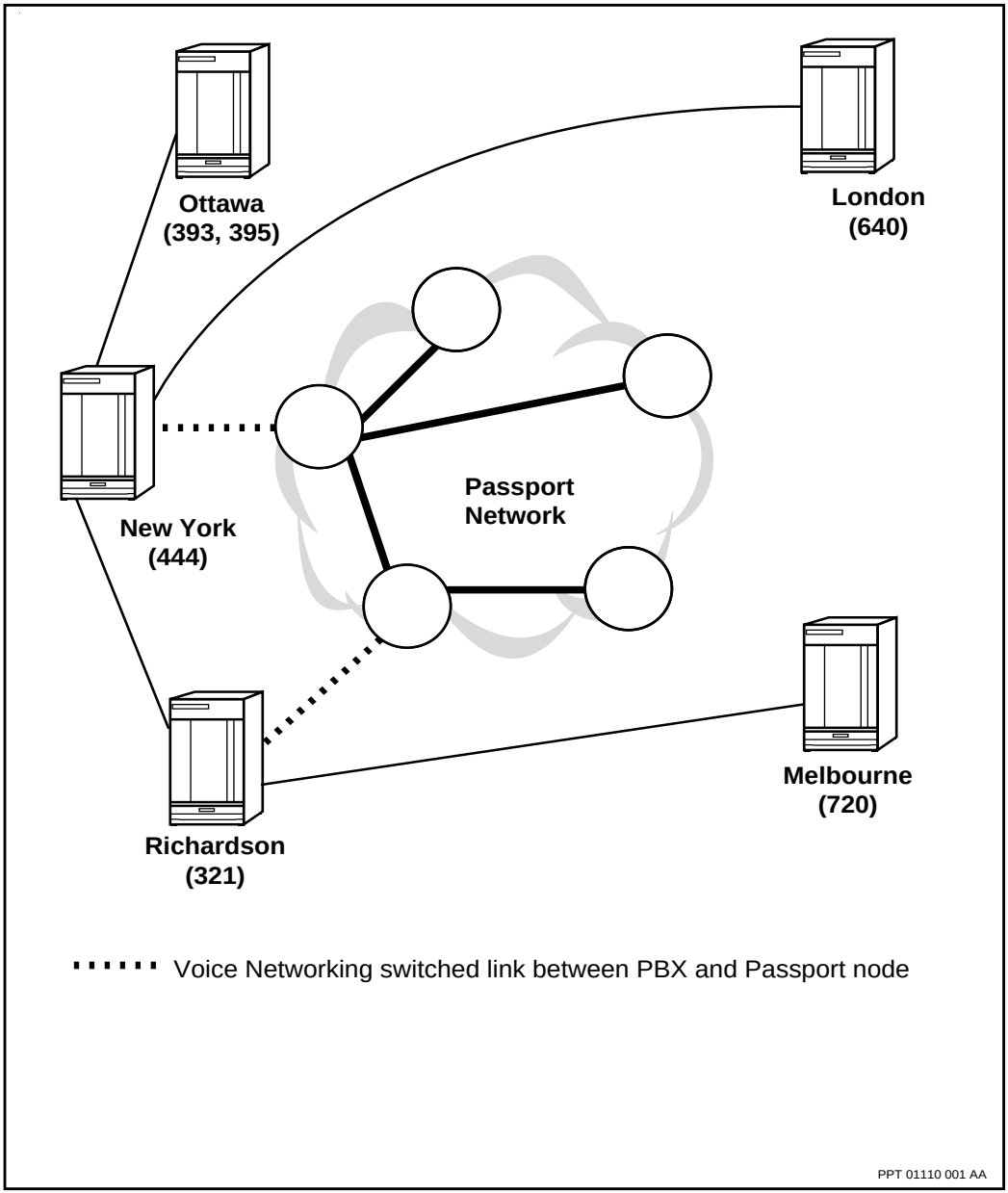


The following is an example of adding a Passport subnet to a private network:

**1** Initially, the New York to Richardson link will be migrated to Voice Networking, See the figure “Stage 1—conversion from PABX ISDN network to Passport ISDN network” (page 209). In this example, this is the highest traffic link. Add parallel SVCs connecting the Passport node with the PBX. These are commissioned and replace the PBX TIE trunk links.

**2** Duplicate the selected link. Program the Passport nodes at the two ends (New York, Richardson) for “fixed end point” operation. Program the PBX to send all calls to the Voice Networking link instead of the TIE trunk. Prove in the new link.

**Figure 46**
**Stage 1—conversion from PABX ISDN network to Passport ISDN network**

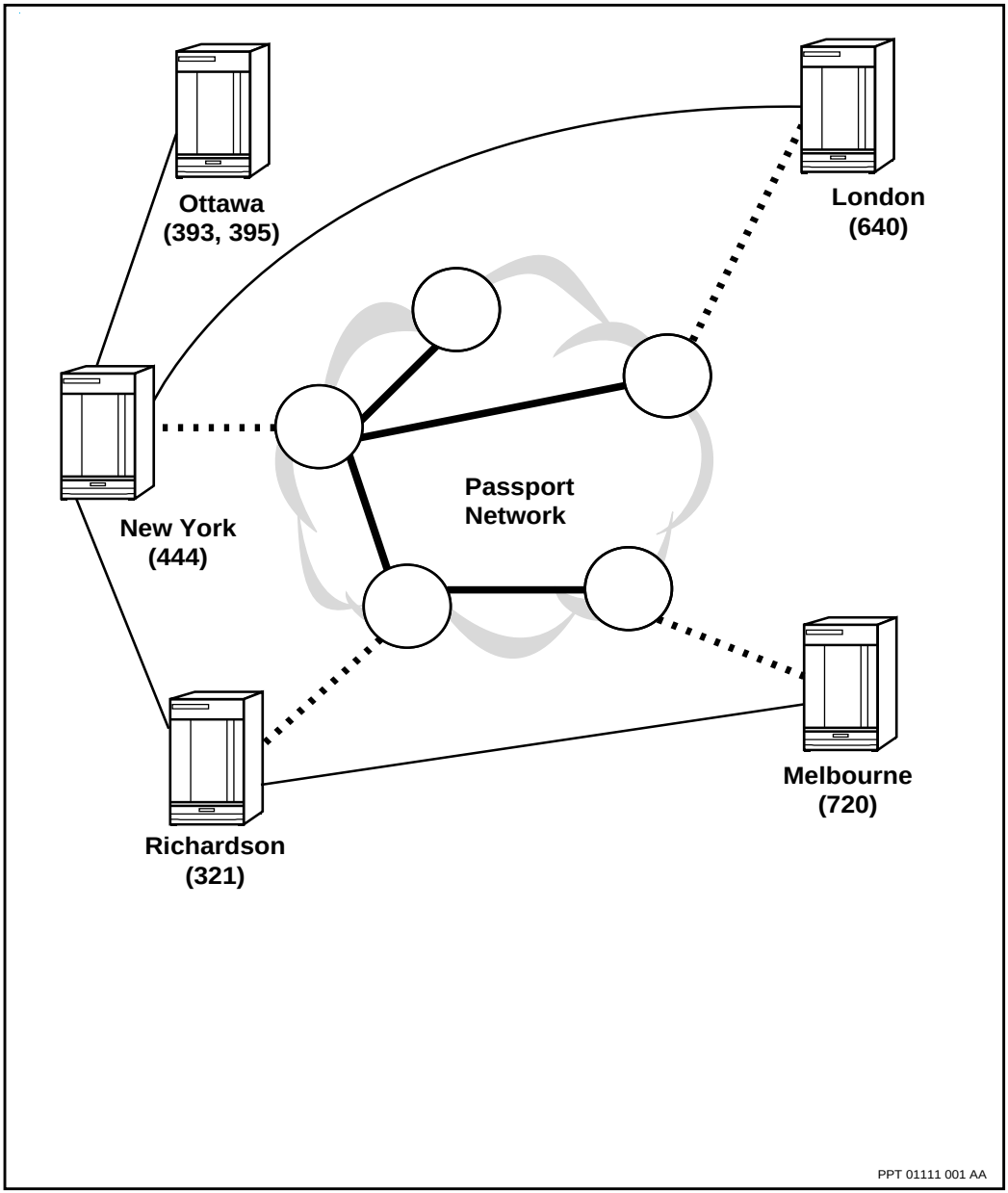


**3** Prove-in the links, de-commission the PBX links (New York - Richardson).

The initial link is now in. See the figure “Stage 1—conversion from PABX ISDN network to Passport ISDN network” (page 209). The New York to Ottawa, New York to London, and Richardson to Melbourne stubs remain. Since traffic to Canada represents the lowest cost (in this example), leave Ottawa for last.

**4** Duplicate the selected links. Program the Passport nodes at the two ends of each link (Richardson and Melbourne, New York and London) for “fixed end point” operation. Program the PBX to send all calls to the Voice Networking link instead of the TIE trunk. Prove in the new links.

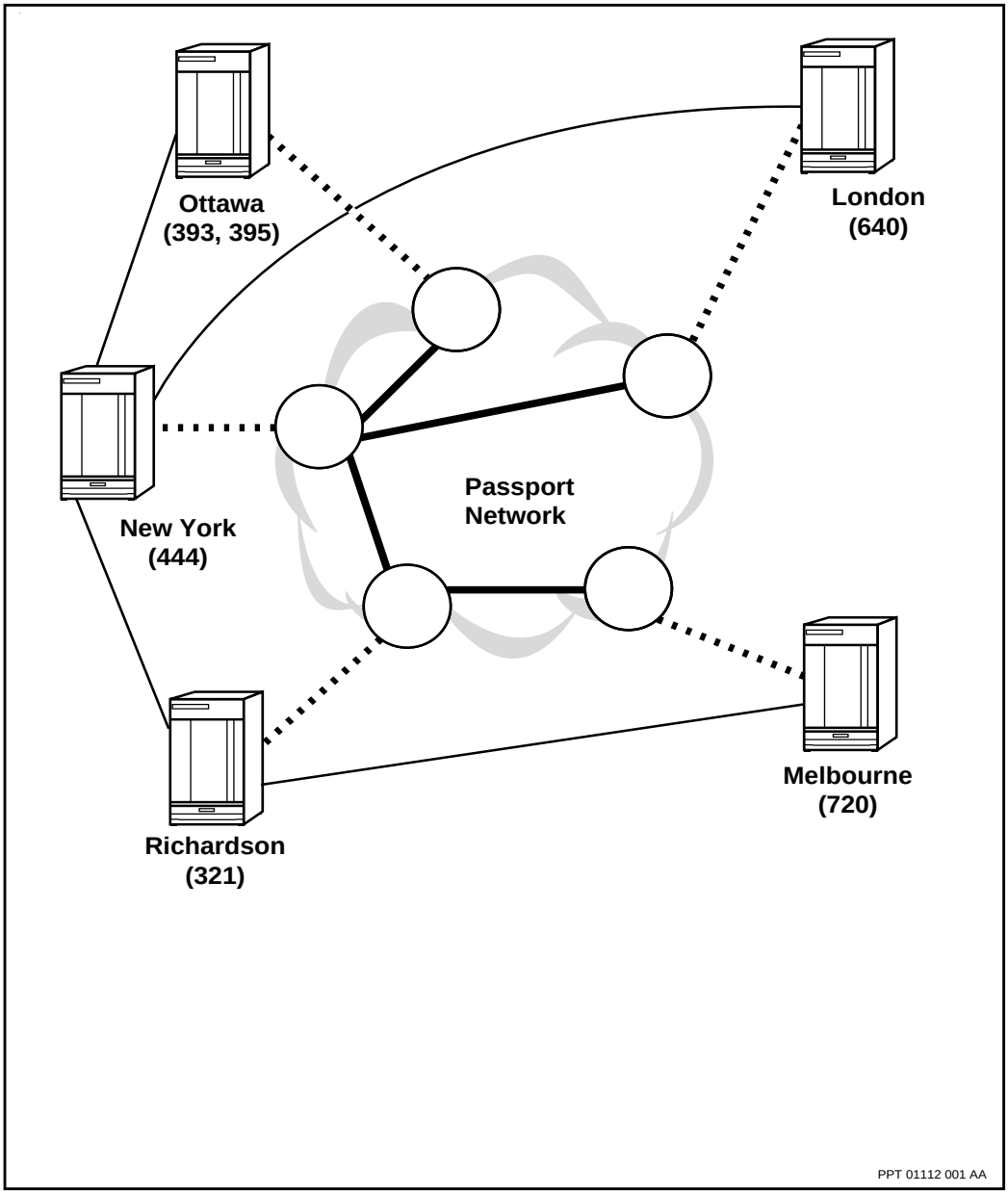
**Figure 47**
**Stage 2—conversion from PABX ISDN network to Passport ISDN network**



5 Prove-in the switched voice links, decommission and remove the PBX links. See the figure “Stage 2—conversion from PABX ISDN network to Passport ISDN network” (page 211). If updating the VNCS data as each node is configured, reprogram the New York, Richardson, Melbourne and London Passport nodes to access the applicable routes. (This may be phased; it may be desirable to confirm the new Voice Networking links and then reprogram the Passport nodes).

6 Duplicate the remaining link. Program the Passport nodes at the two ends of the link (New York and Ottawa) for “fixed end point” operation. Program the PBX to send all calls to the Voice Networking link instead of the TIE trunk. Prove in the new link.

**Figure 48**
**Final stage—conversion from PABX ISDN network to Passport ISDN network**



**7** Prove-in the link from New York to Ottawa, decommission the PBX link and reprogram the subnet. All Passport nodes should now require identical configuration. If it was not already done, provision the VNCS tables, and confirm that calls route correctly.

**8** Having completed the hardware conversion, the links are as shown in the figure “Final stage—conversion from PABX ISDN network to Passport ISDN network” (page 213). The configuration required to access these is required.

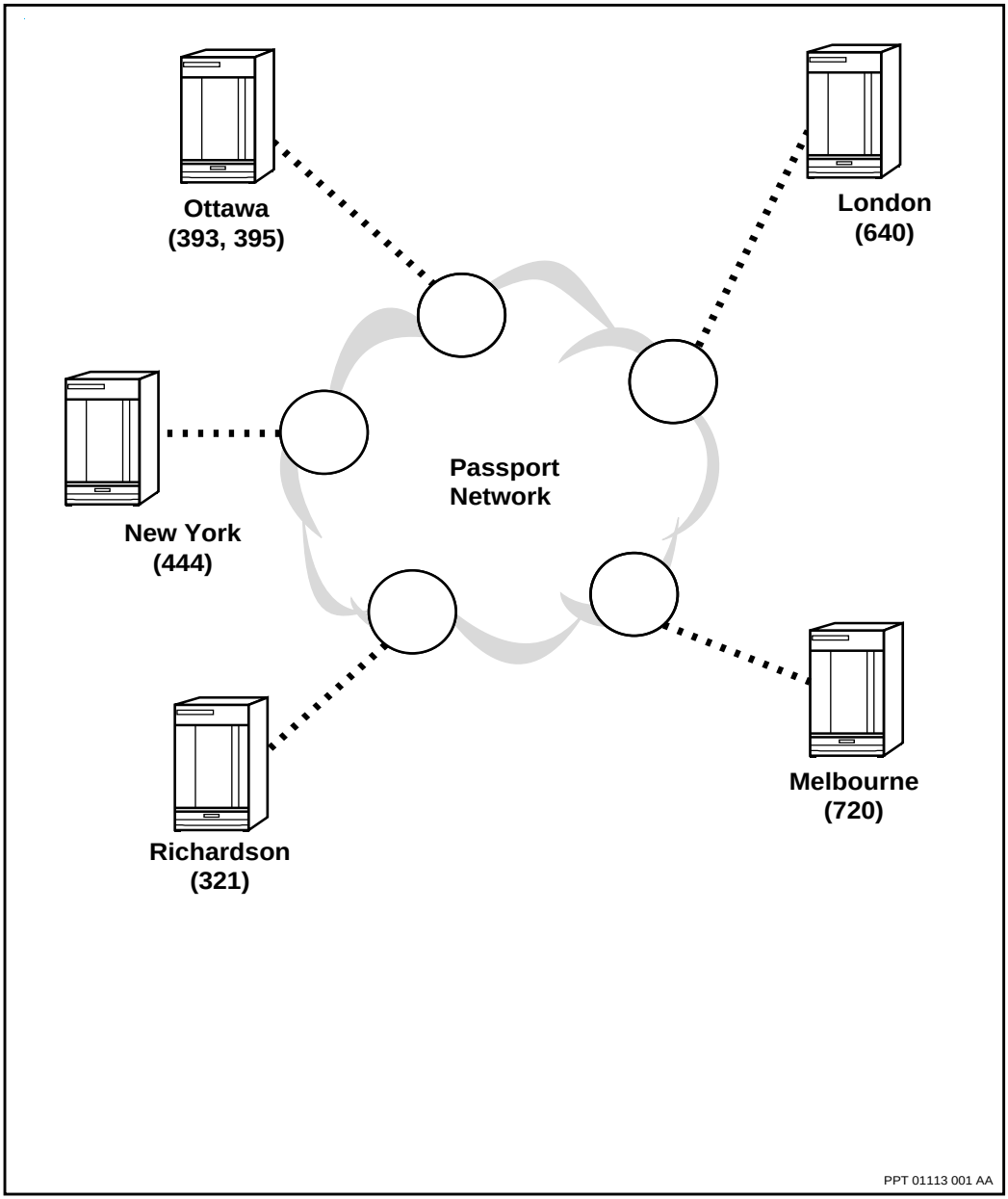
The table “Passport node VNCS database programming for migration example two” (page 214) shows all of the configuration data for VNCS tables, although sites do not use their own node names. That is, a call from Melbourne to Melbourne does not enter the subnet so it never accesses that entry.

**Table 27**
**Passport node VNCS database programming for migration example two**

| Prefix | Destination | NodeId | Route | VoiceProfile |
|---|---|---|---|---|
| 393, 395 | Ottawa | 632 | VRoute/x | 2 |
| 444 | New York | 1021 | VRoute/x | 2 |
| 321 | Richardson | 1019 | VRoute/x | 2 |
| 720 | Melbourne | 32 | VRoute/x | 1 |
| 640 | London | 97 | VRoute/x | 1 |
| | | | | |

Having replaced all PBX TIE trunks, inefficiencies may exist, and it may be advisable (but not necessary) to optimize. If all routes to a single PBX created during the switchover could be grouped in one single route on the PBX, the hardware and software can be optimized. If, for example, only two links worth of traffic now uses the three links at Richardson (since no tandem calls now use the PBX), one link can be removed.

**Figure 49**
**Completed conversion from PABX ISDN network to Passport ISDN network**

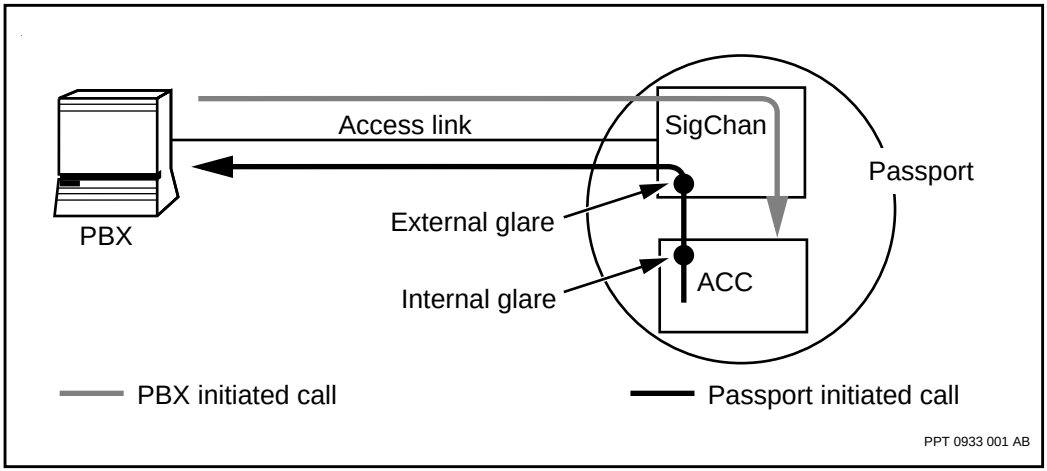


# Appendix C
# Glare processing

Glare, also referred to as call collision or dual seizure, occurs if both a PBX and its access Passport node try to seize the same channel at the same time (see the figure "Glare (call-collision) on access signaling" (page 217)). From the Passport node's perspective, glare can be

- "Internal glare" (page 218)
- "External glare" (page 218)

**Figure 50**
**Glare (call-collision) on access signaling**



## Internal glare

The Passport node initiated call has not yet been passed to the SignallingChannel component when glare is detected. This glare is not detectable by the PBX. The call handling is as follows:

1 The PBX gets the requested channel for its call.

2 The Passport node selects another channel for its call. That is, the Passport node backs-off; no channel negotiation is necessary.

## External glare

The call initiated by the Passport node has already been passed to the SignallingChannel component when glare is detected. Call handling proceeds differently depending on which access signaling—CAS or CCS—is being used.

For CAS signaling, the Passport node backs off and tries to reestablish the call.

For CCS signaling, call handling depends on who is the network and who is the user, and proceeds as follows:

1 The network side gets the requested channel for its call (call setup is done using ‘exclusive’).

2 The user side may be offered another channel for its call by the network side. That is, the user backs-off; a channel negotiation is initiated by the network (call setup is done using ‘preferred’).

# Appendix D
# Call release cause codes

Call release cause codes only apply to Voice Networking calls that use common channel signaling (CCS) protocols. Voice Networking handles internally and externally generated call release cause codes for failed (for example, during call establishment) and clearing calls. Internal cause codes are those generated by the Passport node or from within the subnet. External cause codes are those received by a Passport node from a PBX. The table "Call release cause code values" (page 220) contains all cause values in the range 0 to 127. In most cases, Voice Networking transports without modification the cause values that it receives. However, some cause values do not apply at all points in a connection and not all cause values are supported by all protocols and PBXs.

Standard cause code values are commonly defined for all protocols in the ITU-T Q.850 specification. However, each individual protocol may introduce a different handling of a given cause value or additional cause code values. The word *No* in the table "Call release cause code values" (page 220) means the cause value is not supported by the particular CCS protocol. The word *Yes* means that the CCS protocol supports the cause value and Voice Networking does not map the value to another, unless otherwise indicated. The table "Call release cause code values" (page 220) also notes the cause values that Voice Networking maps to more common values. See also "Internal cause code definition" (page 225) for more information about cause code handling.

> ***Note:*** Unless otherwise specified, the supported Euro-ISDN cause values include the ETSI Austrian and German user variants and the generic ETSI ISDN network variant.

**Table 28**
**Call release cause code values**

| Cause value | Description | Supported by... | | | |
|---|---|---|---|---|---|
| | | **NIS A211-1** | **ETSI QSIG** | **Euro-ISDN** | **MCDN** |
| 0 | Unallocated (unassigned) number | No (maps to 31) | No (maps to 31) | No | No (0 is sent) |
| 1 | Unassigned (unallocated) number | Yes | Yes | Yes | Yes |
| 2 | No route to specified transit network | Yes | No (maps to 3) | Yes (maps to 3) | Yes |
| 3 | No route to destination | Yes | Yes | Yes | Yes |
| 4, 5 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 6 | Channel unacceptable | No (maps to 31) | No (maps to 31) | Yes | Yes |
| 7 | Call awarded and being established | No (maps to 31) | No (maps to 31) | No | Yes |
| 8 - 15 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 16 | Normal call clearing | Yes | Yes | Yes | Yes |
| 17 | User busy | Yes | Yes | Yes | Yes |
| 18 | No user responding | Yes | Yes | Yes | Yes |
| 19 | No answer from user (user alerted) | Yes | Yes | Yes | Yes |
| 20 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 21 | Call rejected | Yes | Yes | Yes | Yes |
| 22 | Number changed | Yes | Yes | Yes | Yes |
| 23 - 25 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 26 | Non-selected user clearing | No (maps to 31) | No (maps to 31) | Yes (maps to 31) | Yes |

(Sheet 1 of 6)

**Table 28 (continued)**
**Call release cause code values**

| Cause value | Description | Supported by... | | | |
|---|---|---|---|---|---|
| | | **NIS A211-1** | **ETSI QSIG** | **Euro-ISDN** | **MCDN** |
| 27 | Destination out-of-service | Yes | Yes | Yes | Yes |
| 28 | Invalid number format | Yes | Yes | Yes | Yes |
| 29 | Facility rejected | Yes | No (maps to 31) | Yes (maps to 31) | Yes |
| 30 | Response to status enquiry | No (maps to 31) | No (maps to 31) | No | Yes |
| 31 | Normal, unspecified | Yes | Yes | Yes | Yes |
| 32 - 33 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 34 | No circuit/channel available | Yes | Yes | Yes | Yes |
| 35 - 37 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 38 | Network out of order | No (maps to 31) | No (maps to 41) | Yes (maps to 41) | Yes |
| 39 - 40 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 41 | Temporary failure | Yes | Yes | Yes | Yes |
| 42 | Switching equipment congestion | Yes | No (maps to 41) | Yes (maps to 41) | Yes |
| 43 | Access information discarded | Yes | No (maps to 31) | Yes (maps to 31) | Yes |
| 44 | Requested circuit/ channel not available | Yes | Yes | Yes | Yes |
| 45 | Channel preempted | No (maps to 31) | No (maps to 41) | No | Yes |
| 46 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |

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**Table 28 (continued)**
**Call release cause code values**

| Cause value | Description | Supported by... | | | |
|---|---|---|---|---|---|
| | | **NIS A211-1** | **ETSI QSIG** | **Euro-ISDN** | **MCDN** |
| 47 | Resources unavailable, unspecified | Yes | No (maps to 44) | Yes | Yes |
| 48 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 49 | Quality of service unavailable | No (maps to 31) | No (maps to 58) | Yes (maps to 58) | Yes |
| 50 | Requested facility not subscribed | Yes | No (maps to 31) | Yes (maps to 31) | Yes |
| 51 | Call barred due to access group restrictions | No (maps to 31) | No (maps to 31) | No | Yes |
| 52 | Outgoing call barred | No (maps to 31) | No (maps to 31) | No | Yes |
| 53 | Outgoing call barred within customer user group (CUG) | No (maps to 31) | No (maps to 31) | Yes (ETSI Germany only; otherwise maps to 31) | No (maps to 31) |
| 54 | Incoming call barred | Yes | No (54 is sent) | No | Yes |
| 55 | Incoming calls barred within CUG | No (maps to 31) | No (maps to 31) | Yes (ETSI Germany only; otherwise maps to 31) | No (maps to 31) |
| 56 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 57 | Bearer capability not authorized | Yes | Yes | Yes | Yes |
| 58 | Bearer capability not presently available | Yes | Yes | Yes | Yes |

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**Table 28 (continued)**
**Call release cause code values**

| Cause value | Description | Supported by... | | | |
|---|---|---|---|---|---|
| | | **NIS A211-1** | **ETSI QSIG** | **Euro-ISDN** | **MCDN** |
| 59 - 62 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 63 | Service or option unavailable, unspecified | Yes | Yes | Yes | Yes |
| 64 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 65 | Bearer capability not implemented | Yes | Yes | Yes | Yes |
| 66 | Channel type not implemented | Yes | No (maps to 65) | Yes (maps to 65) | Yes |
| 67 - 68 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 69 | Requested facility not implemented | No (maps to 31) | No (maps to 63) | Yes (maps to 63) | Yes |
| 70 | Only restricted digital information bearer capability is available | Yes | No (maps to 63) | Yes (maps to 63) | Yes |
| 71 - 78 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 79 | Service or option not implemented | Yes | No (maps to 63) | Yes (maps to 63) | Yes |
| 80 | N/A | No (maps to 31) | No (maps to 31) | No | No (80 is sent) |
| 81 | Invalid call reference value | Yes | Yes | Yes | Yes |
| 82 | Identified channel does not exist | Yes | No (maps to 44) | Yes (maps to 44) | Yes |
| 83 | A suspended call exists, but this call identity does not | Yes (maps to 31) | No (maps to 0) | No | Yes (maps to 0) |

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**Table 28 (continued)**
**Call release cause code values**

| Cause value | Description | Supported by... | | | |
|---|---|---|---|---|---|
| | | **NIS A211-1** | **ETSI QSIG** | **Euro-ISDN** | **MCDN** |
| 84 | Call identity in use | Yes (maps to 31) | No (maps to 0) | No | Yes (maps to 0) |
| 85 | No call suspended | Yes (maps to 31) | No (maps to 0) | No | Yes (maps to 0) |
| 86 | Call having the requested call identity has been cleared | Yes (maps to 31) | No (maps to 0) | No | Yes (maps to 0) |
| 87 | User not member of CUG | No (maps to 31) | No (maps to 31) | Yes (ETSI Germany only; otherwise maps to 31) | No (maps to 31) |
| 88 | Incompatible destination | Yes | Yes | Yes | Yes |
| 89 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 90 | Non-existent CUG | Yes | No (maps to 31) | Yes (ETSI Germany only; otherwise maps to 31) | No (maps to 31) |
| 91 | Invalid transit network selection | No (maps to 31) | No (maps to 31) | No | Yes |
| 92 - 94 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 95 | Invalid message, unspecified | Yes | No (maps to 31) | Yes (maps to 111) | Yes |
| 96 | Mandatory information element missing | No (maps to 31) | No (maps to 31) | Yes | Yes |
| 97 | Message type non-existent or not implemented | No (maps to 31) | No (maps to 31) | Yes | Yes |

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**Table 28 (continued)**
**Call release cause code values**

| Cause value | Description | Supported by... | | | |
|---|---|---|---|---|---|
| | | NIS A211-1 | ETSI QSIG | Euro-ISDN | MCDN |
| 98 | Message type either incompatible with current state or not implemented | No (maps to 97 or 101) | No (maps to 31) | Yes | Yes |
| 99 | Information element non-existent or not implemented | No (maps to 31) | No (maps to 31) | Yes | Yes |
| 100 | Invalid information element contents | No (maps to 31) | No (maps to 31) | Yes | Yes |
| 101 | Message not compatible with call state | No (maps to 31) | No (maps to 31) | Yes | Yes |
| 102 | Recovery on timer expiry | No (maps to 31) | No (maps to 31) | Yes | Yes |
| 103 - 110 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 111 | Protocol error, unspecified | No (maps to 31) | No (maps to 31) | Yes | Yes |
| 112 - 126 | N/A | No (maps to 31) | No (maps to 31) | No | No (maps to 31) |
| 127 | Interworking, unspecified | No (maps to 31) | No (maps to 31) | Yes | No (maps to 31) |

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# Internal cause code definition

For certain internal cause codes relating to call establishment you can configure Voice Networking to transmit a particular cause code value. This capability allows you to exactly match the cause value required by an originating PBX to establish a call on an alternate route. The capability to define a cause value applies to calls using the following CCS protocols: ETSI QSIG, NIS A211-1, and Euro-ISDN (the MCDN protocol uses the drop back busy capability to reroute calls).

The table “Definable internal cause codes” (page 226) contains the cause, typical clearing reason(s), and examples of the cause value that is sent for each protocol. When defining a cause value for any of the causes listed in the table, any value in the range 0 to 127 is acceptable. See “Configuring the SignalingChannel component” (page 40) for procedural information on how to specify cause code values.

**Table 29**
**Definable internal cause codes**

| Cause | Typical clearing reason(s) | Example cause value sent | | |
|---|---|---|---|---|
| | | NIS A211-1 | ETSI QSIG | Euro-ISDN |
| Egress link out of service | • The data link to the remote PBX is down.<br>• The channels at the remote end are in maintenance mode. | 27 | 27 | 27 |
| No circuit/channel available | • No timeslot available for an incoming call. | 34 | 34 | 34 |
| Temporary failure | • path-oriented routing system (PORS) fails to establish a call due to subnet congestion.<br>• A trunk is disabled.<br>• Calling side fails to decode the payload capabilities sent by the called side. | 41 | 41 | 41 |
| Switched equipment congestion | • Allocation of internal resources for an incoming call fails (for example, the SignallingChannel component is down). | 42 | 41 | 42 |
| Requested circuit/ channel not available | • An exclusively requested channel is not in service.<br>• A channel request is unsuccessful. | 44 | 44 | 44 |

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**Table 29 (continued)**
**Definable internal cause codes**

| Cause | Typical clearing reason(s) | Example cause value sent | | |
|---|---|---|---|---|
| | | **NIS A211-1** | **ETSI QSIG** | **Euro-ISDN** |
| Resource unavailable, unspecified | • Path establishment timer expires.<br>• An H-channel (hybrid channel) is requested.<br>• The SwitchedVoiceService component associated with a particular call is locked on the egress Passport node. | 47 | 44 | 47 |
| Service not allowed | • The requested call type (voice or data) is not compatible with the configured value of the typeOfRoute attribute under the VoiceRoute component at the calling side. | 63 | 63 | 63 |
| No such channel | • The requested channel does not exist. | 82 | 44 | 82 |
| Incompatible destination | • The capabilities of the called and calling side do not match. | 88 | 88 | 88 |

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Passport 7400

# Voice Networking

Guide

Release 5.2

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