Passport 7400

Voice Transport Guide

241-7401-750

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5.2S2 Standard
General availability. Contains information on Passport 7400 for the PCR5.2 GA release.

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About this document

This guide describes the Passport 7400 Voice Transport service.

The term Voice Transport in this guide refers to the Passport 7400 Voice Transport service unless otherwise specified.

The term network in this guide refers in general to any switching network, and often to the Passport 7400 network in particular.

The term interworking in this guide refers to Voice Transport connections between voice function processors (FPs) and Multipurpose Voice Platform Enhanced (MVP-E) FPs.

Who should read this guide

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

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

What you need to know

This guide assumes that you understand the Passport 7400 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 241-7401-750 *Passport 7400 Voice Transport Guide* contains the following sections:

- “Voice Transport configuration” (page 21) presents an overview of the tasks required to configure the Voice Transport service.
- “Basic Voice Transport components configuration” (page 25) contains the procedures required to configure the basic Voice Transport components.
- “Voice Transport connection configuration” (page 33) contains the procedures for configuring the Voice Transport service.
- “Voice Transport configuration considerations” (page 41) contains information on the details you need to consider when configuring the Voice Transport service, and some configuration checklists.
- “Voice Transport fundamentals” (page 53) presents an overview of the Voice Transport service and lists the features which make it unique.
- “Voice Transport capabilities and system parameters” (page 65) describes the features and system parameters of Voice Transport.
- “Voice Transport, PORS, and route selection” (page 95) describes the Path-Oriented Routing System and Passport 7400 Network Clock Synchronization. Both of these elements support Voice Transport.
- “Voice Transport application examples” (page 117) describes how to provision Voice Transport for typical DS1 and E1 PABX trunk interconnections.
- “Monitoring and troubleshooting Voice Transport” (page 159) outlines how to monitor, and if necessary, troubleshoot problems that can occur during installation of Voice Transport.
- “Data stream refresher” (page 173) briefly describes the relationship of frames, timeslots, channels, and bits in DS1 and E1 signal formats.
- “Signalling refresher” (page 177) describes channel associated signalling (CAS) and common channel signalling (CCS).
- “Voice Transport function processor migration information” (page 183) contains information about migrating the Voice Transport service to MVP-E FPs from voice FPs.

What’s new in this document

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.
- Legacy voice FPs and MVP FPs are no longer supported in PCR 5.2

Other changes made 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

Text conventions

This document uses the following text conventions:

- 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 options 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 Passport documents for information on installing and operating Voice Transport in your network:

- 241-5701-030 *Passport 7400, 15000, 20000 Overview*
- 241-7401-200 *Passport 7400 Hardware Description*
- 241-7401-240 *Passport 7400 Hardware Installation, Maintenance and Upgrade*
- 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference*
- 241-5701-270 *Passport 7400, 15000, 20000 Software Installation Guide*
- 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide*

- 241-5701-050 *Passport 7400, 15000, 20000 Commands*
- 241-5701-420 *Passport 7400, 15000, 20000 Trunking Guide*
- 241-5701-435 *Passport 7400, 15000, 20000 Path-Oriented Routing System Guide*
- 241-5701-060 *Passport 7400, 15000, 20000 Components*
- 241-5701-500 *Passport 6400, 7400, 15000, 20000 Alarms*

How to get more help

For information on training, problem reporting, and technical support, see the “Nortel Networks support services” section in the product overview document.

Chapter 1
Voice Transport configuration

You can provision Voice Transport using the default settings provided with the package or you can provision it to meet your specific requirements. The default settings are designed to provide a connection using the fewest number of hops across the network.

Prerequisites to Voice Transport configuration

- Review the section “What systems and hardware does Voice Transport require?” (page 55).
- Install Voice Transport and related software according to the instructions in 241-5701-270 *Passport 7400, 15000, 20000 Software Installation Guide*.
- Verify that the version of your routing software contains the Path Oriented Routing System (PORS). Voice Transport cannot operate without PORS.
- Refer to 241-5701-435 *Passport 7400, 15000, 20000 Path-Oriented Routing System Guide* for information to provision with PORS.
- Set the featureList attribute to vtds as described in 241-5701-270 *Passport 7400, 15000, 20000 Software Installation Guide*. For example,

```
set software lpt/<n> featureList vtds
```

Note: Other packaging options are available, such as btds and htds. As well, you can add faxRelay, tandemPassThrough, g728, and g729 to the *featureList* attribute for Voice Transport services on an MVP-E FP. Use the option(s) appropriate to your configuration.

- Verify that the logical processor type for Passport trunks has path-oriented software in its feature list. Use the following command:

  `set software lpt/trunk featureList porsTrunks`

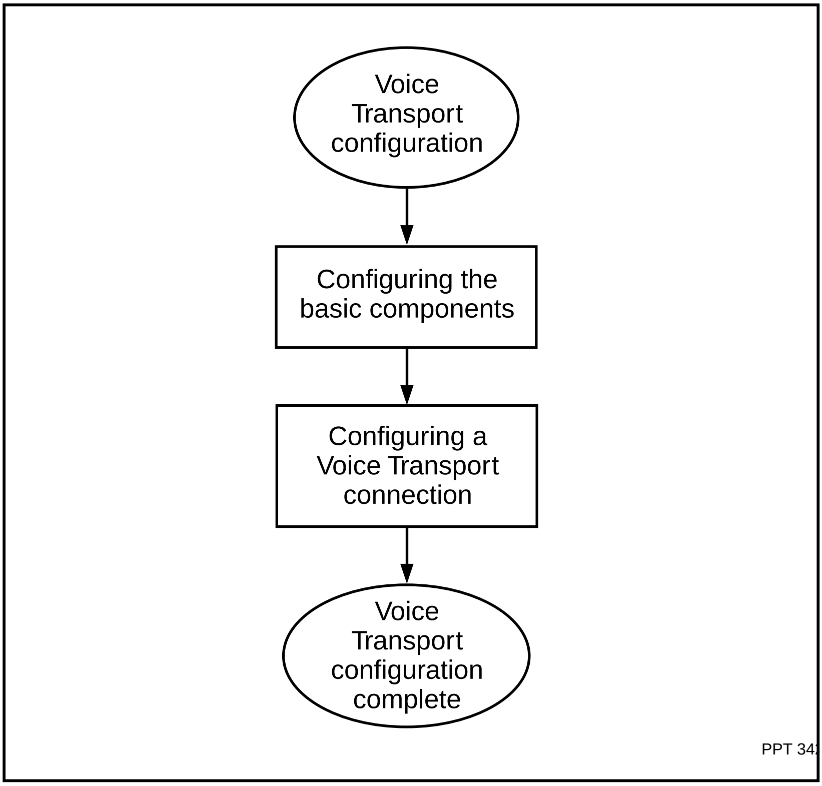
  Note: All nodes in the network that are candidates for Voice Transport traffic must be running the same version of Voice Transport-compatible software and must have the Trunk PathAdministrator component added to the provisioning view.

- Refer to 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference* for card, logical processor, and port provisioning procedures.
- Map out how you want the master clocking signal to be distributed through the network. See 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide* for information on how to structure the delivery of the clocking signal and how to configure network clock synchronization.
- If you choose to provision other attributes associated with the service, be sure that you understand what each attribute does. You can view provisionable component and attribute default settings by using the on-line Help command or by referring to 241-5701-060 *Passport 7400, 15000, 20000 Components*. Refer also to 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide* for basic provisioning information and to 241-5701-050 *Passport 7400, 15000, 20000 Commands* for command information.

Voice transport configuration task

“Voice transport configuration task flow” (page 23) shows you the sequence of tasks and procedures you perform to configure Voice Transport. To link to any task or procedure, go to “Task navigation” (page 23).

Figure 1
Voice transport configuration task flow



Task navigation

- "Basic Voice Transport components configuration" (page 25)
- "Voice Transport connection configuration" (page 33)

Chapter 2
Basic Voice Transport components configuration

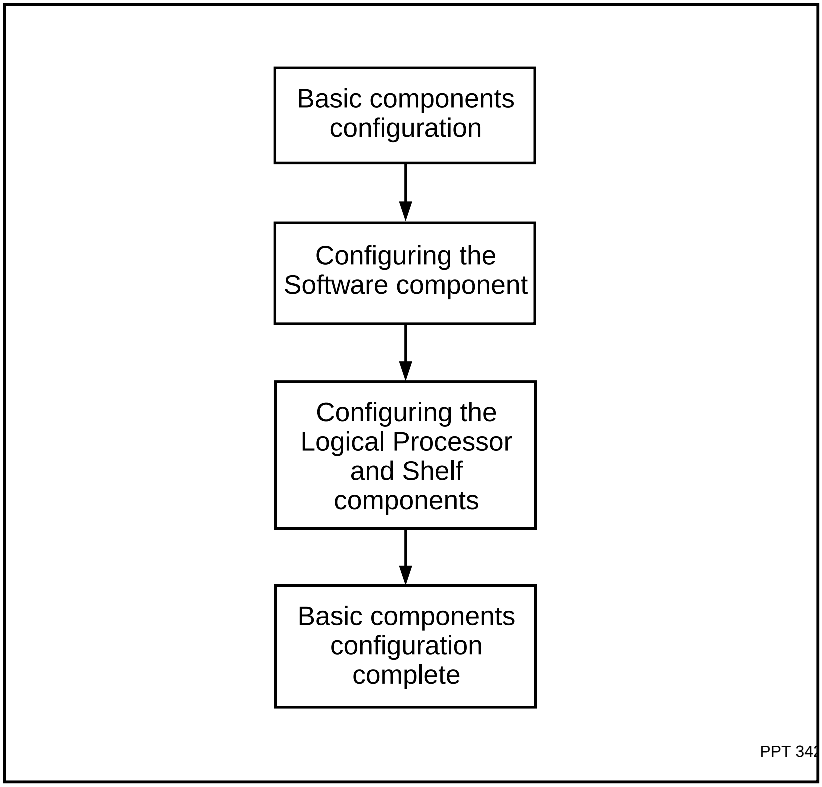
Configure the basic components of the Voice Transport service.

- “Basic Voice Transport components configuration task” (page 25)

Basic Voice Transport components configuration task

“Basic Voice Transport components configuration task flow” (page 26) shows you the sequence of tasks and procedures you perform to configure the basic Voice Transport components. To link to any of the tasks or procedures, go to “Task navigation” (page 26).

Figure 2
Basic Voice Transport components configuration task flow



Task navigation

- "Configuring the Software component" (page 27)
- "Configuring the LogicalProcessor and Shelf components" (page 28)

Configuring the Software component

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

Procedure steps

1 Add Voice Transparent Data Service software to the LogicalProcessorType component instance for the control processor (CP).

```
set sw lpt/CP featureList voiceTransparentDataService
```

2 Add an Lpt component instance for the Voice Transparent Data Service.

```
add sw lpt/vtds
```

3 If you plan to use G.728, G.729, or G.729A voice encoding, tandem pass through, or fax demodulation (fax relay) on an MVP-E FP, add the corresponding features.

```
set sw lpt/vtds featureList <features>
```

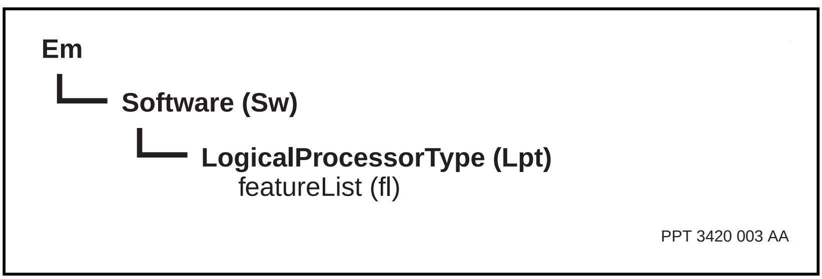
4 Repeat step 1 to step 3 to configure Voice Transparent Data Service software on each Passport node in the network that connects to a PBX.

Variable definitions

Variable	Definition
<features>	Any or all of g728, g729, tandemPassThrough, faxRelay.

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 VTDS software.

set lp/<lp_number> logicalProcessorType sw lpt/vtds

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 to each LP.

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

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

set lp/<lp_number> <port>/<port_number> lineType <framing_format>

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

set lp/<lp_number> <port>/<port_number> clockingSource module

10 For DS1 CAS ports, set the zero coding value.

set lp/<lp_number> ds1/<port_number> zeroCoding <zeroCoding_value>

11 For DS1 CAS ports, set the RAI alarm type value.

set lp/<lp_number> ds1/<port_number> raiAlarmType <raiAlarmType_value>

12 Add channels and assign timeslots to the channels for each port. The type of signaling protocol and port determines how to provision channels and timeslots. For CAS protocols interfacing to an E1 port, define 30 channels to process calls. For CCS protocols interfacing to a DS1 port, define 23 channels for bearer services and one (assigned timeslot 16) 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>

13 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

14 Optionally, for a DS1 port using a CCS protocol, 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

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

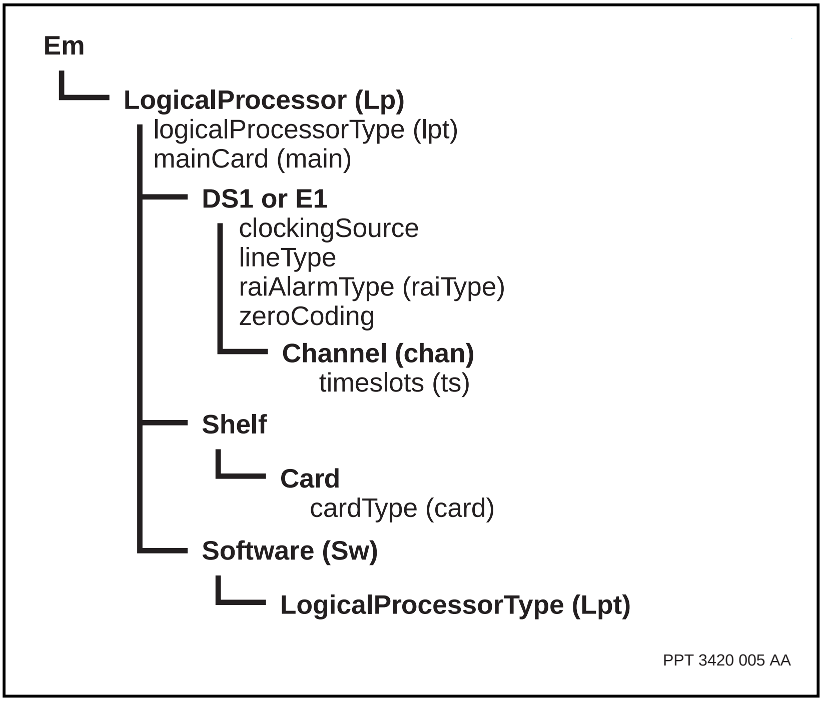
16 Repeat step 1 to step 15 to configure logical and function processors on each Passport node in the network that connects to a PBX.

Variable definitions

Variable	Definition
<card_number>	The instance number of the card.
<FP>	The card type. To determine the value to enter for the FP you are configuring, refer to 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference*. Must contain the same value at both ends of the connection. However, you can interconnect one-port and four-port DS1MVP-E FPs and one-port and four-port E1 MVP-E FPs.
<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.
<port>	ds1 or e1
<port_number>	The instance number of the port.
<raiAlarmType_value>	fdl (for DS1 ESF CAS trunks) or bit2 (for DS1 SF (D4) CAS trunks)
<signaling_channel>	The channel you assign to carry signaling information. 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 define channels 1 to 30 for bearer services and assign timeslot 16 to channel 31.
<timeslot_number>	The timeslot number.
<zeroCoding_value>	b8zs (for DS1 ESF CAS trunks) or bit7stuffing (for DS1 SF (D4) CAS trunks)

Procedure job aid

Figure 4
Configuring the LogicalProcessor and Shelf components component hierarchy



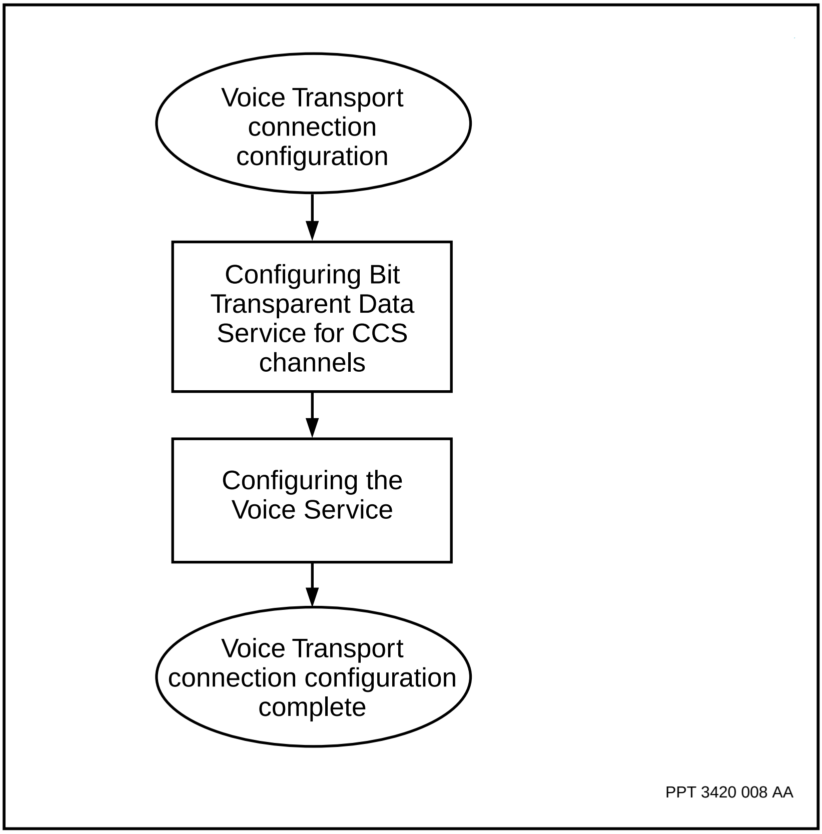
Chapter 3
Voice Transport connection configuration

Configure BTDS for CCS channels, and the Voice Service.

Voice Transport connection configuration task

"Voice Transport connection configuration task flow" (page 34) shows you the sequence of tasks and procedures you perform to configure Voice Transport connections. To link to any task or procedure, go to "Task navigation" (page 34).

Figure 5
Voice Transport connection configuration task flow



Task navigation

- "Configuring Bit Transparent Data Service for CCS channels" (page 35)
- "Configuring the Voice Service" (page 37)

Configuring Bit Transparent Data Service for CCS channels

Configure Bit Transparent Data Service (BTDS) for each CCS channel.

Procedure steps

1 For each CCS channel, add a bit transparent data service (BTDS) application.

```
add btds/<btds_number>
```

2 For each CCS channel, set the remote name of the BTDS application at the far end of the Passport 7400 network.

```
set btds/<btds_number> plc remoteName ""<remote_name>"
```

3 Link each BTDS application with the CCS signaling channel.

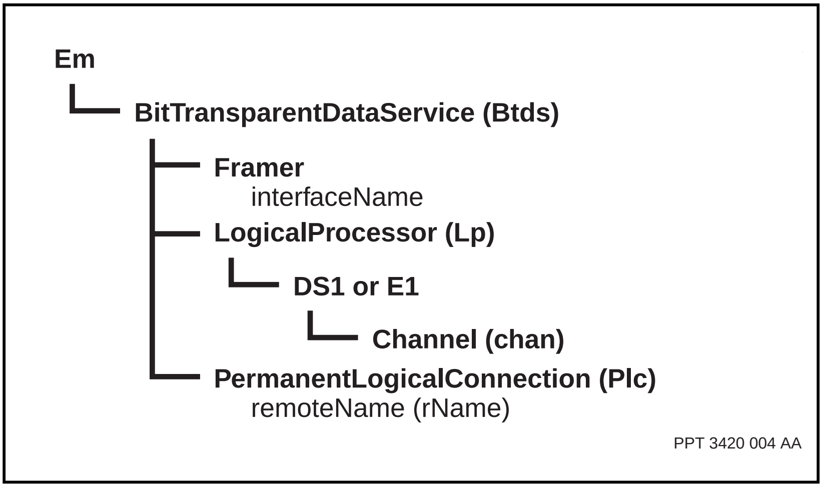
```
set btds/<btds_number> framer interfaceName lp/
<lp_number> <port>/<port_number> chan/
<channel_number>
```

Variable definitions

Variable	Definition
<btds_number>	The instance number of the BTDS application.
<channel_number>	The instance number of the channel. In this case, it is usually timeslot 24 for DS1 and timeslot 16 for E1.
<lp_number>	The instance number of the LP.
<port>	ds1 or e1
<port_number>	The instance number of the port.
<remote_name>	The name of the Voice Service at the far end of the Passport 7400 network.

Procedure job aid

Figure 6
Configuring Bit Transparent Data Service for CCS channels component hierarchy



Configuring the Voice Service

Configure the Voice Service on the network.

Prerequisites

- Refer to 241-5701-060 *Passport 7400, 15000, 20000 Components* for descriptions of the Framer and Plc components and their attributes and values.
- Bandwidth reservation is based on the provisioned encoding for the particular type of connection—voice, modem or facsimile. Review the default values offered by Voice Transport—particularly for connections using g728 or g729 encoding.

Procedure steps

1 Add a Voice Service application for each channel.

```
add vs/<voice_service_number>
```

2 Set the remote name of each Voice Service at the far end of the Passport 7400 network.

```
"set vs/<voice_service_number> plc remoteName
“<remote_name>”
```

3 Link each Voice Service with a voice channel.

```
set vs/<voice_service_number> framer interfaceName lp/
<lp_number> <port_type>/<port_number> chan/
<channel_number>
```

4 Set the voice service to transport the signaling bits across the network.

```
set vs/<voice_service_number> framer casSignalling
<casSignalling_value>
```

5 For CAS connections, set the signalBits attribute.

```
set vs/<voice_service_number> framer signalBits
<signalBits_value>
```

6 For CAS connections that have the same signaling format on both ends, set the required transmit bandwidth.

```
set vs/<voice_service_number> plc requiredTxBandwidth
<tx_value>
```

7 For CAS connections that have the same signaling format on both ends, set the required receive bandwidth.

set vs/<voice_service_number> plc requiredRxBandwidth <rx_value>

8 For connections that have different signaling formats on each end, set the transmitBusyYellow attribute to enable the FP to send a busy ABCD-signaling state to the connected PBX for this voice channel whenever the voice path is down.

set vs/<voice_service_number> framer transmitBusyYellow yes

9 For connections that have different signaling formats on each end, set the transmitCasYellow attribute to enable the FP to transmit a yellow alarm condition to the connected PBX for this voice channel whenever the voice path is down.

set vs/<voice_service_number> framer transmitCasYellow yes

10 For connections that have different signaling formats on each end, set the idleCode attribute on each FP.

set vs/<voice_service_number> framer idleCode <idleCode_value>

11 For connections that have different signaling formats on each end, set the seizeCode attribute on each FP.

set vs/<voice_service_number> framer seizeCode <seizeCode_value>

12 For CAS connections that have different signaling formats on each end, set the aLawConversion attribute on the trunk card to allow conversion from DS1 mu-Law to E1 A-law.

set vs/<voice_service_number> framer aLawConversion on

13 Verify that the default values of the attributes under the Framer component meet network requirements, and make changes as necessary.

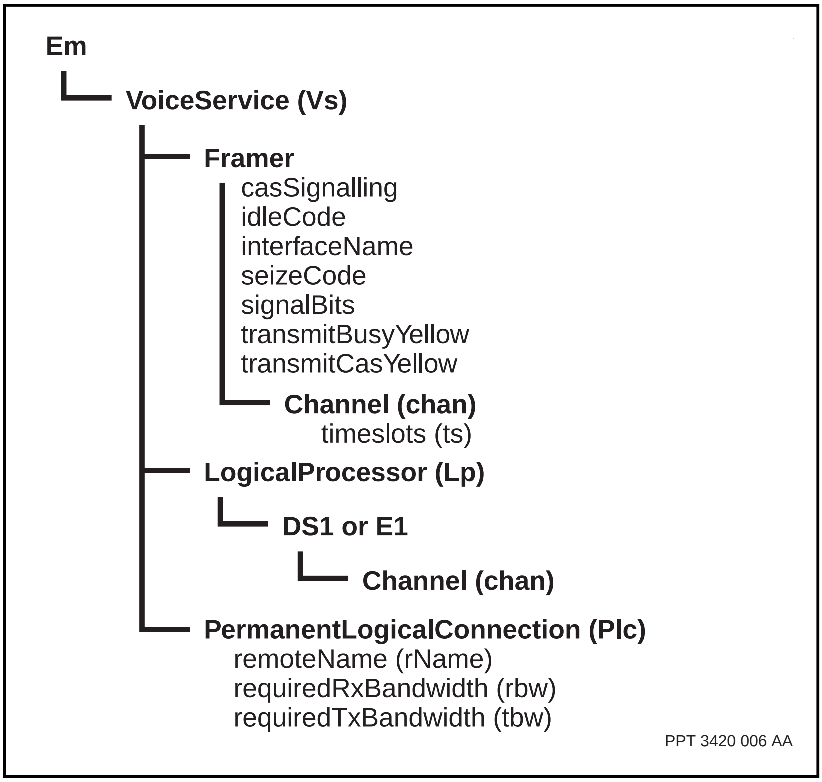
14 Verify that the default values of the attributes under the Plc component meet network requirements, and make changes as necessary.

Variable definitions

Variable	Definition
<casSignalling_value>	transparent (when both ends of the connection are using the same signaling format), interpret (when each end of the connection is using a different signaling format), or none (for CCS signaling)
<channel_number>	The instance number of the channel.
<idleCode_value>	The idleCode value, for example: a 1 b 1 c 0 d 1
<lp_number>	The instance number of the LP.
<port>	ds1 or e1
<port_number>	The instance number of the port.
<remote_name>	The name of the Voice Service at the far end of the Passport 7400 network.
<rx_value>	32000 (for E1 CAS and TTC CAS), or the required Rx bandwidth, in kbit/s (for DS1 CAS).
<seizeCode_value>	The seizeCode value, for example: a 1 b 1 c 1 d 1
<signalBits_value>	abcd (for DS1 ESF CAS or E1 CAS), a for TTC CAS, ab (for DS1 SF (D4) CAS)
<tx_value>	32000 (for E1 CAS and TTC CAS), or the required Tx bandwidth, in kbit/s (for DS1 CAS).
<voice_service_number>	The instance number of the Voice Service.

Procedure job aid

Figure 7
Configuring the Voice Service component hierarchy



Chapter 4
Voice Transport configuration considerations

This section contains information on the details you need to consider when configuring the Voice Transport service, as well as some configuration checklists.

End-to-end negotiation provisioning guidelines and considerations

By using the *VoiceService Framer* component's group of provisionable attributes, you define how to encode and send audio data across a Voice Transport connection. To successfully establish a Voice Transport connection, the source and destination function processors (FP) must verify during end-to-end negotiation that they have compatible *Framer* component provisioning information. In terms of provisioning, end-to-end negotiation

- discovers encoding and signaling incompatibilities between FPs, rejects the connection attempt, and updates the VoiceService component's serviceFailureReason operational attribute
- allows FPs to modify certain provisioned values to successfully establish a connection

Other provisioning problems are handled by way of warnings or semantic check errors when you attempt to set attributes or check your provisioning.

Some encoding types require that you provision a corresponding feature under the *featureList* attribute (see table "Relationship between encoding types and FPs" (page 42)). In other cases, an MVP-E FP supports an encoding type but modifies certain provisioned rate values. Also, if you plan to deploy

Voice Transport in an interworking environment, you must be aware of the impacts certain encoding types and features have when you provision them (see “Interworking provisioning considerations” (page 46)).

Table “Relationship between encoding types and FPs” (page 42) lists the available encoding types for voice, modem/fax, and fax traffic, the FPs that support each encoding type, and supported rates.

See the following sections for information on setting other *Framer* component provisionable attributes:

- “Speech activity detection and fax idle suppression provisioning considerations” (page 43)
- “Tandem pass through provisioning considerations” (page 44)
- “DTMF detection/regeneration provisioning considerations” (page 45)

Table 1
Relationship between encoding types and FPs

voiceEncoding attribute value	**FP type**	**featureList attribute value**	**Supported rate(s) in kbit/s**
g711G726	MVP-E	N/A	24, 32, 64
g728at16	MVP-E	g728	16
g729at8	MVP-E	g729	8
modemFaxEncoding attribute value	**FP type**	**featureList attribute value**	**Supported rate(s) in kbit/s**
g711G726	MVP-E	N/A	32, 64
(Sheet 1 of 2)			

Table 1 (continued)
Relationship between encoding types and FPs

voiceEncoding attribute value	FP type	featureList attribute value	Supported rate(s) in kbit/s
faxRelayG711G726	MVP-E	faxRelay	Depends on the traffic type. For modem/fax traffic, the rates correspond to those negotiated for g711, g711g726, or g726 encoding (see the *modemFaxEncoding* attribute value g711G726 for possible rates). For fax traffic, the maximum possible rate is used once the MVP-E detects a fax preamble tone (see the *modemFaxEncoding* attribute value faxRelayOnly for possible rates).
useVoiceEncoding (see Note:)	MVP-E	N/A	N/A
faxRelayOnly (see Note:, Note 1:)	MVP-E	faxRelay	2.4, 4.8, 7.2, 9.6

Note: When you set the *modemFaxEncoding* attribute to useVoiceEncoding, Voice Transport handles modem/fax and fax traffic the same as voice traffic. However, to support modem and facsimile calls, the negotiated encoding for voice traffic must be g711, g711g726, or g726. If the negotiated encoding is g728 or g729, modem and facsimile calls are not supported.

Note: When you set the *modemFaxEncoding* attribute to faxRelayOnly, Voice Transport does not support modem calls.

Note 1: 4-port MVP-E FPs only support V.17 fax relay rates of 14.4 and 12.0 kbit/s.

(Sheet 2 of 2)

Speech activity detection and fax idle suppression provisioning considerations

MVP-E FPs support speech activity detection and fax idle suppression. For calls between MVP-E FPs or for interworking calls between voice and MVP-E FPs, you require, at a minimum, Passport R5.1 software for speech activity detection and fax idle suppression to work.

To enable speech activity detection, you provision the *silenceSuppression* attribute. The setting of the *silenceSuppression* attribute applies to voice traffic. The *silenceSuppression* attribute can have one of the following values:

- off specifies that silence suppression is never applied.
- on, the default setting, specifies that silence suppression is always applied.
- congested, specifies that silence suppression is applied only when the network is congested.
- slow specifies that silence suppression is applied after a certain period of silence has elapsed. For MVP-E FPs, it is applied after 20 seconds.
- slowAndCongested specifies that silence suppression occurs according to the slow and congested options.
- casIdleCode, which only applies to voice FPs, specifies that the channel only goes into silence suppression after the idle code is active for 20–40 seconds. If casIdleCode is specified, the casSignalling attribute must be set to interpret.

End-to-end negotiation determines the supported setting of the *silenceSuppression* attribute.

To enable fax idle suppression, you provision the *faxIdleSuppressionG711G726* attribute. The setting of the *faxIdleSuppressionG711G726* attribute applies to fax traffic. The *faxIdleSuppressionG711G726* attribute can be set to on (the default value) or off. For fax idle suppression to function, the source and destination FPs must have the *modemFaxEncoding* attribute set to g711G726 and the *faxIdleSuppressionG711G726* attribute must be set to on. End-to-end negotiation determines if fax idle suppression is supported by the source and destination FPs.

Tandem pass through provisioning considerations

MVP-E FPs support tandem pass through (TPT). To enable TPT, you provision the *tandemPassThrough* attribute. For TPT to operate,

- you require Passport R5.1 or later software on the source and destination Passport nodes and the tandem Passport nodes on either side of the intermediate PBX

- you must add the value 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 end-to-end negotiation must determine that enabled is the supported setting
- the voiceEncoding attribute on the tandem nodes must be provisioned with the same compression algorithm. the voiceEncoding attribute on the tandem nodes must be provisioned 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
  - 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 destination node. You adjust the jitter buffer by provisioning the *insertedOutputDelay* attribute. For more information about network jitter, see "Configurable egress buffer" (page 76).

DTMF detection/regeneration provisioning considerations

MVP-E FPs support DTMF detection and regeneration. To enable DTMF detection and regeneration, you provision the *dtmfRegeneration* attribute. This attribute specifies whether to convert DTMF tones to actual digits before transporting them across the subnet or to transport DTMF tones transparently. The *dtmfRegeneration* attribute only affects DTMF tones generated after voice call establishment (for example, when pressing digits to access voice mail). Because voice calls can experience many compression and decompression cycles while crossing a network, the quality of both the voice signal and DTMF tones can be negatively impacted. By enabling the detection, conversion, and regeneration of DTMF tones, you can preserve the quality of DTMF tones.

When end-to-end negotiation determines that on is the supported setting for the *dtmfRegeneration* attribute, DTMF tones received from an incoming PBX are

- converted to a digit at the source node
- transported across the subnet
- regenerated as the appropriate DTMF tone by the destination Passport node

The source and destination MVP-E FPs must have *dtmfRegeneration* set to on for DTMF regeneration to operate. The value on is the recommended setting for voice applications using ITU-T G.728 and G.729 or G.729A encoding. When set to off (the default setting), DTMF tones are transported transparently across the subnet. For calls between a voice and MVP-E FP, end-to-end negotiation disables DTMF regeneration, regardless of the value provisioned under the *dtmfRegeneration* attribute.

Interworking provisioning considerations

As with any Voice Transport connection, a MVP-E FP must have compatible provisioning data. The following sections describe the differences between the encoding types, rates, and capabilities supported by voice and MVP-E FPs:

- “Voice encoding” (page 47)
- “Modem/fax encoding” (page 47)
- “Dynamic up- and down-speeding” (page 48)

See table “Voice Transport capability-to-card relationships” (page 91) for information on the capabilities supported by MVP-E FPs. Refer to table “Relationship between encoding types and FPs” (page 42) for the voice, modem/fax, and fax traffic encoding types and rates supported by MVP-E FPs.

For more information about provisioning interworking connections, see the section “Provisioning an interworking data call connection” (page 49).

Voice encoding

MVP-E FPs support the g728at16 and g729at8 voice encoding values under the *voiceEncoding* attribute. You can provision the g728 and g729 features only on MVP-E FPs. If a voice FP receives g728at16 or g729at8 during end-to-end negotiation, then a compatibility error occurs. In this case, the *Vs* component’s *serviceFailureReason* operational attribute contains the compatibility error mismatchedVoiceEncoding.

MVP-E FPs support the voice encoding value g711G726. However, MVP-E FPs only support rates of 24, 32, and 64 kbit/s. For MVP-E handling of G.711/G.726 voice encoding rates, use the information in table “Relationship between encoding types and FPs” (page 42) as a guideline.

Table 2
MVP-E handling of g711G726 voice encoding rates

<table>
<tr><th>Connection request source</th><th>minVoiceG711-G726Rate</th><th>maxVoiceG711-G726Rate</th><th>Details</th></tr>
<tr><td rowspan="4">MVP-E FP</td><td>16</td><td>16</td><td>Unsupported. You cannot set the maxVoiceG711G726Rate attribute to 16 kbit/s on an MVP-E FP. In this case, a semantic check error occurs during provisioning.</td></tr>
<tr><td>16</td><td>24</td><td rowspan="3">The MVP-E modifies the value of the minVoiceG711G726Rate attribute to 24 kbit/s prior to sending end-to-end negotiation information to the voice FP on the destination Passport node.</td></tr>
<tr><td>16</td><td>32</td></tr>
<tr><td>16</td><td>64</td></tr>
</table>

Modem/fax encoding

MVP-E FPs support the faxRelayOnly and faxRelayG711G726 modem/fax encoding values under the *modemFaxEncoding* attribute. You can only provision the faxRelay feature on an MVP-E FP. If a voice FP receives the value faxRelayOnly or faxRelayG711G726, end-to-end negotiation results in a compatibility error. In this case, the *Vs* component’s *serviceFailureReason* operational attribute contains the compatibility error mismatchedModemFaxEncoding. MVP-E FPs support the modem/fax encoding value g711G726. However, MVP-E FPs only support the following rates: 32 and 64 kbit/s. For MVP-E handling of G.711/G.726 modem/fax encoding rates, use the information in table “MVP-E handling of g711G726 modem/fax encoding rates” (page 48) as a guideline.

Table 3
MVP-E handling of g711G726 modem/fax encoding rates

Connection request source	minModemFax-G711G726Rate	maxModemFax-G711G726Rate	Details
MVP-E FP	16	16	Unsupported. You cannot set the *maxModemFaxG711G726Rate* attribute to 16 or 24 kbit/s on an MVP-E FP. In this case, a semantic check error occurs during provisioning.
	16	24	
	24	24	
	16	32	The MVP-E modifies the value of *minModemFaxG711G726Rate* to 32 kbit/s prior to sending end-to-end negotiation information to the voice FP on the destination Passport node.
	16	64	
	24	32	
	24	64	

Dynamic up- and down-speeding

MVP-E FPs support dynamic up- and down-speeding of voice, modem/fax, and fax traffic. However, voice and MVP-E FPs support different rates for the voice encoding value g711G726 and the modem/fax encoding values g711G726 and faxRelayG711G726.

For voice traffic, dynamic up- and down-speeding can occur when the *voiceEncoding* attribute is set to g711G726 and different rates are provisioned under the *minVoiceG711G726Rate* and *maxVoiceG711G726Rate* attributes. For modem/fax and fax traffic, dynamic up- and down-speeding can occur when the *modemFaxEncoding* attribute is set to g711G726 and different rates are provisioned under the *minModemFaxG711G726Rate* and *maxModemFaxG711G726Rate* attributes. However, end-to-end negotiation between a voice and MVP-E FP can result in the modification of the provisioned minimum and maximum rate values. To provision dynamic up-and down-speeding for voice, modem/fax, and fax traffic between MVP-E FPs, use the information contained in table “MVP-E handling of g711G726 voice encoding rates” (page 47) and table “MVP-E handling of g711G726 modem/fax encoding rates” (page 48).

Provisioning an interworking data call connection

You can provision Voice Transport to support a data call connection between a voice and MVP-E FP. Both the voice and MVP-E FP’s *Vs Framer* component must be provisioned with the following values:

> ***Note:*** There is no interworking between voice FPs and 4-port MVP-E FPs

MVP-E FP’s *Vs Framer* component must be provisioned with the following values:

```
voiceEncoding = g711G726
minVoiceG711G726Rate = 64 kbit/s
maxVoiceG711G726Rate = 64 kbit/s
modemFaxEncoding = g711G726
minModemFaxG711G726Rate = 64 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s
echoCancellation = off
silenceSuppression = off
faxIdleSuppressionG711G726 = off
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
```

Configuration checklists

To avoid having problems establishing a Voice Transport connection, refer to the following checklists:

- “Installation” (page 50)
- “Network clock synchronization” (page 51)
- “Provisioning Voice Transport with default values” (page 51)
- “Changing default values” (page 51)

Installation

When you installed Voice Transport software, did you

- install PORS and Voice Transport according to the instructions in 241-5701-270 *Passport 7400, 15000, 20000 Software Installation Guide*?
- check that the same version of Voice Transport-compatible software is loaded on every Passport 7400 node?
- add Trunk Pa components to every Passport 7400 node that could be a path candidate?

Network clock synchronization

When you provisioned network clock synchronization, did you

- identify a master clock reference?
- plan and establish a tree structure of nodes; each synchronizing their clock signal with the last, branching back to the master reference?
- check for synchronization loops?

For information and procedures on configuring network clock synchronization, see 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide*.

Provisioning Voice Transport with default values

When provisioning Voice Transport using the default values, did you

- link the Framer component to the hardware? Did you use the correct logical-processor value and port number? Is the syntax correct?
- use the remoteName attribute to identify the other end of the connection? Did you use the exact node name and correct syntax?
- display your provisioning and check it for errors? Did you check your spelling?
- use the check, save, activate and confirm commands?
- provision both ends of the connection?

Changing default values

When changing Voice Transport's default values, did you

- ensure that sufficient bandwidth is available? (Remember that you are probably sharing the total bandwidth with connectionless routing.)
- provision the attributes under the Trunk component identically at both ends of the connection?
- use the information contained in "End-to-end negotiation provisioning guidelines and considerations" (page 41) and, for connections between voice and MVP-E FPs, "Interworking provisioning considerations" (page 46)?

- refer to 241-5701-060 *Passport 7400, 15000, 20000 Components*? Several provisionable attributes have provisioning dependencies.

Chapter 5
Voice Transport fundamentals

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

What is Voice Transport?

Voice Transport provides an end-to-end digital connection for both voice and non-voice data across a Passport network. Examples of non-voice data include facsimile, modem, and video data.

In order to provide Voice Transport capabilities across a Passport network, network operators have to deploy service-related hardware and software components.

The Voice Transport function processors and related software modules provide the end-to-end high priority, reserved path, non-reordering, non-duplicating, data delivery capability that is required for voice signals.

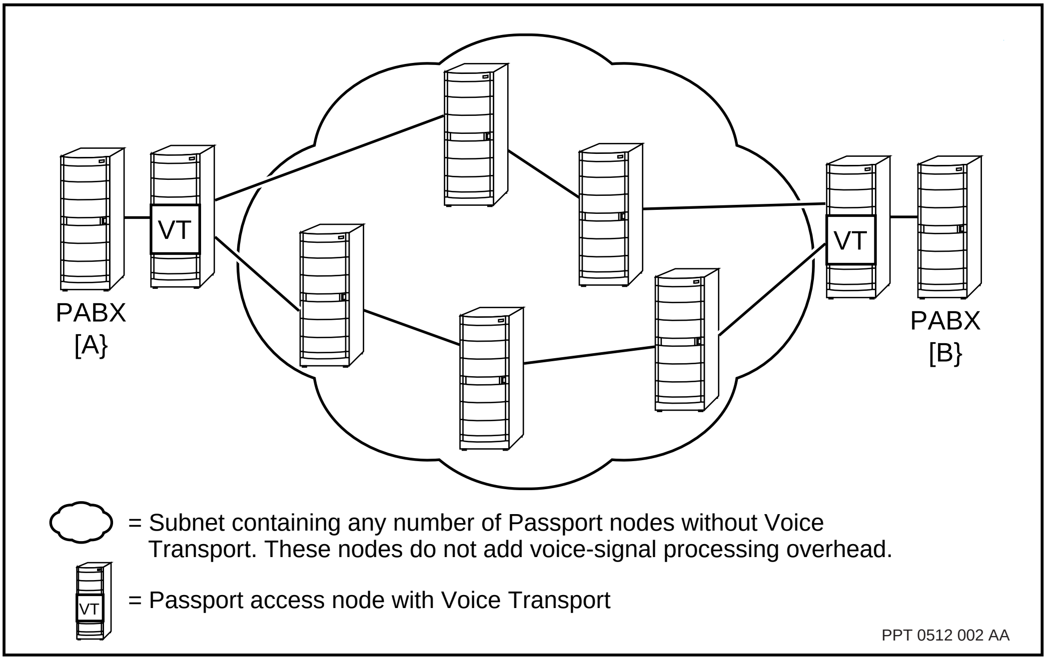
Where is Voice Transport needed?

Voice Transport can be implemented transparently in a voice and non-voice data environment to provide traffic concentration on a Passport subnet.

Voice Transport hardware and software is required on any Passport node that is directly connected to a Private Automatic Branch Exchange (PABX) or similar, customer-supplied equipment to provide access to voice transport services across a network. Voice Transport hardware and software is not required on a Passport 7400 node that is not directly connected to a PABX or similar customer-supplied equipment.

A typical implementation example would be the deployment of Voice Transport to eliminate dedicated facilities between two PABXs. Each PABX would be connected to a Passport 7400 access node with Voice Transport capabilities; the traffic between the two PABXs would then be routed transparently through the Passport 7400 subnet. See figure “Voice Transport on a Passport 7400 network” (page 55).

Figure 8
Voice Transport on a Passport 7400 network



What systems and hardware does Voice Transport require?

Voice Transport requires the Path Oriented Routing System (PORS) and network clock synchronization to be operational. PORS routes data along a predetermined end-to-end path to maintain packet ordering and to minimize delay variations. Network clock synchronization provides network-wide clock synchronization to ensure accurate transmission and reproduction of data throughout the network. For more information see "Voice Transport, PORS, and route selection" (page 95).

The Voice Transport service operates on the following hardware platforms:

- DS1 MVP enhanced echo cancellation (DS1 MVP-E), E1 MVP-E, and TTC2M MVP-E FPs
- 4-port DS1 MVP enhanced echo cancellation (4-port DS1 MVP-E) and 4-port E1 MVP-E

For a description of the hardware elements of the voice, MVP, and 1 and 4-port MVP-E FPs, see 241-7401-200 *Passport 7400 Hardware Description*.

What are Voice Transport's key capabilities?

The Voice Transport service's capabilities include

- end-to-end negotiation between function processors (FPs) to establish connections
- interworking between voice and MVP-E FPs. Interworking between voice and MVP-E FPs is based on specific features and dependencies. See "System parameters of Voice Transport" (page 90) for details.
- separate provisionable encoding rates for voice, modem, and facsimile data
- conservation of bandwidth using speech activity detection (SAD)—with provisionable support for capping the generation of background or comfort noise—and fax idle suppression (FIS)
- provisionable support for the prevention of clipping for speech calls and facsimile transmissions
- echo cancellation on all voice channels
- bidirectional A-Law (international) to mu-Law (North American) conversion of voice data
- congestion management (dynamic up- and down-speeding of voice traffic)
- tandem node detection for routing compressed voice calls through an intermediate PBX by way of tandem Passport nodes (tandem pass through)
- provisionable support for dealing with cell delay variation across the network (configurable egress buffering)
- provisionable loss/gain adjustment (for network loss planning)
- detection of the 2100 Hz inband tone that identifies modem and facsimile calls, with the ability to recover from the false detection of a 2100 Hz tone (facsimile/speech discrimination)

- transport of channel associated signaling (CAS) with the associated voice channel; transport of common channel signaling (CCS) as transparent data
- discrimination between voice or data calls when using CCS

The preceding capabilities are described in detail in “Voice Transport capabilities and system parameters” (page 65). For specific information on capability-to-FP relationships, see table “Voice Transport capability-to-card relationships” (page 91).

How does Voice Transport work?

The following sections describe the physical and data path Voice Transport connections (including timeslot processing) from PABXs to Passport 7400 nodes and between Passport 7400 nodes:

- “The physical connection” (page 58)
- “The data connection” (page 59)
- “Timeslot data processing” (page 60)

The physical connection

A PABX accesses Voice Transport by means of digital trunks connected to their respective function processor on a Passport 7400 node.

The 1-port MVP-E FPs connect by means of DS1, E1 and TTC digital trunks. The 4-port MVP-E FPs connect only by means of DS1 and E1 digital trunks.

See figure "Connection to the Passport 7400 network" (page 58).

The Voice Transport service on a Passport 7400 node connects to other Passport 7400 nodes (the subnet) over transport options that include V.11, V.35, DS1, E1, TTC, DS3, and E3 digital trunks. See figure "Transport of cell data across the subnet" (page 59).

Figure 9
Connection to the Passport 7400 network

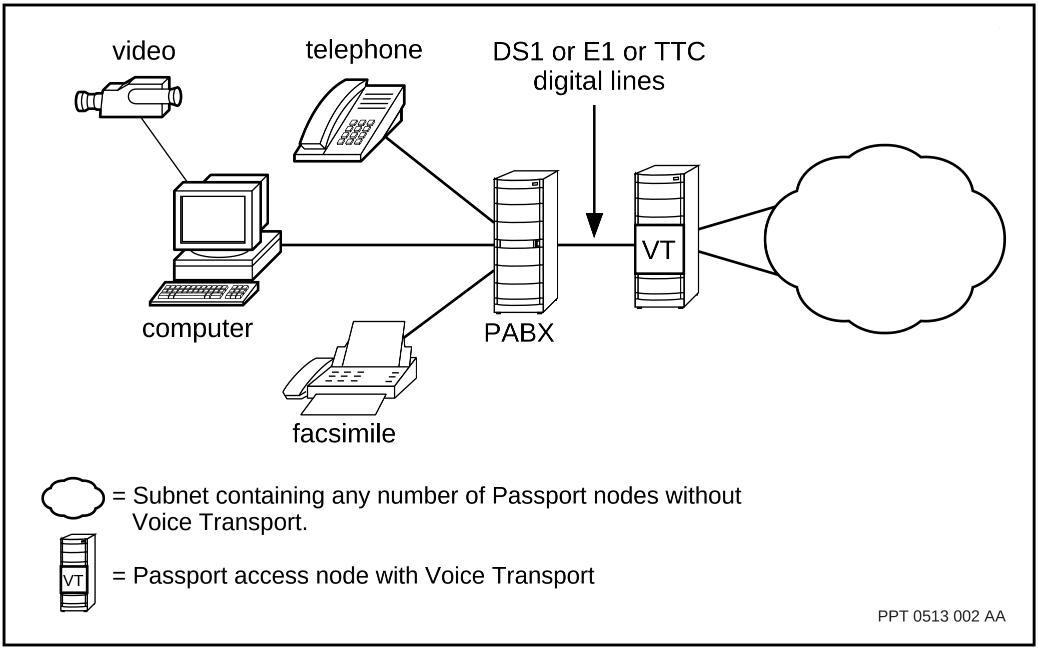
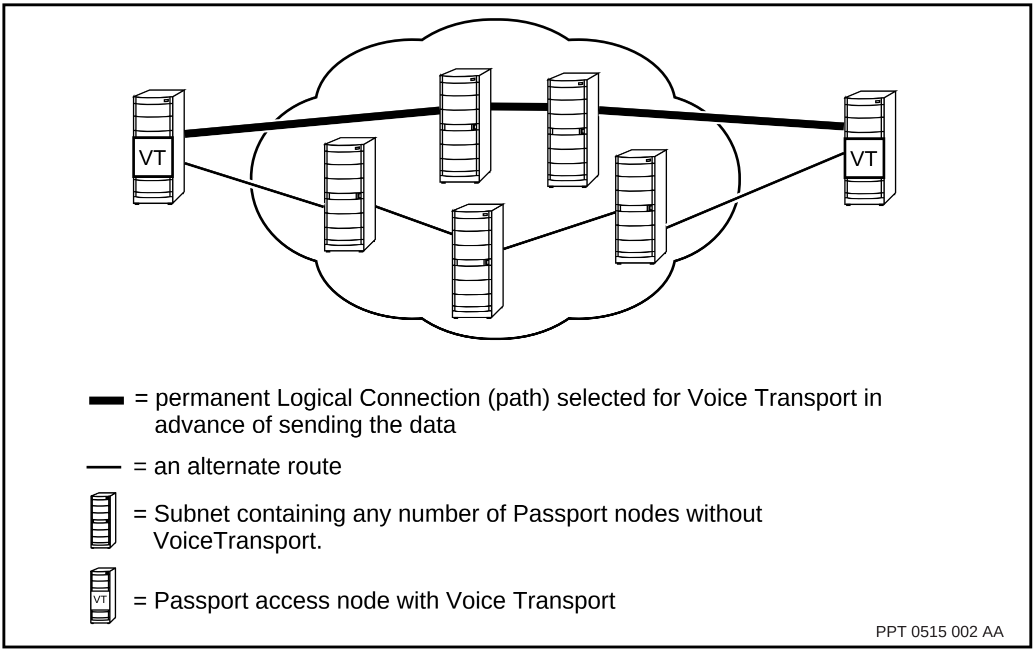


Figure 10
Transport of cell data across the subnet



The data connection

The PABX sends data to and receives data from the Passport 7400 node as streams of serial Time Division Multiplexed (TDM) timeslots of Pulse Code Modulation (PCM) data.

> ***Note:*** See appendix “Data stream refresher” (page 173), for a more detailed description of the relationship between DS1, E1, and Voice Transport cells.

Before transmitting data, the Path Oriented Routing System (PORS) establishes a communications path between the sending and receiving nodes. See figure “Transport of cell data across the subnet” (page 59). Within the path, Voice Transport gets the bandwidth required by the data to be transmitted. PORS carries out bandwidth reservation, guarantees packet ordering, and minimizes delay variations across the network. Bandwidth is instantly available for connectionless traffic if the reserved bandwidth is not fully utilized by Voice Transport path-oriented traffic.

Timeslot data processing

Voice Transport processes the timeslot data from a PABX according to capabilities that you provision (for more information on Voice Transport audio handling capabilities, see “Voice Transport capabilities and system parameters” (page 65))

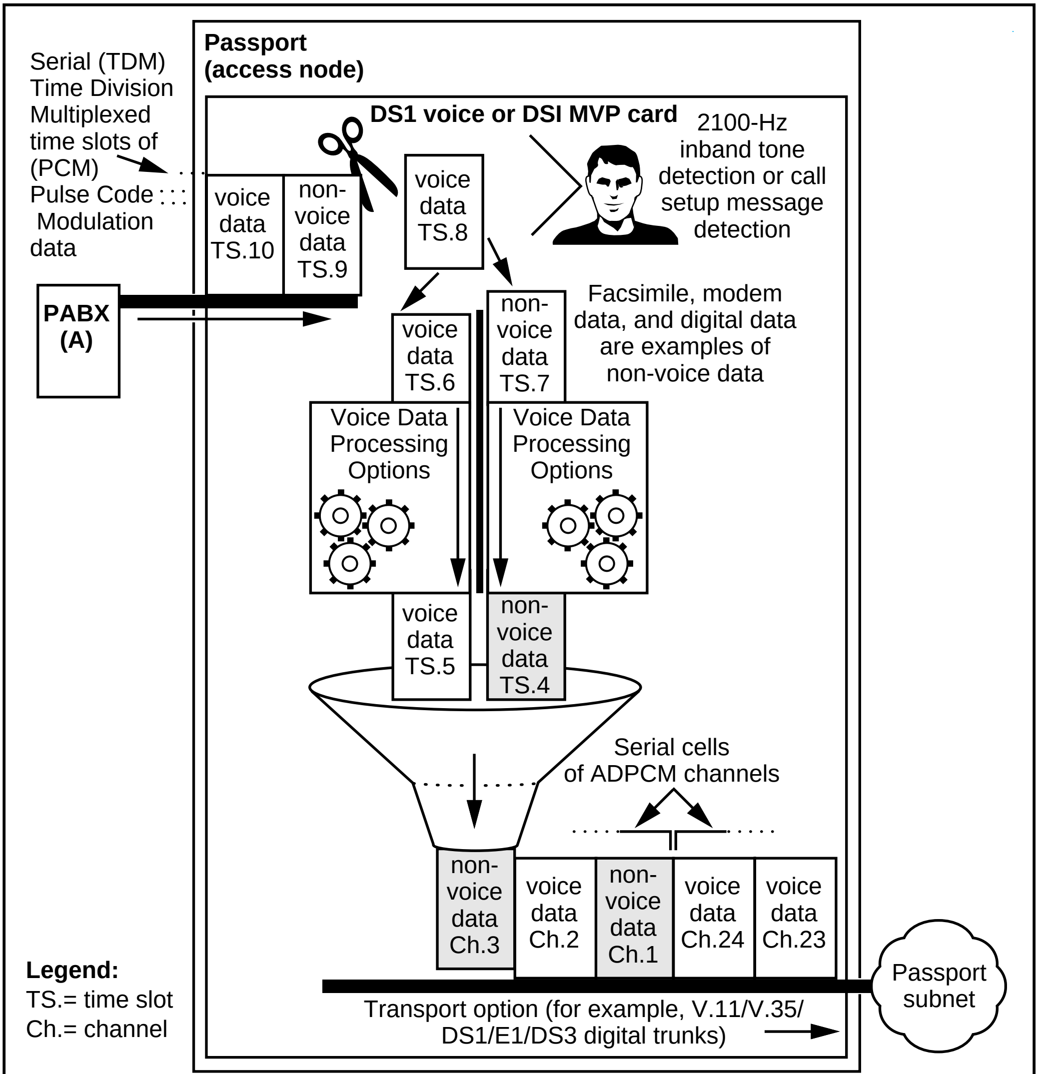
. The TDM timeslot data from the PABX is distributed over many fixed-length cells. These cells are sent across the Passport 7400 subnet. See figure “Timeslot to cell conversion within Voice Transport” (page 61).

Voice Transport cells are composed of channels. The channel assignments within any one cell can be

- all voice data
- all non-voice data (for example, facsimile and modem)
- a combination of voice and non-voice data

At the destination node of the Passport 7400 network, the cells of data that were transported across the subnet are converted to timeslot data. Voice Transport recreates the individual timeslots of data and delivers them to the receiving PABX over a DS1, E1, or TTC digital line. See figure “Conversion of cell data to timeslot data” (page 62).

Figure 11
Timeslot to cell conversion within Voice Transport



Note: This example is based on a 24-channel DS1 signal format without silence suppression or compression. An E1 or a J2M voice signal with up to 30 voice-only channels or a combination of up to 31 voice and non-voice channels follows the same general concepts.

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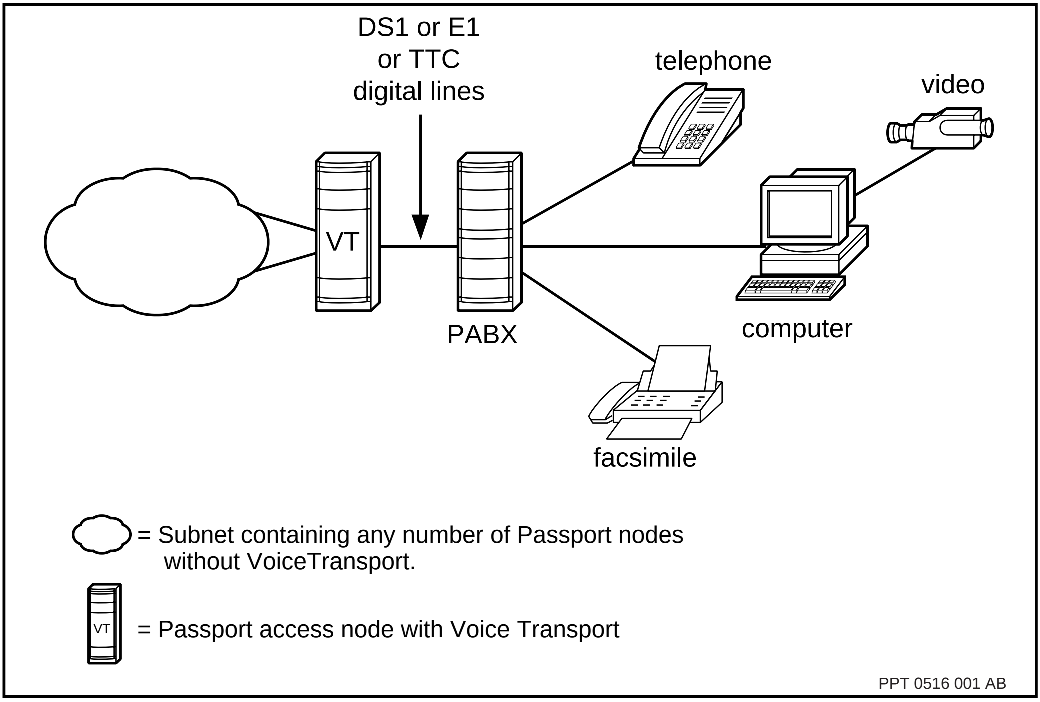
Figure 12
Conversion of cell data to timeslot data



Voice Transport standards compliance

Voice Transport conforms to certain sections of the ITU-T standards shown in table “ITU-T standards for Passport 7400 Voice Transport” (page 63). The Passport 7400 Voice Transport service also corresponds to the TTC standards shown in table “TTC standards for Passport 7400 Voice Transport” (page 63), which apply to TTC2M MVP-E FPs only.

Table 4
ITU-T standards for Passport 7400 Voice Transport

Description	ITU-T standard
Echo cancellation	G.164, G.165 and G.168 (depending on the FP
Physical/electrical interface characteristics	G.703
mu-Law/A-Law coding	G.711
ADPCM compression at 16, 24, and 32 kbit/s	G.726 (includes G.721 and G.723)
LD-CELP compression at 16 kbit/s	G.728
CSA-CELP compression at 8 kbit/s	G.729 or G.729A
Fault conditions and consequent actions	G.732
Fax modulation and demodulation (fax relay)	based on T.30, V.21, V.27 and V.29
Modems	up to V.34

Table 5
TTC standards for Passport 7400 Voice Transport

Description	TTC standard
Digital interface between PBX and TDM (channel-associated signaling) outline.	JJ-20.10
Digital interface between PBX and TDM (channel-associated signaling) PBX and PBX signaling specification.	JJ-20.12
Digital interface between PBX and TDM (channel-associated signaling) Electrical and Physical condition.	JJ-20.11
Note: For more information on channel associated signaling (CAS), see appendix “Signalling refresher” (page 177).	

Chapter 6
Voice Transport capabilities and system parameters

The following sections describe the capabilities and system parameters of Voice Transport:

- "End-to-end negotiation" (page 65)
- "Transport of voice traffic" (page 68)
- "Transport of modem/fax and fax traffic" (page 76)
- "Configurable egress buffer" (page 76)
- "Supported signaling protocols and related information" (page 85)
- "System parameters of Voice Transport" (page 90)

Use this information to establish Voice Transport services that take full advantage of your system's resources.

Some Voice Transport features are only available on a particular function processor (FP). For more information on feature-to-card dependencies and interworking between FPs, see "System parameters of Voice Transport" (page 90) and "Interworking provisioning considerations" (page 46).

End-to-end negotiation

When Voice Transport attempts to establish a connection between two FPs running Passport R5.1 or later software, end-to-end negotiation occurs. During end-to-end negotiation, FPs exchange provisioning information. A Voice Transport connection attempt succeeds when the source and destination

FPs verify that they have compatible provisioning data. See “Relationship between call types, traffic types and encoding choices” (page 67) for information about how Voice Transport handles audio traffic.

In some cases, the end-to-end negotiation process causes an FP to modify specific provisioned values (see “End-to-end negotiation provisioning guidelines and considerations” (page 41)). If the end-to-end negotiation process finds that the provisioning data for both FPs is not compatible, the connection fails. The operational attribute *serviceFailureReason*, found under the *VoiceService* (*Vs*) component, contains the reason the connection failed. For example, if the *casSignalling* attribute on the source and destination FPs specify, respectively, interpret and transparent, the connection fails. In this instance, the *Vs* component’s *serviceFailureReason* attribute on both Passport nodes contains the compatibility error mismatchedCasSignalling. For more information on failed connection attempts, see “remoteName attribute” (page 66).

A series of operational attributes under the *Vs Framer* component’s *Negotiation* group indicate the results of end-to-end negotiation. Negotiated values include, for example, the encoding choices and rates for particular traffic types. For more information, see “Monitoring and troubleshooting Voice Transport” (page 159).

remoteName attribute

The *Vs PermanentLogicalConnection* (*Plc*) component’s *remoteName* attribute has an important role in the end-to-end negotiation process. If the *Vs Plc* component’s *remoteName* attribute is set incorrectly (for example, because of improper syntax), a connection request is rejected even if the source and destination FPs share compatible provisioning data. Alternately, if you do not specify a value for the *remoteName* attribute, the *Vs Plc* component accepts connection requests from any remote *Vs Plc* component. However, with no value specified for the *remoteName* attribute, the *Vs Plc* component cannot originate a connection request and the connection establishment process is slower. See 241-5701-060 *Passport 7400, 15000, 20000 Components* for more information about setting the *remoteName* attribute.

Relationship between call types, traffic types and encoding choices

Voice Transport supports three audio call types: voice, modem, and facsimile. Table "Example of relationship between Voice Transport call, traffic, and encoding types" (page 68) describes, by way of an example call sequence, the three call types in relation to each traffic type—voice, modem/fax, and fax. The traffic type defines which of the *VoiceService (Vs) Framer* component's negotiated encoding and rate values applies to each call type as a call progresses.

Initially, all Voice Transport audio calls are treated as voice traffic. That is, MVP-E FPs always process traffic according to the negotiated value of the *voiceEncoding* attribute. Idle periods between calls are also handled as voice traffic. A 2100 Hz inband tone identifies the start of a modem or facsimile transmission. Upon detection of a 2100 Hz tone, Voice Transport handles traffic as modem/fax traffic; that is, according to the negotiated value of the *modemFaxEncoding* attribute. If a fax preamble (identifying the start of a facsimile transmission) follows a 2100 Hz tone, traffic is handled as fax traffic, according to the negotiated value of the *modemFaxEncoding* attribute.

Table 6
Example of relationship between Voice Transport call, traffic, and encoding types

Event	Call type/ status	Treated as...	Vs Framer provisionable attribute	Vs Framer Negotiation operational attribute
Start of call	Facsimile	voice traffic	voiceEncoding	negotiatedIgEncoding negotiatedIgRates
2100 Hz tone	Facsimile	modem/fax traffic	modemFaxEncoding	
Fax preamble	Facsimile	fax traffic	modemFaxEncoding	
Call release	Idle	voice traffic	voiceEncoding	
Start of call	Modem	voice traffic	voiceEncoding	
2100 Hz tone	Modem	modem/fax traffic	modemFaxEncoding	
Call release	Idle	voice traffic	voiceEncoding	
Start of call	Voice	voice traffic	voiceEncoding	
Call release	Idle	voice traffic	voiceEncoding	

Transport of voice traffic

Transport of voice traffic consists of the following audio handling capabilities:

- “Echo cancellation” (page 69)
- “Speech activity detection (SAD)” (page 70)
- “Provisionable voice traffic encoding rates” (page 72)
- “Mu-law/A-law translation” (page 73)
- “Gain control and adjustment” (page 74)
- “Tandem pass through (TPT)” (page 74)
- “Configurable egress buffer” (page 76)

Echo cancellation

Echo is reflected speech energy. Echo is audible and objectionable when a person's voice is reflected with sufficient strength and noticeable round-trip delay. The amount of round-trip delay, measured in milliseconds, depends on the distance between the point of transmission (for example, the telephone) and the point of reflection (for example, the PBX).

Voice Transport's echo cancellation capabilities vary according to the type of function processor (FP) used (see "Echo cancellation implementations on each FP type" (page 69)). Voice Transport cancels echo on a per-channel basis at the ingress and egress points of the network (see figure "Echo cancellation" (page 70)) and maintains acceptable speech quality by

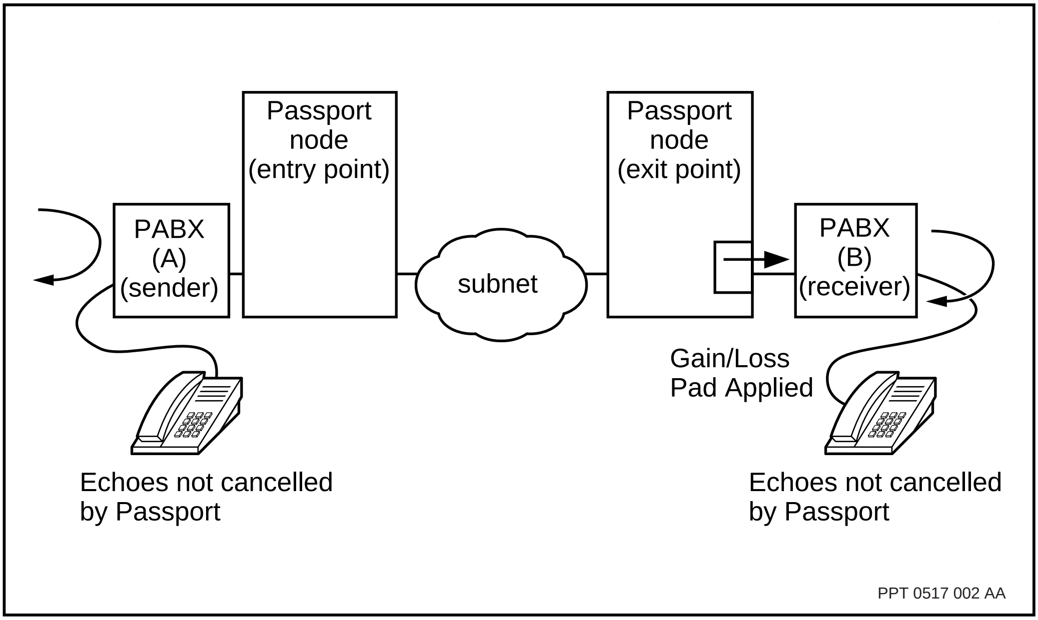
- matching the characteristics of the echo-generating hybrid, the typical source of echo. Hybrid refers to the points outside the Passport network where conversions from 2- to 4-wire or digital connections occur.
- reducing the energy of reflected far-end speech. The reduction of echo is known as attenuation.
- replacing the echo signal with background or comfort noise whenever it falls below a set threshold level

Non-voice cell streams, such as those comprised of modem, facsimile, and digital data, bypass echo cancellation. Modem and facsimile data bypass echo cancellation by using a 2100 Hz disabling tone.

Echo cancellation implementations on each FP type

The on-board echo cancellers on MVP-E FPs provide echo cancellation according to ITU-T Recommendations G.164 and G.165 and provide G.168 audio quality. MVP-E FPs provide enhanced echo cancellation capabilities, including fast convergence time and effective double talk detection and handling. MVP-E FPs also allow you to define echo tail delay parameters, depending on the coverage required in your network, and echo return loss parameters, according to line quality.

Figure 13
Echo cancellation



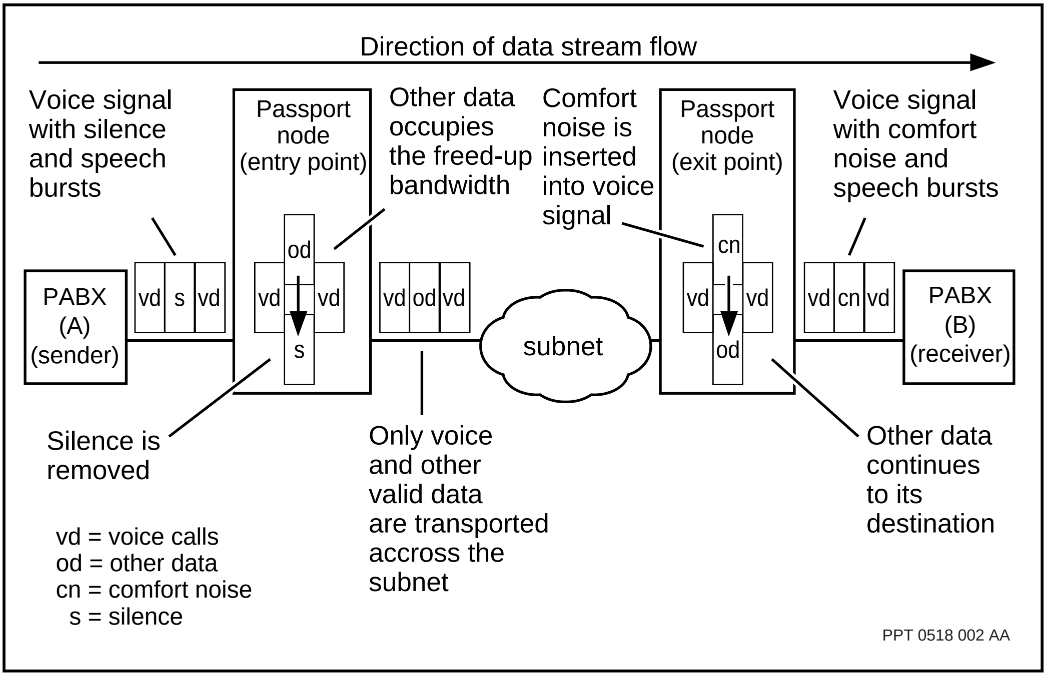
Speech activity detection (SAD)

MVP-E FPs allow you to provision Voice Transport to suppress the silent portions of a conversation. When speech activity detection (SAD) is operational, you can conserve bandwidth within the Passport subnet (see figure “Speech activity detection” (page 71)).

With SAD enabled, Voice Transport suppresses the silence between bursts of speech in the voice signal entering the Passport 7400 subnet. In other words, when SAD is enabled there is no transmission of cells if there is no speech activity. Upon exit from the subnet, Voice Transport inserts background or comfort noise of the same level and duration as the silent portions of the voice signal that were removed. You can also specify on a per-channel basis the maximum level of comfort noise you want to generate. You can also configure SAD to operate under certain network conditions. For example, you can specify that SAD operate only after 20 seconds of silence occurs or only when network congestion occurs, or a combination of both. For more information, see “Congestion management” (page 81). End-to-end negotiation determines the supported type of silence suppression to be used for voice traffic.

On MVP-E FPs, you can prevent SAD from cutting off or clipping parts of a conversation when the negotiated setting for SAD is on, congested, or slowAndCongested. The *speechHangoverTime* attribute allows you to specify a certain amount of delay—between 10 and 500 milliseconds—after the end of a speech burst before suppression begins. See “Provisioning notes” (page 155) for information on how to set the *speechHangoverTime* attribute.

Figure 14
Speech activity detection



Provisionable voice traffic encoding rates

Voice and MVP-E FPs can process voice traffic using the encoding rates and compression ratios shown in table “Voice traffic encoding rates and compression ratios” (page 73). Both ends of a Voice Transport connection must be provisioned with the same encoding and compatible rates. Otherwise, calls fail during end-to-end negotiation.

MVP-E FPs can be provisioned to adjust the voice encoding rate to meet network traffic conditions when using adaptive differential pulse code modulation (ADPCM) encoding. If the network becomes congested, the encoding rate is reduced. In other words, the compression ratio is increased until congestion clears. When congestion clears, each channel can up-speed by increasing the encoding rate and reducing the compression ratio. For more information, see “Supported signaling protocols and related information” (page 85).

MVP-E FPs can be provisioned to recover from a change in the voice encoding rate during a speech call. The change in the encoding rate can be caused by a 2100 Hz tone generated by a facsimile machine in the background of a conversation. The detection of a 2100 Hz tone can cause a switch from the current voice encoding rate to the provisioned modem/facsimile encoding rate. When a Voice Transport connection is in voice mode and an MVP-E FP detects speech immediately after detecting a 2100 Hz tone, only a temporary switch to the provisioned modem/facsimile encoding rate occurs. In this situation, the Voice Transport connection reverts to the provisioned voice encoding rate.

Table 7
Voice traffic encoding rates and compression ratios

Encoding rate	Compression ratio	Provisioning details
64 kbit/s (PCM)	1:1 (no compression)	Available on MVP-E FPs when you set the *voiceEncoding* attribute to **g711G726**.
32 kbit/s (ADPCM)	2:1	Available on MVP-E FPs when you set the *voiceEncoding* attribute to **g711G726** and the negotiated value is **g711G726** or **g726**.
24 kbit/s (ADPCM)	2.667:1	
16 kbit/s (G.728)	4:1	Available only on MVP-E FPs when you set the *voiceEncoding* attribute to **g728at16** (you must also provision the **g728** feature under the *Software* component's *featureList* attribute).
8 kbit/s (G.729 or G.729A)	8:1	Available only on MVP-E FPs when you set the *voiceEncoding* attribute to **g729at8** (you must also provision the g729 feature under the *Software* component's *featureList* attribute).

Mu-law/A-law translation

The DS1 MVP-E FP is capable of converting mu-law to A-law and A-law to mu-law. This is required when a path is established with an E1 MVP-E FP. For information on how to provision the *aLawConversion* attribute, see the procedure in "Connecting DS1 ESF CAS and E1 CAS PBX trunks" (page 137). Table "Local compander law for each Voice Transport FP" (page 73) describes the local compander law for each type of FP.

Table 8
Local compander law for each Voice Transport FP

Card type	Local compander law
1pE1Mvpe	A-law
1pDS1Mvpe	mu-law
(Sheet 1 of 2)	

Table 8 (continued)
Local compander law for each Voice Transport FP

Card type	Local compander law
4pE1Mvpe	A-law
4pDS1Mvpe	mu-law
1pTTC2mMvpe	mu-law
(Sheet 2 of 2)	

Gain control and adjustment

To assist with network loss planning, gain control allows you to adjust the signal level of a call through a network.

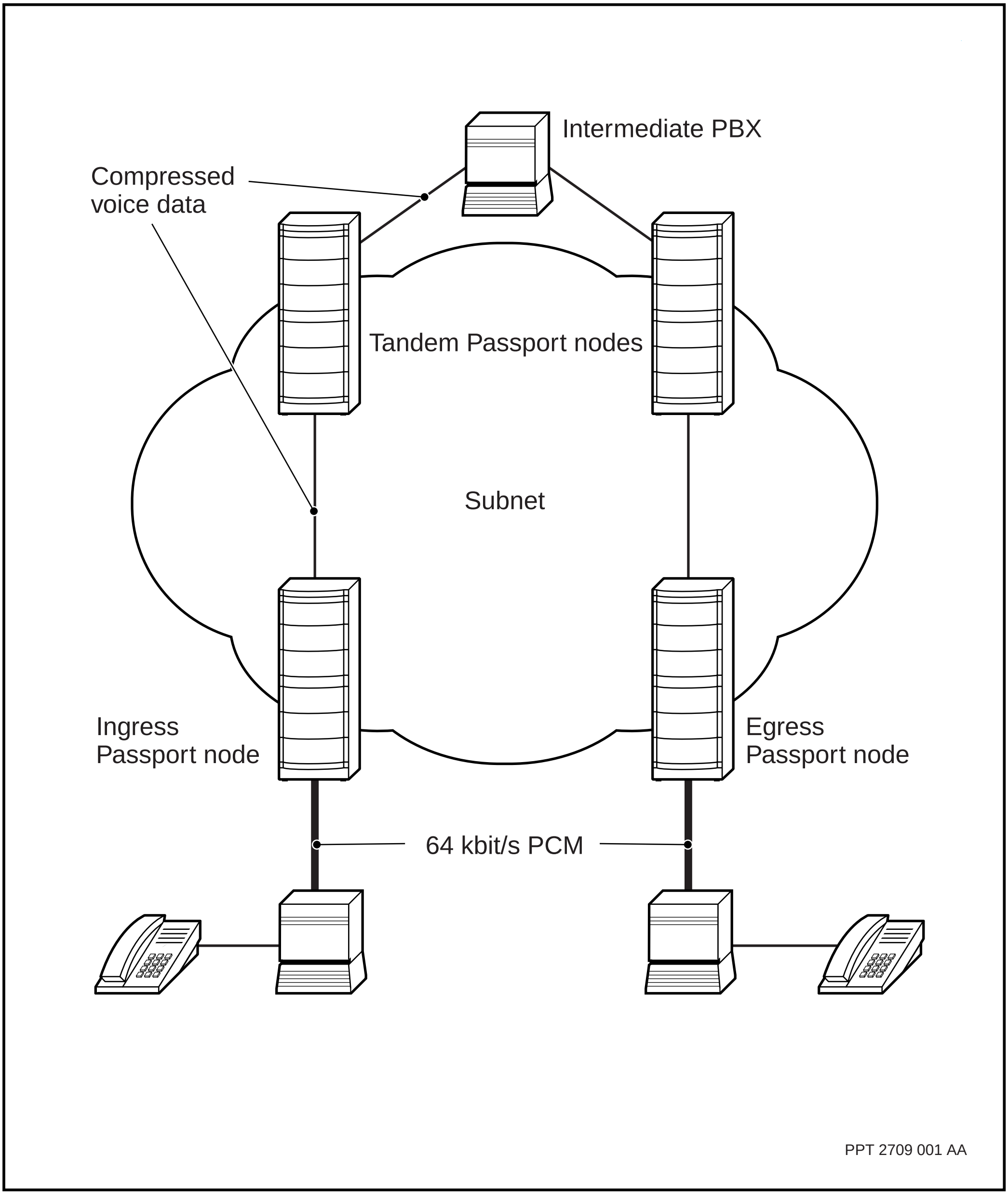
For MVP-E FPs, the outgoing signal level can be adjusted in 1 dB increments between +12 and -12 dB.

MVP-E FPs can also adjust the incoming signal level. Like the outgoing signal level, the incoming signal level can be adjusted in 1 dB increments, between +12 and -12 dB.

Tandem pass through (TPT)

The tandem pass through (TPT) feature operates on MVP-E FPs. TPT allows compressed voice calls to be routed transparently over Voice Transport connections involving an intermediate PBX and tandem Passport nodes (see figure “Voice traffic during tandem pass through mode” (page 75)). The MVP-E FP on the source and destination Passport nodes compresses and decompresses the voice call. With TPT enabled, tandem nodes dynamically detect each other and process, without modification, the compressed voice data. TPT reduces delay and ensures the quality of the voice signal by eliminating the series of compression and decompression cycles normally involved when processing signals over multiple tandem connections. For information on the parameters governing the operation of tandem pass through, see “Tandem pass through provisioning considerations” (page 44).

Figure 15
Voice traffic during tandem pass through mode



Configurable egress buffer

MVP-E FPs allow you to compensate for cell delay variation (CDV), also known as network jitter, by configuring a buffer on the egress Passport node. CDV can negatively effect the quality of the high priority, constant bit rate traffic—voice, modem, and facsimile calls—that Voice Transport carries. You configure the buffer to control the amount of delay that Voice Transport traffic can experience. By controlling the amount of delay at the egress node (measured in milliseconds), traffic can experience variations in the time required to pass through the network without affecting the quality of the signal. By ensuring the quality of the signal, you can, for example, prevent gaps from occurring during a telephone conversation when voice cells crossing the network experience varying amounts of delay.

Transport of modem/fax and fax traffic

Transport of modem/fax and fax traffic consists of the following audio handling capabilities:

- “Fax idle suppression (FIS)” (page 76)
- “Provisionable modem/fax and fax traffic encoding rates” (page 77)
- “Fax relay” (page 79)

An inband 2100 Hz tone distinguishes modem/fax and fax traffic from voice traffic. Modem/fax and fax traffic can also be impacted by network jitter. See “Configurable egress buffer” (page 76) for information on how to control network jitter.

Fax idle suppression (FIS)

MVP-E FPs support FIS. Voice Transport can be provisioned to suppress the idle periods of a facsimile transmission to conserve bandwidth within the Passport subnet.

During a typical facsimile transmission, FIS can reduce bandwidth use by as much as 20% in the sending direction and 80% in the receiving direction. FIS supports the ITU-T group 3 facsimile standard up to 14.4 kbit/s.

A 2100 Hz tone followed by a fax preamble indicates the start of a facsimile transmission. *Voice Transport performs FIS on fax traffic when the negotiated value of the faxIdleSuppressionG711G726* attribute is on *and the negotiated value for the modemFaxEncoding* attribute is g711, g711G726 or g726.

On MVP-E FPs, you can prevent FIS from cutting off or clipping portions of a facsimile transmission when the negotiated setting for FIS is on. The *faxHangoverTimeG711G726* attribute allows you to specify a certain amount of delay—between 300 and 20 000 milliseconds—after the end of a facsimile burst before suppression begins. See "Provisioning notes" (page 155) for information on how to set the *faxHangoverTimeG711G726* attribute.

Provisionable modem/fax and fax traffic encoding rates

MVP-E FPs can process modem/fax and fax traffic at the encoding rates specified in table "Modem/fax and fax traffic encoding rates and compression ratios" (page 78). As with voice encoding, both ends of a Voice Transport connection must be provisioned with the same encoding and compatible rates. Otherwise, calls can fail during end-to-end negotiation.

MVP-E FPs can be provisioned to adjust the encoding rate for modem/fax and fax traffic to meet network traffic conditions when using ADPCM encoding. If the network becomes congested, the encoding rate is down-speeded. In other words, the compression ratio is increased until congestion clears. In the absence of congestion or when congestion clears, the encoding rate can be up-speeded.For more information, see "Supported signaling protocols and related information" (page 85).

Table 9
Modem/fax and fax traffic encoding rates and compression ratios

Encoding rate	Compression ratio	Provisioning details
64 kbit/s (PCM)	1:1 (no compression)	Available on MVP-E FPs when you set the *modemFaxEncoding* attribute to g711G726.
32 kbit/s (ADPCM)	2:1	Available on MVP-E FPs when you set the *modemFaxEncoding* attribute to g711G726 and the negotiated value is **g711G726** or **g726**.
V.27/V.29 fax relay (supported rates include 2.4, 4.8, 7.2, and 9.6 kbit/s)	variable	Available only on MVP-E FPs when you set the *modemFaxEncoding* attribute to **faxRelayOnly** or **faxRelayG711G726** (you must also add the **faxRelay** feature to the *Software* component's *featureList* attribute). The value **faxRelayOnly** means that modem calls are not supported. For more information, see "Fax relay" (page 79).
Note: If the negotiated value of the *modemFaxEncoding* attribute is useVoiceEncoding, all audio traffic is treated as voice traffic. In this case, the negotiated value of the *voiceEncoding* attribute must be g711, g711g726 or g726 for modem/fax and fax traffic to be encoded. If the negotiated value of the *voiceEncoding* attribute is g728 or g729, modem and facsimile calls are not supported.		

Fax relay

MVP-E FPs support fax relay. Fax relay conserves bandwidth by allowing Voice Transport to demodulate modulated fax traffic prior to transporting it clear channel through the Passport subnet. At the destination Passport node, fax traffic is remodulated and sent to the remote facsimile terminal. Fax relay supports the following demodulation/modulation rates:

- 2.4 and 4.8 kbit/s, according to ITU-T V.27
- 7.2 and 9.6 kbit/s, according to ITU-T V.29
- 12.0 and 14.4 kbit/s, according to ITU-T V.17 (MVP-E only)

On the MVP-E FPs, fax relay does not support ITU-T V.17. However, V.17 fax machines can use fax relay encoding rates by sending fax traffic using V.29. Alternately, you can allow V.17 fax machines to use V.17 rates by provisioning Voice Transport to encode V.17 fax traffic using G.711 or G.726 encoding. For V.17 fax traffic to be encoded using G.711 or G.726,

- you must provision the *modemFaxEncoding* attribute with the value faxRelayG711G726
- you must provision the *v17EncodedAsG711G726* attribute with the value yes
- the negotiated value for modem/fax traffic must be g711, g711G726, or g726, v27v29Relay for fax traffic, and yes for the *v17EncodedAsG711G726* attribute

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

The actual rate used for a given call depends on line quality and the rates supported by the sending fax machine. During the negotiation or handshake phase of a facsimile call (governed by the T.30 protocol, which uses signals in accordance with ITU-T V.21), fax relay disables all of the called fax machine’s proprietary features (those features supported only between fax machines produced by the same manufacturer). This enables fax machines from different manufacturers to connect when using fax relay.

MVP-E support for SG3

MVP-E FPs provide support to enable the completion of SG3 fax calls at the supported rates provided by V.17, V.29 and V.27ter (for example 14.4 kbps and lower). This functionality does not support the V.34 standard but rather simply allows SG3 calls to downspeed to support G3 rates.The downspeeding of SG3 fax calls to G3 rates is only supported in the faxRelay mode. The completion of SG3 fax calls at G3 rates is also supported in a 4400-6400 interworking scenario.

Congestion management

MVP-E FPs support congestion management techniques for voice and modem/fax traffic. See the following sections for more information:

- "About congestion" (page 81)
- "Reaction to congestion" (page 81)
- "Reaction to the absence of congestion" (page 83)

The congestion management techniques described in the following sections also include the application or removal of silence suppression. For more information on silence suppression, see "Speech activity detection (SAD)" (page 70).

About congestion

The two types of congestion indication are explicit and implicit indication. Explicit indication is set by the forward congestion indication (FCI) bit. The FCI bit tags cells that encounter Passport trunk congestion. Implicit indication is signified by cell loss. When a Passport trunk encounters a high level of congestion, it can start to discard cells. The discarding of cells is detected by the destination Passport node through the use of a sequence number.

The following conditions indicate the onset of congestion:

- two instances of cell loss within one second
- one instance of cell loss and an FCI bit within one second
- five consecutive FCI bits

Congestion clears or is absent when no FCI bits are encountered and no cell loss occurs for a certain amount of time.

Reaction to congestion

Table "Sequence of events when congestion occurs" (page 83) describes the reactions of voice and MVP-E FPs to the onset of congestion. If enabled by means of provisioning, both voice and MVP-E FPs can react to the onset of congestion by first enabling SAD. Voice Transport enables SAD during periods of congestion only if the negotiated value of the *silenceSuppression* attribute is congested or slowAndCongested.

If congestion persists and if enabled by means of provisioning, MVP-E FPs can continue to react to congestion by down-speeding the encoding rate. Channel down-speeding can occur for voice, modem/fax, and fax traffic and on a channel-by-channel basis. For voice traffic, down-speeding involves a reduction in the encoding rate when the negotiated value of the *voiceEncoding* attribute is g711G726 or g726. For modem/fax and fax traffic, down-speeding involves a reduction in the encoding rate when the negotiated value of the *modemFaxEncoding* attribute is g711G726 or g726. The minimum rate a channel can use depends on the negotiated minimum rate value for the *minVoiceG711G726Rate* attribute for voice traffic and the *minModemFaxG711G726Rate* attribute for modem/fax traffic. For interworking calls between voice and MVP-E FPs, the minimum rate value of the *minVoiceG711G726Rate* and *minModemFaxG711G726Rate* attributes can be modified during end-to-end negotiation.

After down-speeding, a channel ignores all indications of congestion for up to 415 milliseconds, depending on the FP and the current encoding rate.

Table 10
Sequence of events when congestion occurs

Order	Reaction to congestion	Traffic type	FP type
1	SAD enabled	voice	MVP-E
2	Down-speed from 64 to 32 kbit/s If the negotiated encoding value is faxRelayG711G726 for the *modemFaxEncoding* attribute, down-speeding only occurs on modem/fax traffic between MVP-E FPs (that is, until the MVP-E FPs detect a fax preamble).	voice, modem/fax, and fax	MVP-E
3	Down-speed from 32 to 24 kbit/s	voice only	MVP-E

Reaction to the absence of congestion

Table “Sequence of events when congestion clears or is absent” (page 84) describes the reactions of voice and MVP-E FPs to the absence of congestion or when congestion clears. If enabled by means of provisioning, both voice and MVP-E FPs can react to the absence of congestion by first up-speeding the encoding rate. Channel up-speeding can occur for voice, modem/fax, and fax traffic and on a channel-by-channel basis. For voice traffic, up-speeding involves an increase in the encoding rate when the negotiated value of the *voiceEncoding* attribute is g711G726 or g726. For modem/fax and fax traffic, up-speeding involves an increase in the encoding rate when the negotiated value of the *modemFaxEncoding* attribute is g711G726 or g726.

Channel up-speeding delay is a function of its current encoding rate, as shown in table “Sequence of events when congestion clears or is absent” (page 84). As well, the maximum rate a channel can use depends on the negotiated maximum rate value of the *maxVoiceG711G726Rate* attribute for voice traffic and the *maxModemFaxG711G726Rate* attribute for modem/fax and fax traffic. The higher the encoding rate (for example 64 kbit/s is higher than 24 kbit/s), the longer it takes to up-speed. For interworking calls between voice and MVP-E FPs, the maximum rate value of the *maxVoiceG711G726Rate* and *maxModemFaxG711G726Rate* attributes can be modified during end-to-end negotiation. The maximum number of channels an FP can up-speed is one channel every two seconds.

If enabled by means of provisioning, MVP-E FPs can continue to react to the absence of congestion by disabling SAD.

Table 11
Sequence of events when congestion clears or is absent

Order	Encoding rate change	Traffic type	FP type	Up-speed delay
1	24 to 32 kbit/s	voice only	MVP-E	3 – 4 minutes of egress speech
2	32 to 64 kbit/s	voice, modem/fax, and fax	MVP-E	7 – 8 minutes of egress speech
3	Maximum negotiated rate to SAD off	voice, modem/fax, and fax	MVP-E	13 – 14 minutes of egress speech

Supported signaling protocols and related information

Voice Transport supports both channel associated signaling (CAS) and common channel signaling (CCS) protocols. CAS information is transported with the associated voice channel. For CCS, one channel or link is dedicated to transporting signaling information for a number of channels. See the following sections for more information on Voice Transport signaling protocols and related capabilities:

- “Channel associated signaling (CAS)” (page 85)
- “Common channel signaling (CCS)” (page 86)
- “Call discriminator” (page 87)
- “Idle channel activity” (page 89)
- “Channel busy out” (page 89)

See appendix “Signalling refresher” (page 177) for more information on CAS and CCS formats.

Channel associated signaling (CAS)

Voice Transport supports end-to-end line and dual tone multi-frequency (DTMF) register signaling across a network on a channel-by-channel basis. It does this by using the following generic attributes of the various CAS signaling protocols:

- signaling on A, AB, or ABCD bits
- signaling bit inversion
- ABCD code identifying idle and seize conditions
- two provisionable modes—interpret or transparent—for transporting signaling information

CAS behaves in two ways depending on the provisioned signaling mode (*VoiceService Framer* component’s *casSignalling* attribute).

If *casSignalling* is set to transparent, the signaling bits are transported transparently through the network. You set *casSignalling* to transparent when both ends of a connection are using the same signaling tables. In other words, the *VoiceService Framer* component's *signalBits* attribute must be set to the same value at both ends of a voice service connection: both A, both AB or both ABCD.

If *casSignalling* is set to interpret, then the only recognized signaling changes are the ones where the signaling bits match the provisioned idle and seize codes. In other words, if the attributes *signalBits* = AB, *idleCode* = 1010 (ABCD), and *seizeCode* = 1110 (ABCD), then the AB bit pattern 10 will be recognized as an idle, and an AB bit pattern 11 will be recognized as a seize. All other signaling bit changes (for example AB = 01 and AB = 00) will be filtered out at the local-end and the state remains unchanged. The main reason for using signaling interpretation is when both ends of a connection are using different signaling formats (for example, E1 to DS1).

> ***Note:*** A bit signalling is only available on the TTC2M cards.

Voice Transport can translate 2-state signaling protocols such as E&M TIE trunk signaling between voice FPs or between MVP-E FPs.

When a DS1 MVP-E FP is connected to an E1 MVP-E FP, then voice or data calls cannot be distinguished using CAS. If the user has a requirement for data and voice call separation on either side, then the user must designate specific Passport trunks as data and others as voice.

Common channel signaling (CCS)

CCS is transported as bit transparent data. This requires you to configure a bit transparent data service (BTDS) on the appropriate signaling channel of one of the following FPs: 1-port DS1 MVP-E, 4-port DS1 MVP-E, 1-port E1 MVP-E or 4-port E1 MVP-E. The TTC2M MVP-E FP does not support CCS protocols.

See 241-7401-775 *Passport 7400 Bit Transparent Data Service Guide* for information on how to provision a BTDS to carry CCS information. For CCS with the call discriminator capability provisioned, refer to "Call discriminator" (page 87) in this chapter.

> ***Note:*** Data call discrimination is only supported on legacy voice FPs. It is not supported on MVP-E FPs.

Call discriminator

Only voice FPs support the call discriminator feature. The call discriminator feature interprets CCS messages sent by the PABX or similar customer supplied equipment and dynamically configures the operational mode of an associated voice service to be either idle, voice or data (for more details, see "Example call discriminator configuration" (page 87)). In idle mode, the voice FP does not transmit packets into the Passport network (for information on MVP-E FP activity when a channel is idle, see "Idle channel activity" (page 89)). In voice mode, all provisioned voice capabilities are active. In data mode, a 64 kbit/s data channel is made available for data transmission (essentially as for Btds). The voice services and the BTDS that monitors the CCS messaging must be on the same voice FP. This feature supports CCS messaging using British Telecom's Digital Network signaling No. 1 protocol (DPNSS1) and Nortel Network's Meridian 1 ISDN Primary Rate Interface protocol (MCDN).

You provision the call discriminator capability by adding either the *Dpnss1* or *Mcdn* bit transparent data service (BTDS) subcomponent. You provision the *Dpnss1* subcomponent if a voice FP is to interpret DPNSS 1 protocol messages. You provision the *Mcdn* subcomponent if a voice FP is to interpret MCDN protocol messages. For more information on provisioning the call discriminator, refer to the provisioning procedures in 241-7401-775 *Passport 7400 Bit Transparent Data Service Guide*.

Example call discriminator configuration

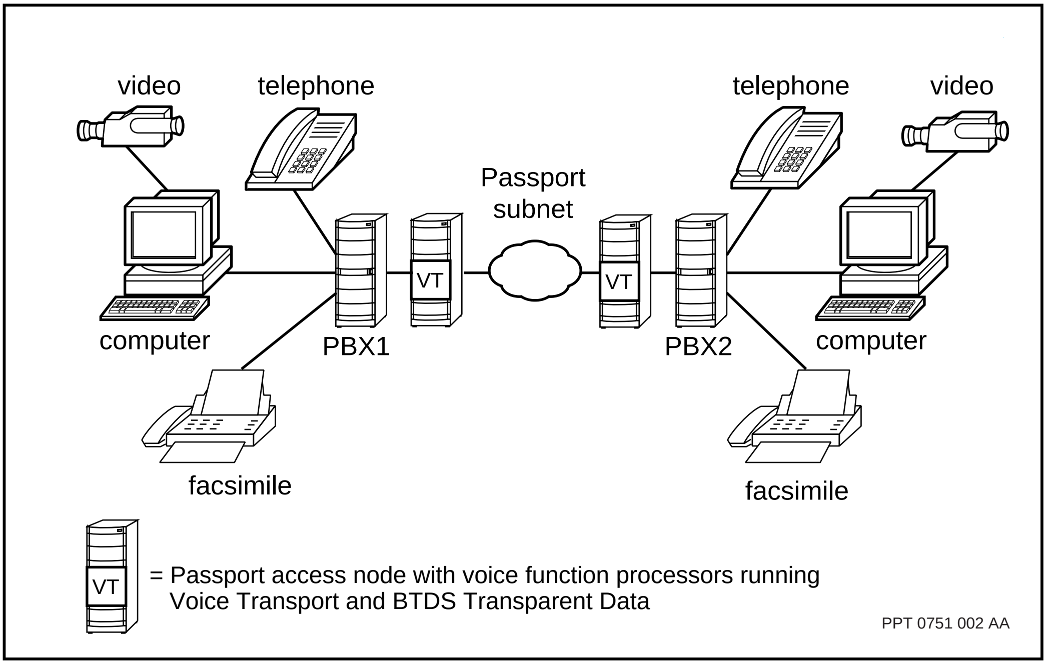
Figure "Passport network utilizing the call discriminator capability" (page 88) depicts a typical scenario where the voice/data call capability is utilized. In this figure, two PBXs are connected transparently through a Passport network. PBX 1 receives voice and data calls from various sources. These calls are forwarded to the Passport subnet. The Passport node is able to distinguish the type of call and adjust the operational mode of the voice service as follows. If the call being set up is required for data communication, then the voice service is automatically adjusted as follows:

- minimum and maximum voice compression rates are set to 64 kbit/s
- silence suppression is disabled
- echo cancellation is disabled
- A-mu law conversion is disabled

In other words, a clear 64 kbit/s traffic channel is available for the data call. The adjustments made to the voice service being used for the data call only last for the duration of the call. Once the call terminates, the voice service reverts to the previously provisioned values.

If the call being set up is required for voice communication, all of the previously provisioned features for the voice service are enforced. The call is handled as a voice call. Once a call terminates (whether it be a data or voice call), the timeslot becomes idle and the voice service (channel) stops transmitting data into the Passport network. This frees up bandwidth on Passport trunks.

Figure 16
Passport network utilizing the call discriminator capability



Idle channel activity

MVP-E FPs can conserve bandwidth between Voice Transport calls when channels become idle. When a call terminates, the corresponding channel becomes idle. Connected PBXs transmit an idle pattern—a particular sequence of bits corresponding to the particular interface (E1, DS1, TTC)—to indicate when a channel is idle. MVP-E FPs do not send cells into the subnet upon detection of the idle pattern, regardless of the value provisioned under the *silenceSuppression* attribute (to conserve bandwidth between CAS calls on voice FPs, the *silenceSuppression* attribute must be set to casIdleCode). The idle pattern can correspond to the value provisioned under the *idleCode* attribute for channel associated signaling (CAS) calls or the *endOfCallPattern* attribute for common channel signaling (CCS) calls.

Channel busy out

When a channel cannot be set up for an incoming call, Voice Transport notifies the end user. By provisioning the *transmitBusyYellow* and *transmitCasYellow* attributes, you enable the notification mechanisms.

For CCS links provisioned with the *transmitBusyYellow* attribute, Voice Transport transmits a yellow alarm to the PBX. For CAS links, the associated timeslot is idled and then seized. Voice Transport transmits a busy out signal to the PBX which ensures that calls cannot be placed on the service.

CAS links can also be provisioned with the *transmitCasYellow* attribute. You can set this attribute to yes for all or a specific number of *VoiceService* components associated with a particular port. When all *VoiceService* components that have *transmitCasYellow* set to yes experience subnet connectivity problems, Voice Transport generates a yellow alarm. CAS links provisioned with both attributes will still generate “busy out” signals for individual channels experiencing connection problems. A yellow alarm generated by the *transmitCasYellow* attribute resets the channel busy out capability so that it can be activated again once the yellow alarm condition no longer exists.

> ***Note:*** It is important that the user understand how the PABX or similar customer supplied equipment responds to receiving a yellow alarm prior to enabling this feature.

System parameters of Voice Transport

The following sections provide details on the system parameters of Voice Transport:

- “Dependencies” (page 90)
- “Voice Transport feature-to-card relationships” (page 90)

Dependencies

Passport software releases prior to R5.1 do not support end-to-end negotiation for Voice Transport connections. An FP running R5.1 and later software can successfully establish a connection with another FP running a previous software release. However, no end-to-end negotiation occurs between the FPs. Without end-to-end negotiation, provisioning errors are more difficult to troubleshoot as the *serviceFailureReason* attribute contains the value undetermined.

Passport R5.1 and later software also supports interworking between voice and MVP-E FPs. The interworking supported by Voice Transport is based on a specific set of features and functions (see “Interworking provisioning considerations” (page 46) for more information).

> ***Note:*** The 4-port MVP-E FPs do not interwork with any FP using a software release prior to release PCR 2.3.

Voice Transport feature-to-card relationships

Table “Voice Transport capability-to-card relationships” (page 91) links Voice Transport capabilities to the function processors which support them.

Table 12
Voice Transport capability-to-card relationships

	Card type				
Voice Transport capability	**DS1 MVP-E**	**E1 MVP-E**	**DS1 MVP-E 4-port**	**E1 MVP-E 4-port**	**TTC2M MVP-E**
G.711 (PCM) 64 kbit/s clear channel transport	X	X	X	X	X
G.726 (ADPCM) 16, 24, 32 kbit/s voice and modem/fax compression	X (see Note 1:)	X (see Note 1:)	X (see Note 2:)	X (see Note 2:)	X (see Note 1:)
G.728 (LD-CELP) 16 kbit/s voice compression (see Note 3:)	X	X	X	X	X
G.729 or G.729A (CSA-CELP) 8 kbit/s voice compression (see Note 4:)	X	X	X	X	X
Fax relay	X (see Note 5:)	X (see Note 5:)	X (see Note 6:)	X (see Note 6:)	X (see Note 5:)
Speech activity detection (SAD), with comfort noise cap	X (see Note 10:)	X (see Note 10:)	X (see Note 9:)	X (see Note 9:)	X (see Note 10:)
Fax idle suppression (FIS)	X (see Note 10:)	X (see Note 10:)	X (see Note 10:)	X (see Note 10:)	X (see Note 10:)
Echo cancellation (see Note 7:)	X	X	X	X	X
A-law (international) to mu-law (North American) conversion (see Note 8:)	X		X		
(Sheet 1 of 3)					

Table 12 (continued)
Voice Transport capability-to-card relationships

	Card type				
Voice Transport capability	**DS1 MVP-E**	**E1 MVP-E**	**DS1 MVP-E 4-port**	**E1 MVP-E 4-port**	**TTC2M MVP-E**
Dynamic up/down speeding (G.711/ G.726)	X (see Note 1:)	X (see Note 1:)	X (see Note 11:)	X (see Note 11:)	X (see Note 1:)
Gain adjustment	X	X	X	X	X
Channel associated signaling (CAS) support	X	X	X	X	X
Common channel signaling (CCS) support (see Note 9:)	X	X	X	X	
Voice/data call discriminator					
Fax/speech discriminator (see Note 10:)	X	X	X	X	X
Provisionable hangover time for FIS and SAD (see Note 10:)	X	X	X	X	X
Configurable egress buffer (see Note 10:)	X	X	X	X	X
(Sheet 2 of 3)					

Table 12 (continued)
Voice Transport capability-to-card relationships

	Card type				
Voice Transport capability	**DS1 MVP-E**	**E1 MVP-E**	**DS1 MVP-E 4-port**	**E1 MVP-E 4-port**	**TTC2M MVP-E**
Tandem pass through (see Note 12:)	X (see Note 10:)	X (see Note 10:)	X (see Note 11:)	X (see Note 11:)	X (see Note 10:)

Note 1: Requires Passport R5.1 or later software.

Note 2: MVP-E FPs only support G.726 voice encoding at 24 and 32 kbit/s and G.726 modem/ fax encoding at 32 kbit/s.

Note 3: You must add g728 to the *Sw Lpt* component's *featureList* attribute; this voice encoding type is not included with the vtds feature.

Note 4: You must add g729 to the *Sw Lpt* component's *featureList* attribute; this voice encoding type is not included with the vtds feature.

Note 5: You must add faxRelay to the *Sw Lpt* component's *featureList* attribute; this facsimile encoding type is not included with the vtds feature.

Note 6: Since 4-port MVP-E FPs support fax relay V.17 fax calls, you must provision the *v17EncodedAsG711G726* attribute with the value *no*

Note 7: Voice FPs provide echo cancellation according to ITU-T G.165. MVP-E FPs provide echo cancellation according to ITU-T G.164, G.165, and G.168. MVP-E echo cancellation requires Passport R5.1 or later software.

Note 8: Only DS1 FPs (MVP-E) handle compander law conversion for connections with E1 FPs (MVP-E).

Note 9: Requires that you configure a bit transparent data service (BTDS).

Note 10: Requires Passport R5.1 or later software.

Note 11: Requires PCR 2.3 software.

Note 12: You must add tandemPassThrough to the *Sw Lpt* component's *featureList* attribute; this feature is not included with the vtds feature.

(Sheet 3 of 3)

Chapter 7
Voice Transport, PORS, and route selection

See the following sections for information on PORS and route selection:

- “Voice Transport and PORS” (page 95)
- “Route selection” (page 106)

For fundamentals on network clock synchronization, see 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide*.

Voice Transport and PORS

Voice Transport relies on the Path Oriented Routing System (PORS) to set up a permanent connection between the two endpoints of a network path for every voice service. Once the path is established, the Voice Transport user may regard it as an end-point to end-point wire. See the following sections for more information:

- “Establishing a path” (page 96)
- “Using default values” (page 98)
- “Creating the path” (page 99)
- “Path bumping” (page 99)
- “Optimizing paths” (page 99)
- “Path establishment failures” (page 100)
- “Established path failures” (page 100)
- “Passport trunk bandwidth allocation” (page 101)
- “Tips for setting up Voice Transport on your system” (page 105)

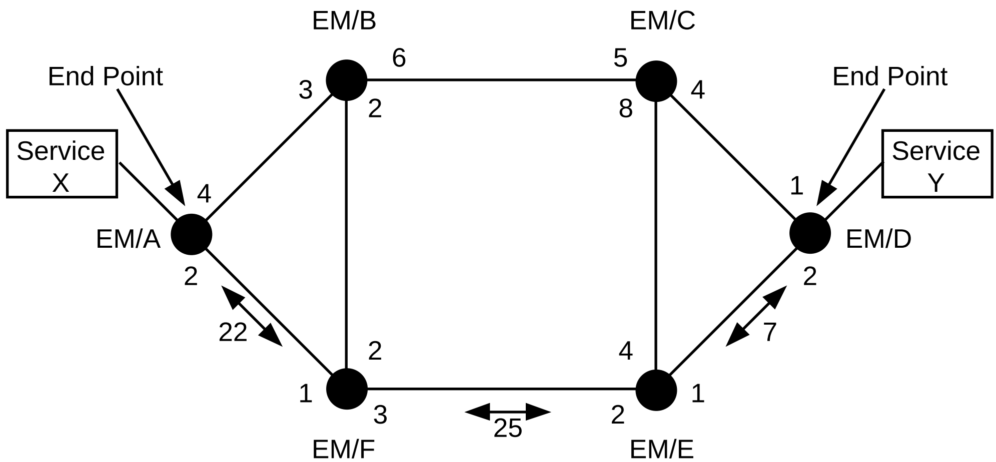
Establishing a path

Figure “Path and path description” (page 97) shows a path across a six-node network and illustrates some of the terms used in this section.

When a voice service is provisioned on a node, you provide an instance value under the voice service *PermanentLogicalConnection* (*Plc*) component for the other end of the connection by provisioning the *remoteName* attribute. Path establishment is automatic. For more information about PORS components see 241-5701-060 *Passport 7400, 15000, 20000 Components* or 241-5701-435 *Passport 7400, 15000, 20000 Path-Oriented Routing System Guide*.

> ***Note 1:*** Some provisioning data must point to the exact identifier of the other end of the connection (*Plc* component *remoteName* attribute). If this value is not correct, a path is not established. See “remoteName attribute” (page 66), for more information.
>
> ***Note 2:*** Should the network be divided into clusters and/or topology regions, it should be noted that the *path* attribute can only display information about the current cluster or region segment. For example, should a service traverse a cluster or inter-region link, the *path* attribute shall indicate termination at the cluster or region gateway respectively, and not at the service end point.

Figure 17
Path and path description



Legend:

2 = Trunk instance

25 = LCh- Logical Channel Number

= Node

Path = EM/A Service/X, Em/A Trunk/2 LCH/22. EM/F Trunk/3 LCh/25,
Description EME/ Trunk/1 LCh/7. EM/D Service/Y

NOTE: By convention, outbound trunk names are used to describe paths and routes.
EM refers to a Passport-node enterprise module.

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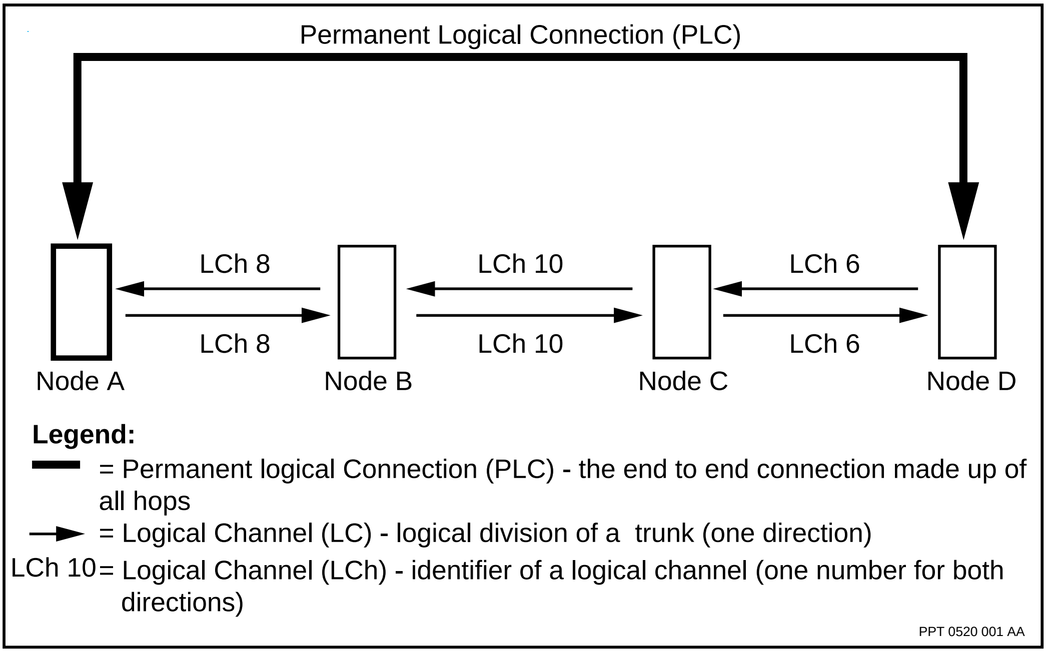
Using default values

Voice Transport comes with a set of default values for most of the attributes associated with the service; you do not need to provision them. The default values are designed to set up a Permanent Logical Connection using the optimal route across the network. It is a good idea to use the default values for the initial setup, adding options only as needed.

In many networks the default values are sufficient; however, you can choose how PORS selects a route. Attributes that allow selection are explained in the section entitled "Route selection" (page 106). Figure "PLCs, LCs, and LChs" (page 98) illustrates the numbering convention for Logical Channels.

Figure 18
PLCs, LCs, and LChs



Creating the path

The path is established on a hop-by-hop basis. A set-up packet is sent down the route chosen by the *RouteSelector* (*Rs*) component. As the packet follows the route it uses the Passport trunks that will be necessary to complete the path. At each point along the route the following actions are triggered:

- creation of the *LogicalChannel* (*LCh*) components on a Passport trunk
- allocation of the logical channels to be used
- verification of bandwidth availability
- reservation of bandwidth

When the path-setup packet reaches the destination end point, a path setup confirmation packet is returned to the source. This enables the path for data transfer.

Path bumping

Path bumping is the forced rerouting of an existing path by a new higher priority path of another logical connection. Bumping happens when there is not enough bandwidth in the network to establish a new path. The rerouting can in turn cause bumping of other paths. It may happen that a bumped path cannot be reinstated if the network is heavily loaded.

Optimizing paths

Over time, a PORS connection may end up on a less than optimal path due to link failures, node software upgrades, Passport trunks being locked, or other possible scenarios. Path optimization periodically attempts to move the PORS connection back to a more optimal path. The first step of the optimization process begins when the routing system determines the best available path and compares it with the path currently used by the connection. If this new path provides better metrics, the connection is moved to the new path and the original path is released.

If the new path does not provide better metrics, path optimization will then attempt to balance the PORS load on the link groups used to carry the path. This process involves moving the connection (which is being optimized) to a new path established on a different link in the link group. This will only occur if it contributes to re-balancing the load on the link groups.

The optimization process is administered by PORS Connection Control which resides on each Passport node in the network. For more information on path optimization, refer to 241-5701-435 *Passport 7400, 15000, 20000 Path-Oriented Routing System Guide*.

> ***Note:*** Path optimization is an optional feature. To activate it on a node, this feature must be provisioned.

Path establishment failures

The selected path can fail to establish under the following conditions:

- there is not enough bandwidth available, according to the *Trunk PathAdministrator* (*Pa*) component
- there is a failure (node, FP, or Passport trunk) along the chosen route
- the Passport trunk has reached the maximum number of paths (logical channels) that it can support (*maxLc* attribute)

Established path failures

An established path can fail under the following conditions:

- there is a failure (node, FP, access line, or Passport trunk) along the chosen route
- PORS path bumping occurs

In the case of a failure, a path-setup failure packet is returned from the point of failure back to the source end point. The end point reports the failure reason to the *Rs* component and requests a new route. If another route is not available even with bumping, the *Rs* component informs the end point that the path cannot be set up. If another route is selected by the *Rs* component, the end point starts the path-setup procedure again.

Passport trunk bandwidth allocation

The following sections list provisionable parameters, provided by a *Plc* or *Trunk Pa* component, that allow different policies of bandwidth allocation to be enforced.

- “Using the Interrupt queue” (page 101)
- “Trunk Path Administrator” (page 101)
- “Permanent Logical Connection (PLC)” (page 102)

It is up to the network engineers to decide what constitutes an efficient sharing of resources.

Using the Interrupt queue

Highest emission-priority Voice Transport cells can be sent through the interrupt queue ahead of less urgent cells. The *framingType* attribute of the *Framer* component under *Trunk Unack* can be set to allow the use of the interrupt queue. When the *framingType* attribute is set to interrupting, the interrupt queue is activated.

> ***Note:*** *Trunk* components should be set to the same value on both ends of the connection. Failure to do so can result in a failure to obtain a path.

Voice Transport transmissions can use the Passport trunk without the interrupt queue but the quality of service may be reduced.

Trunk Path Administrator

The following are the provisionable parameters provided by a *Trunk Pa* component which can be used for bandwidth allocation:

- Reserving bandwidth—Bandwidth on a Passport trunk is shared between connectionless and connection-oriented traffic. Bandwidth that is unused by one traffic type can be used by the other. PORS reserves bandwidth in both directions on each Passport trunk in the path. This reservation is not enforced by PORS but is used to determine the number and size of the paths that can be set up on a given Passport trunk. Bandwidth is expressed in bit/s in each direction. Path instantiation on a Passport trunk is delimited by *Trunk Pa* component provisionable attributes *maxLc* and *maxReservedBwOut*. Use the *requiredRxBandwidth* and *requiredTxBandwidth* attributes under the *Plc* component to reserve bandwidth for a path.

Note: Bandwidth reservation is based on the provisioned encoding for the particular type of connection—voice, modem or facsimile. Review the default values offered by Voice Transport—particularly for connections using g728 or g729 encoding—and modify them if they do not meet the needs of your particular traffic and encoding type.

- *maxLc* attribute—This is the limit on the number of individual Logical Channels (or paths) that traverse this Passport trunk. When this number is reached, no new paths can be established over this Passport trunk until some existing paths clear.
- *maxReservedBwOut* attribute—This is the percentage of total Passport trunk bandwidth which PORS can allocate among individual Logical Channels. Once this percentage is reached, the Passport trunk has no more reservable bandwidth. No paths can establish over this Passport trunk until some existing paths clear.

  For example, on a DS-1 trunk using all timeslots at 1.536 Mbit/s, a value of 65% for this Passport trunk attribute makes this Passport trunk capacity appear to be 0.9984 Mbit/s for path-oriented routing. Connectionless traffic can use the remaining 0.5376 Mbit/s. Hence, PORS never reserves more than 0.9984 Mbit/s of this Passport trunk.

Permanent Logical Connection (PLC)

The following are the provisionable parameters provided by a PLC which can be used for bandwidth allocation:

- Specifying setup and holding priorities (path bumping)—All PLCs in PORS have setup and holding priorities assigned to them. If a route with sufficient unreserved bandwidth cannot be found for a PLC, existing paths may be moved elsewhere to free up bandwidth. This process is called path bumping. Existing path-holding priorities and new path-setup priorities are compared to determine when a new path may bump an existing path. An attempt is made to reroute a path which has been bumped.

Setup and holding priorities accommodate scenarios where customers would like to determine which paths are allocated bandwidth at setup time (setup), but once set up, the paths have to remain (holding) to minimize disruption. For example, if a network is carrying video through BTDS, voice through the voice service, and data through HTDS, and the user considers video to be the highest priority, data to be next, and voice to be the lowest, one way of accommodating such a requirement is shown in table “Example setup and holding priorities” (page 103).

Table 13
Example setup and holding priorities

Traffic Type	Setup Priority	Holding Priority
Data	Medium	Medium
Voice	Low	High
Video	High	High

Note: The values listed in table “Example setup and holding priorities” (page 103) provide an example only and are not the default settings.

The *setupPriority* and *holdingPriority* attributes of the *Plc* component specify these priorities. A high holding-priority path will not be moved by a lower setup-priority path. Conversely, a high setup-priority path may bump lower holding priority paths.

Each priority may have one of five values, ranging from zero (0) to four (4), where 0 is the most important path and has the highest priority, and 4 is the least important and has the lowest priority. A new path can bump an existing path only if the new path’s *setupPriority* attribute is numerically lower than the existing path’s *holdingPriority* attribute; that is, the new path has the higher priority.

Setup and holding priority have a default setting of medium (2). PLCs of more, or of less, importance than the default can be reassigned other values.

For more details on path bumping, see document 241-5701-435 *Passport 7400, 15000, 20000 Path-Oriented Routing System Guide*.

- Specifying emission and discard priorities—Emission priority is a measure of how urgently a cell will be emitted to the Passport trunk. The higher the emission priority, the faster the cell is sent to the Passport trunk. Emission and discard priorities are set independently. Care must be taken when setting emission and discard priority values for specific traffic types. Setting the discard priority of data traffic to a high value (for example a value of 1) could result in adverse effects on voice and video traffic. Refer to "Tips for setting up Voice Transport on your system" (page 105) for more information on the effects of emission and discard priority values on traffic types.

  Note: Emission and discard priorities can have implications for congestion management in your network. Do not adjust these values until you have considered all of the implications for network traffic.

  The *emissionPriority* and *discardPriority* attributes of the *Plc* component affect all cells on a particular path. Discard reflects the importance that a cell reach its destination while emission reflects the urgency that a cell reach its destination as quickly as possible.

  These attributes are relative to other traffic values for other transmissions. For example, setting all traffic using a particular Passport trunk to the highest emission priority would not accomplish anything since all traffic must wait the same average time before emission to the Passport trunk.

- Specifying that a path terminate and not reroute—Some applications using Voice Transport may not tolerate the delays caused by rerouting. To cause a path to terminate instead of rerouting, set the *pathFailureAction* attribute of the *Plc* component to disconnectConnection. The default setting is reRoutePath.

Tips for setting up Voice Transport on your system

The following sections are meant to be a general set of guidelines for using Voice Transport with other types of traffic:

- “Emission and discard priority” (page 105)
- “Other bandwidth considerations” (page 106)

> ***Note:*** The term highest discard priority means last to be discarded. Highest emission priority means first to be emitted.

Emission and discard priority

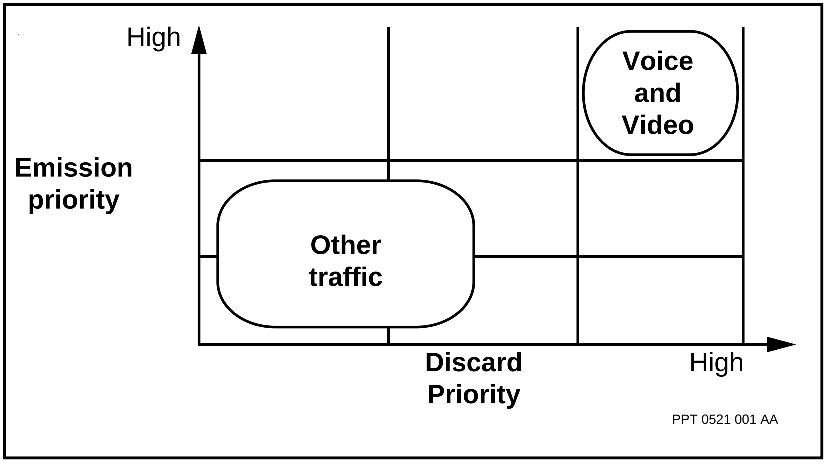
Voice Transport uses a strict priority system – higher emission-priority cells get the necessary amount of bandwidth faster. For this reason, too much high-priority traffic will restrict the flow of low-priority traffic. As a general rule, do not set the amount of frame-cell trunk interrupting mode traffic (highest priority) at greater than 80% (limit the value of the *maxReservedBwOut* attribute to 80% or less).

In general, Voice Transport traffic should be setup as shown in figure “Emission versus discard priority” (page 106). In this figure, voice and video traffic have a higher discard priority than data traffic. As a general rule to avoid losses in voice or video traffic, the following equation should be followed:

total voice traffic + total video traffic + other high priority data < total available bandwidth

If this general rule is not followed, loss of sensitive voice and video traffic may result during times of congestion. For networks which are running voice and video only, it recommended that the discard priority of video traffic be set higher than that of voice traffic since video traffic is more sensitive to losses.

Figure 19
Emission versus discard priority



Other bandwidth considerations

When considering running Voice Transport over a pre-existing Passport trunk, determine the usual bandwidth used by the pre-existing connectionless and connection-oriented traffic. Account for burstiness and control traffic. Estimate the amount of bandwidth needed by the Voice Transport traffic. Be sure that the total combined bandwidth is available.

Route selection

This section discusses the following topics:

- “Selecting paths” (page 107)
- “Restricting traffic” (page 109)
- “Restricting paths” (page 110)

You can use the following criteria to tailor the path that PORS selects to meet your requirements. This can be done during the initial provisioning session or at any later time should you wish to fine-tune the use of your network resources.

> ***Note 1:*** Reprovisioning causes service interruption. If you reprovision a connection, you terminate and re-establish it. The reprovisioning process temporarily stops data flow.

Note 2: Avoid unnecessary restrictions when provisioning a path. The more restrictions you add, the greater your chance of causing conflicts that will not allow a connection. For example, your restrictions from the security option may require a path that conflicts with the path needed by the general parameters that you have used or that may not support the type of traffic that you want to use. In cases like these, PORS will not be able to set up a connection.

Selecting paths

This section discusses the following topics:

- “Minimization criteria: cost and delay” (page 107)
- “Specifying a maximum cost for a path” (page 109)
- “Specifying a maximum delay for a path” (page 109)

Minimization criteria: cost and delay

PORS can select a path based on either the lowest cost or lowest delay. Both cost and delay cannot be minimized. Use the *minimizationCriterion* attribute under the *Plc* component to specify cost or delay.

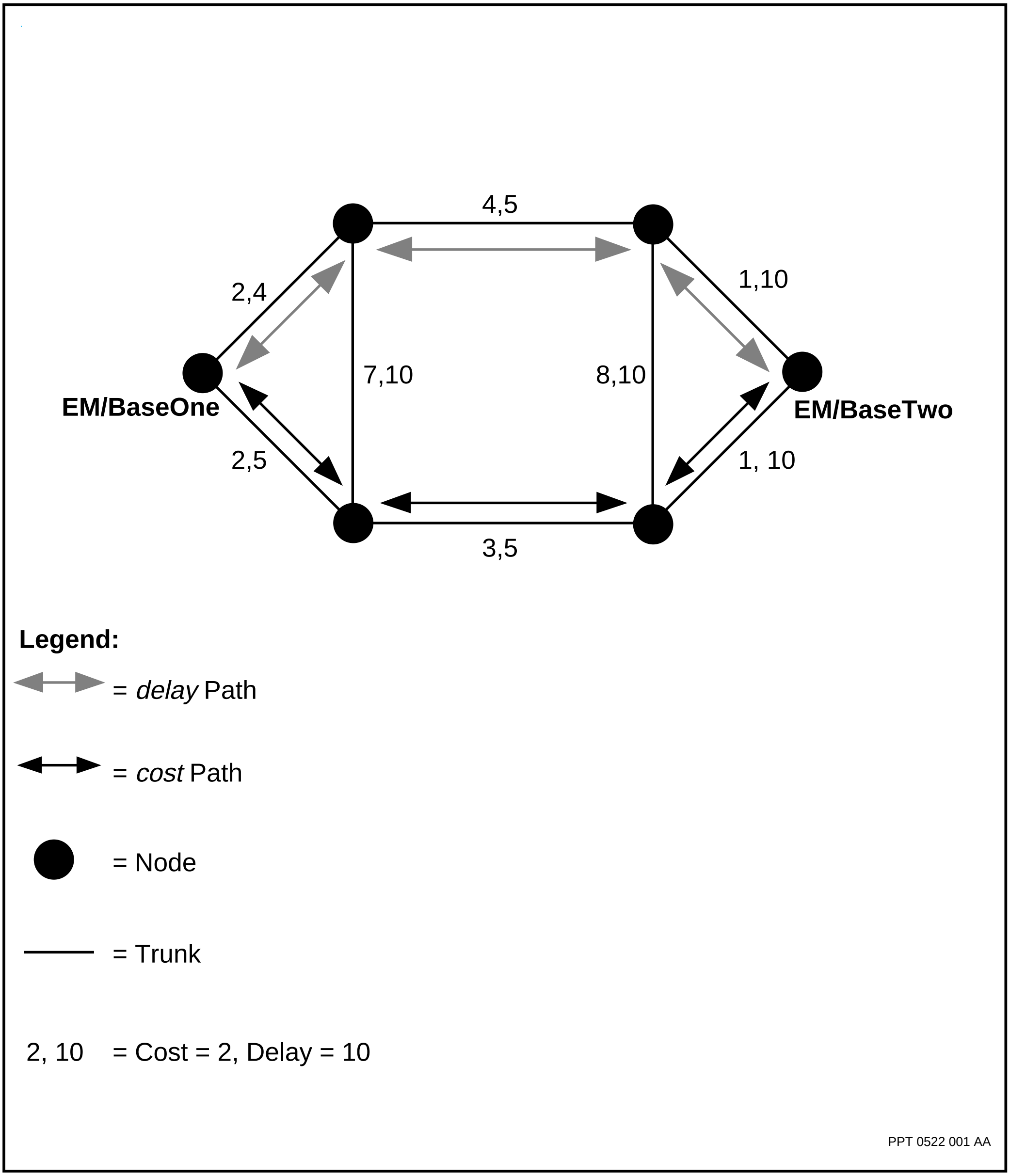
The routing system computes a minimum path from the values that you have assigned to the Passport trunks (cost) or from measured delay values that are associated with each Passport trunk (see figure “Path for cost or delay using trunkAttributeToMinimize” (page 108)).

To assign a cost to a Passport trunk, use the *trunkCost* attribute under the Trunk component.

Cost can be an actual dollar value or any parameter that you want to use. If default values are used, cost represents the hop count. Thus the number of hops across the network is minimized.

If you use a parameter for cost that reflects, in some manner, the actual cost of facilities, high-cost facilities will receive less use and reduce the cost of operating the network. This is the recommended method of using this option.

Figure 20
Path for cost or delay using trunkAttributeToMinimize



Specifying a maximum cost for a path

Providers of network services may wish to restrict some parameters for a particular circuit. This can only be done if all the Passport trunks have identical cost.

To specify the maximum total cost value of a path, you can use the *maximumAcceptableCost* attribute under the *Plc* component. Although this value is called cost, you may use it to reflect a variety of considerations, including geographical distance, hop count, or real dollar value.

The sum of the *trunkCost* attribute values of all Passport trunks used in the path will be less than or equal to the value specified by the *maximumAcceptableCost* attribute.

Specifying a maximum delay for a path

Passport trunk delay in PORS is measured for a 512-byte packet in one direction at the time of Passport trunk staging. Over time, this measured delay may change to reflect the updated operating delay but will not affect existing paths unless a Passport trunk restages.

To specify the maximum delay value of a path, use the *maximumAcceptableDelay* attribute under the *Plc* component. The sum of the delay values associated with all Passport trunks used in the path will be less than or equal to the value specified by *maximumAcceptableDelay*.

> ***Note:*** This parameter should be used when large delays are unacceptable for the service – for voice and other interactive data, for example.

Restricting traffic

This section discusses the following topics:

- “Restricting certain types of traffic to specific Passport trunks” (page 110)
- “Restricting traffic to certain types of Passport trunks” (page 110)

Restricting certain types of traffic to specific Passport trunks

PORS allows you to specify which types of traffic are carried on a given Passport trunk.

Use the *supportedTrafficTypes* attribute, under the Trunk component, to create an individual list of traffic types for each Passport trunk in your network (for example, data, voice, and video).

When you provision the connection, use the *requiredTrafficType* attribute to specify which traffic type is to be transported by the path. PORS will choose Passport trunks that include the *requiredTrafficType* list in their *supportedTrafficTypes* list. In other words, the *requiredTrafficType* must be included in the *supportedTrafficTypes* list or the Passport trunk will not be selected for the path.

For example, if the *requiredTrafficType* is data, only Passport trunks in the *supportedTrafficTypes* list that include data would be selected for the path.

Restricting traffic to certain types of Passport trunks

You may want to create an indicator of the type of Passport trunk that various traffic types use. Terrestrial or satellite links are examples of Passport trunking facilities. The *trunkType* attribute, under the *Trunk Pa* component, allows you to do this for up to eight different types of Passport trunks.

The *permittedTrunkTypes* attribute under the *Plc* component, allows a set of possible Passport trunk types to be specified for a route. Only Passport trunks with *trunkType* attributes that are found in the *permittedTrunkTypes* list are used to create the path.

Restricting paths

Restricting paths includes the following topics:

- “Security” (page 111)
- “Defining general parameters to restrict paths” (page 113)
- “Specifying a path manually” (page 114)

Security

PORS allows you to define varying security levels for the Passport trunks of the network. This option can, for example, prevent sensitive data from traveling over certain Passport trunks.

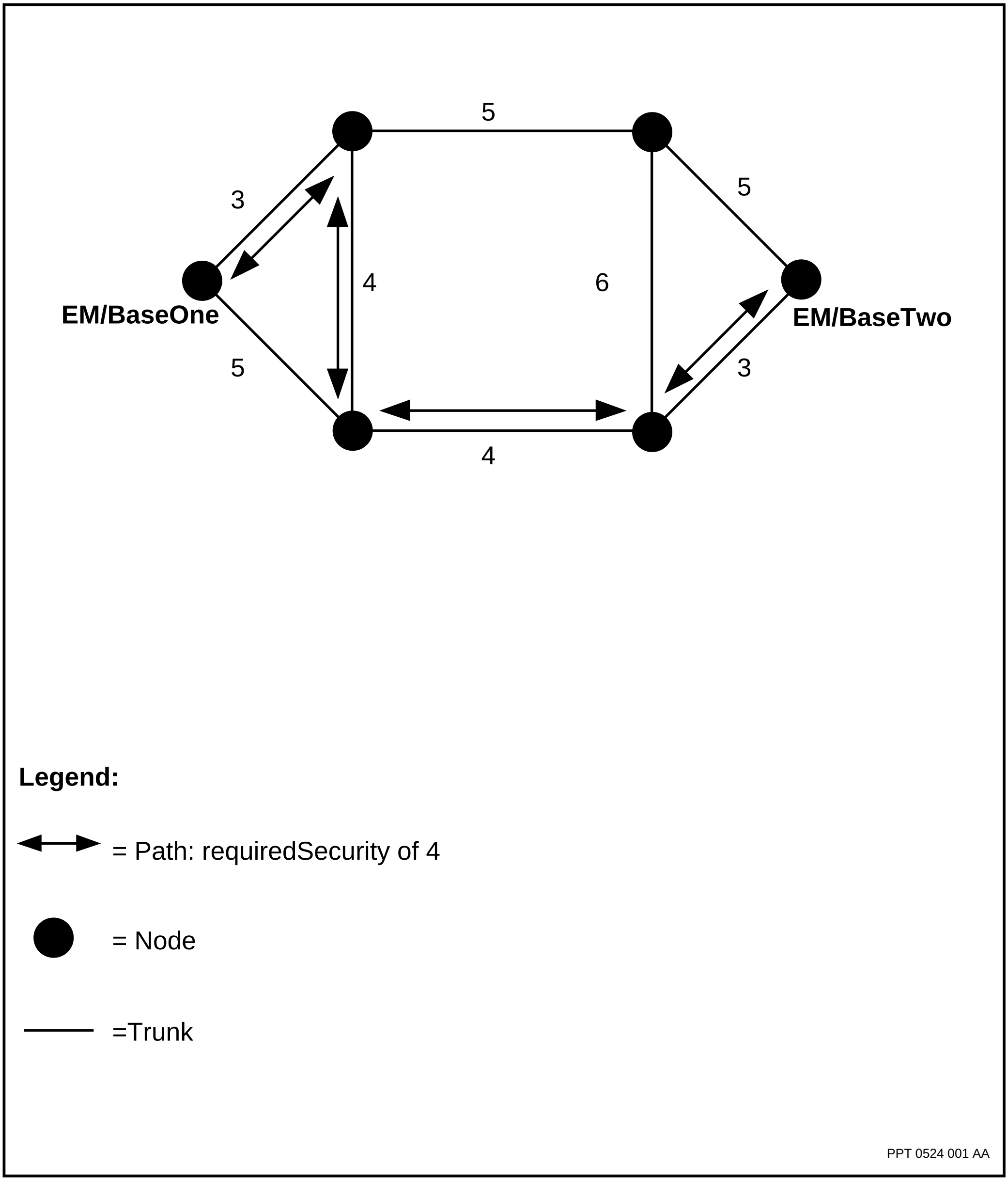
PORS has an option that allows you to specify the minimum security level of a path. To do this, provision a security value for the Passport trunks in your Passport 7400 network using the *trunkSecurity* attribute under the *Trunk Pa* component. When you provision the connection, enter a value for the *requiredSecurity* attribute under the *Plc* component.

The connection will only use Passport trunks that have been assigned security values of an equal or higher level than that of the connection. This is illustrated in the example in figure “Path determined using a requiredSecurity value of 4” (page 112). A lower number always represents a higher security level.

The default value for security is mid-range so that the network administrator can add security with minimal provisioning.

> ***Note:*** Over-use of this option can reduce its usefulness. This option can also reduce the number of recovery paths available to high security routes should an outage occur.

Figure 21
Path determined using a requiredSecurity value of 4



Defining general parameters to restrict paths

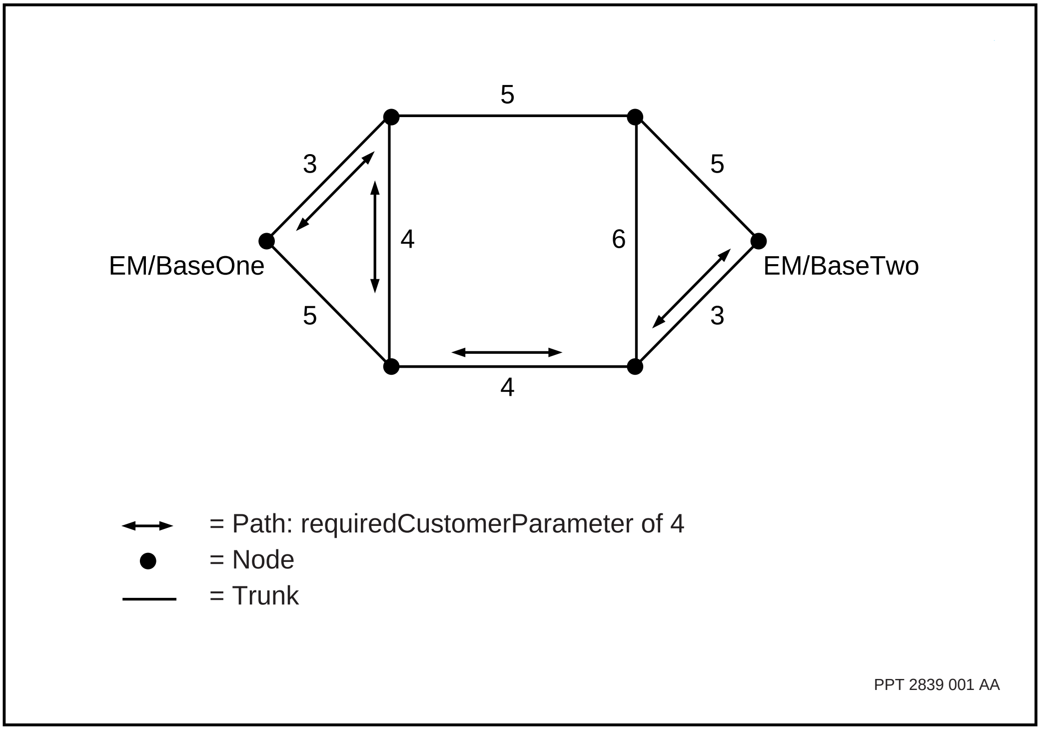
It may be convenient to be able to restrict certain classes of paths to certain Passport trunks. Most of the commonly used qualifiers are represented in security and traffic type. This is an additional option, to be used for any function that you deem appropriate.

PORS allows you to restrict certain paths to certain Passport trunks. This is done in a similar manner to the way that security is provisioned. Values are assigned to various Passport trunks in the network using the *customerParameter* attribute under the *Trunk Pa* component. When the *Plc* component is provisioned, it can be assigned a value using the *requiredCustomerParameter* attribute.

PORS will assign the path to Passport trunks that have an equal or lower number associated with them. This is illustrated in the example shown in figure "Path using a requiredCustomerParameter of 4" (page 114).

All restrictions are applied simultaneously during route selection. Over-restricting Passport trunks and *Plc* components may result in no route being selected, where different Passport trunks would be rejected for different reasons.

Figure 22
Path using a requiredCustomerParameter of 4



Specifying a path manually

PORS is designed to select an appropriate route automatically. In an exceptional case, however, you may wish to define the set of Passport trunks that are to be used.

The route can be defined at both end points. The two routes do not have to use the same set of Passport trunks. If different routes are defined at each end, PORS does not guarantee which one will be used.

Defining different routes at both end points has an advantage. This simple provisioning provides a backup route for manual path in case of a failure impacting the route in use. Different manual routes enhance robustness of Voice Transport.

If you want to override the automatic selection of a path and specify the Passport trunks manually, use the *manualPath* attribute of the *Plc* component. Enter the outbound sequence of Trunk component names for the path that you want.

> ***Note:*** The path still must satisfy the characteristics specified in the other *Plc* component attributes, including bandwidth requirements.

Chapter 8
Voice Transport application examples

The following sections describe how to provision Voice Transport for typical DS1, E1, and TTC PBX trunk interconnections:

To provision the *overrideRemoteName* attribute, see “Setting the overrideRemoteName attribute” (page 154). For information on how to set particular attributes, see “Provisioning notes” (page 155). The appropriate note number appears in bold type to the right of the display of provisioning data (for example, ***5c**).

Since each network has unique requirements, the values contained in the following provisioning procedures are for illustrative purposes only. Procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119) contains the largest number of comments. Successive procedures only contain comments that highlight key points in the provisioning procedure. Return to procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119) if you require more information on a step that the examples have in common.

If you plan to deploy interworking between voice and multipurpose voice platform enhanced echo cancellation (MVP-E) function processors (FPs) in your Voice Transport network, see “Interworking provisioning considerations” (page 46). You can also refer to the information contained in “System parameters of Voice Transport” (page 90).

Guidelines for configuring channel associated signaling

Voice FPs support non-standard settings for the *signalBits* attribute when you configure channel associated signaling (CAS). The *signalBits* attribute helps define CAS parameters for DS1 and E1 interfaces. On voice FPs when the *linetype* attribute is set to d4Cas or esfCas for a DS1 interface and cas for an E1 interface, and the *casSignalling* attribute is set to transparent or interpret, the *signalBits* attribute can specify a number of different signaling bit combinations.

MVP-E FPs only support the following standard settings:

- for an E1 interface with the *linetype* attribute set to cas and the *casSignalling* attribute set to transparent or interpret, the signalBits attribute must be set to ABCD
- for a DS1 interface with the *linetype* attribute set to d4Cas and the *casSignalling* attribute set to transparent or interpret, the *signalBits* attribute must be set to AB
- for a DS1 interface with the *linetype* attribute set to esfCas and the *casSignalling* attribute set to transparent or interpret, the *signalBits* attribute must be set to ABCD

Connecting identical DS1 ESF CAS PBX trunks

The following provisioning procedure uses, as an example, vs/801 on lp/8 as one end of the connection. Provision the other end using vs/501 on lp/5 exactly the same as vs/801 on lp/8.

Procedure 1
Connecting identical DS1 ESF CAS PBX trunks

1 Enter the provisioning mode.

`start Prov`

2 Add the function processor (FP) to the shelf.

`set shelf card/8 cardType <FP>`

where:
<FP> is 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*. For DS1 voice FPs, the cardType attribute must contain the same value at both ends of the connection. However, you can interconnect 1-port and 4-port DS1 MVP-E FPs.

3 Provision a logical processor to run on the card.

`add lp/8`

4 Verify that this lpt is defined.

`d sw lpt/vtds`

```
Sw Lpt/VTDS
commentText = ""
featureList = vtds
systemConfig = default
logicalProcessors = Lp/1,Lp/2,Lp/3,Lp/4,Lp/5,Lp/6,
Lp/7,Lp/8,Lp/9,Lp/10,Lp/11,Lp/12,Lp/13,Lp/14
```

If you plan to use G.728, G.729, or G.729A voice encoding, tandem pass through, or fax demodulation (fax relay) on an MVP-E FP, you must add the corresponding features using the following command:

`set sw lpt/vtds featureList g728 g729 tandemPassThrough faxRelay`

5 Link the lp to the card.

`set lp/8 mainCard shelf card/8`

6 Define the software to be loaded.

`set lp/8 logicalProcessorType sw lpt/vtds`

7 Verify that the lp and card are linked.

`d shelf card/8`

```
Shelf Card/8
cardType = 1pDs1Mvpe or 4pDs1Mvpe
configuredLPs = Lp/8
```

8 Verify that the lp is running the desired lpt software.

`d lp/8`

```
mainCard = Shelf Card/8
spareCard =
logicalProcessorType = Sw Lpt/VTDS
customerIdentifier = 0
```

9 Add a DS1 port.

`add lp/8 ds1/0`

The following components have been created:

`lp/8 ds1/0`

`lp/8 ds1/0 chan/0`

Note: If the FP is a 4-port DS1 MVP-E, the DS1 port number can be a number between 0 - 3.

10 Set the desired DS1 interface. This example shows a typical ESF CAS PBX interface.

`set lp/8 ds1/0 lineType esfcas`

`set lp/8 ds1/0 zeroCoding b8zs`

`set lp/8 ds1/0 raiAlarmType fdl`

11 Include the card in network clock synchronization.

`set lp/8 ds1/0 clockingSource module`

12 Verify that the interface and network clock synchronization are defined.

`d lp/8 ds1/0`

```
Lp/8 DS1/0
lineType = ESFCAS
zeroCoding = b8zs
clockingSource = module
raiAlarmType = fdl
lineLength = 0
customerIdentifier = 0
vendor = ""
commentText = ""
```

13 To simplify provisioning, delete channel 0. Deleting channel 0 allows you to align channel and timeslot numbers as you add channels and timeslots.

del lp/8 ds1/0 chan/0

14 Add a channel and set a timeslot. There is only one timeslot per channel for Voice Transport.

add lp/8 ds1/0 chan/1

set lp/8 ds1/0 chan/1 timeslots 1

Verify that the timeslot is defined.

d lp/8 ds1/0 chan/1

```
Lp/8 DS1/0 Chan/1
timeSlots = 1
applicationFramerName =
customerIdentifier = 0
```

15 Define more voice channels, if desired. For DS1 ports using CAS, you can add up to 24 channels.

add lp/8 ds1/0 chan/2

.
.
.

add lp/8 ds1/0 chan/24

16 Set a timeslot for each channel you add.

set lp/8 ds1/0 chan/2 timeslots 2

.
.
.

set lp/8 ds1/0 chan/24 timeslots 24

17 Add a voice application for each channel.

add vs/801

The following components have been created:

```
Vs/801 Framer
Vs/801 Plc
Vs/801
```

18 Set the remote name of the voice application at the far end of the Passport 7400 network.

set vs/801 plc remoteName "em/noder17 vs/501"

During end-to-end negotiation, if the *Vs Plc* component's *remoteName* attribute is set incorrectly (for example, because of improper syntax), a connection attempt fails even if the source and destination FPs share compatible provisioning data. If you do not specify a value for the *remoteName* attribute, the *Vs Plc* component accepts connection requests from any remote *Vs Plc* component. However, with no value specified for the *remoteName* attribute, the *Vs Plc* component cannot originate a connection request and the connection establishment process is slower.

19 Link this voice application with the desired voice channel.

set vs/801 framer interfaceName lp/8 ds1/0 chan/1

20 Enable CAS signalling to transport the ABCD (for ESF) signalling bits across the network.

set vs/801 framer casSignalling transparent

set vs/801 framer signalBits abcd

21 View the provisioning changes.

d vs/801 framer

```
Vs/801 Framer
interfaceName = Lp/8 DS1/0 Chan/1

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s  *1 (See
"Provisioning notes" (page 155))
minVoiceG711G726Rate = 24 kbit/s
modemFaxEncoding = g711G726
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on  *4
ecanBypassMode = g165
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec  *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion  = off     *5
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
transmitBusyYellow = no
transmitCasYellow = no
casSignalling = transparent  *7
invertBits = no
signalBits = ABCD  *9
idleCode = A : 0
           B : 0
           C : 0
           D : 0
seizeCode = A : 1
            B : 1
            C : 1
             D : 1
```

22 Review the default settings of the PLC component.

d vs/801 plc

```
Vs/801 Plc
remoteName = em/noder17 vs/501
setupPriority = 2
bumpPreference = bumpWhenNecessary
holdingPriority = 2
requiredTxBandwidth = 32000 bit/s
requiredRxBandwidth = 32000 bit/s
requiredTrafficType = voice
permittedTrunkTypes = terrestrial satellite tt1 tt2
tt3 ~tt4 ~tt5~tt6
requiredSecurity    = 4
requiredCustomerParm = 4
pathAttributeToMinimize = cost
maximumAcceptableCost   = 1280
maximumAcceptableDelay = 100000 msec
emissionPriority = 0
discardPriority = 1
pathType = normal
manualPath = 0 : ""
             1 : ""
             2 : ""

             :

             8 : ""
             9 : ""
pathFailureAction = reRoutePath
optimization = enabled
```

23 If necessary, set the required bandwidth to equal the provisioned compression rate. Bandwidth reservation is based on the provisioned encoding for the particular type of connection—voice, modem or facsimile. Review the default values offered by Voice Transport—particularly for connections using g728 or g729 voice encoding—and modify them if they do not meet the needs of your particular traffic and encoding type.

set vs/801 plc requiredTxBandwidth <value>

set vs/801 plc requiredRxBandwidth <value>

where:
<value> is the number of kbit/s corresponding to the compression algorithm you specify

24 Exit the provisioning mode.

end Prov

Connecting identical E1 CAS PBX trunks

The following provisioning procedure uses, as an example, vs/501 on lp/5 as one end of the connection. Provision the other end using vs/801 on lp/8 exactly the same as vs/501 on lp/5.

Procedure 2
Connecting identical E1 CAS PBX trunks

1 Follow the commands below. For more information on each of the commands, refer to the similar scenario described in procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119).

start Prov

set shelf card/5 cardType <FP>

where:
<FP> is 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*. For E1 voice FPs, the cardType attribute must contain the same value at both ends of the connection. However, you can interconnect 1-port and 4-port E1 MVP-E FPs.

add lp/5

set lp/5 mainCard shelf card/5

set lp/5 logicalProcessorType sw lpt/vtds

If you plan to use G.728, G.729, or G.729A voice encoding, tandem pass through, or fax demodulation (fax relay) on an MVP-E FP, you must add the corresponding features using the following command:

set sw lpt/vtds featureList g728 g729 tandemPassThrough faxRelay

add lp/5 e1/0

The following components are created:

```
lp/5 e1/0
```

```
lp/5 e1/0 chan/0
```

Note: If the FP is a 4-port E1 MVP-E, the E1 port number can be a number between 0 -3.

`set lp/5 e1/0 lineType cas`

`set lp/5 e1/0 clockingSource module`

`d lp/5 e1/0`

```
Lp/5 E1/0
lineType = CAS
clockingSource = module
crc4Mode = Off
customerIdentifier = 0
```

For E1 CAS, you can provision up to 30 channel and timeslot pairings. You cannot assign timeslot 16 (it carries signaling information) to a channel. To simplify provisioning, align channel and timeslot numbers as demonstrated.

`del lp/5 e1/0 chan/0`

```
add lp/5 e1/0 chan/1
set lp/5 e1/0 chan/1 timeslots 1

.
.
.

add lp/5 e1/0 chan/15

set lp/5 e1/0 chan/15 timeslot 15

add lp/5 e1/0 chan/17

set lp/5 e1/0 chan/17 timeslot 17

.
.
.

add lp/5 e1/0 chan/31

set lp/5 e1/0 chan/31 timeslot 31
```

`add vs/501`

`set vs/501 plc remoteName "em/noder17 vs/801"`

During end-to-end negotiation, if the *Vs Plc* component's *remoteName* attribute is set incorrectly (for example, because of improper syntax), a connection attempt fails even if the source and destination FPs share compatible provisioning data. If you do not specify a value for the *remoteName* attribute, the *Vs Plc* component accepts connection requests from any remote *Vs Plc* component. However, with no value specified for the *remoteName* attribute, the *Vs Plc* component cannot originate a connection request and the connection establishment process is slower.

set vs/501 plc requiredTxBandwidth 32000

set vs/501 plc requiredRxBandwidth 32000

2 Link this voice application with the desired voice channel.

set vs/501 framer interfaceName lp/5 e1/0 chan/1

Note: Bandwidth reservation is based on the provisioned encoding for the particular type of connection—voice, modem or facsimile. Review the default values offered by Voice Transport—particularly for connections using g728 or g729 encoding—and modify them if they do not meet the needs of your particular traffic and encoding type.

3 View the provisioned Permanent Logical Connection (PLC).

d vs/501 plc

```
Vs/501 Plc
remoteName = em/noder17 vs/801
setupPriority = 2
bumpPreference = bumpWhenNecessary
holdingPriority = 2
requiredTxBandwidth = 32000 bit/s
requiredRxBandwidth = 32000 bit/s
requiredTrafficType = voice
permittedTrunkTypes = terrestrial satellite tt1 tt2
tt3 ~tt4 ~tt5~tt6
requiredSecurity    = 4
requiredCustomerParm = 4
pathAttributeToMinimize = cost
maximumAcceptableCost   = 1280
maximumAcceptableDelay = 100000 msec
emissionPriority = 0
discardPriority = 1
pathType = normal
manualPath = 0 : ""
```

```
            1 : ""
            2 : ""

            :

            8 : ""
            9 : ""
pathFailureAction = reRoutePath
optimization = enabled
```

4 Enable the CAS signalling. Transport the ABCD (for E1 CAS) signalling bits across the network.

set vs/501 framer casSignalling transparent

set vs/501 framer signalBits abcd

5 View the provisioned frames.

d vs/501 framer

```
Vs/501 Framer
interfaceName = Lp/5 E1/0 Chan/1

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s  *1 (See
“Provisioning notes” (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec
echoCancellation   = on  *4
ecanBypassMode = g165
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec  *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
```

```
aLawConversion     = off  *5
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
transmitBusyYellow = no
transmitCasYellow = no
casSignalling = transparent  *7
invertBits = no
signalBits = ABCD  *9
idleCode = A : 0
           B : 0
           C : 0
           D : 0
seizeCode = A : 1
            B : 1
            C : 1
             D : 1
```

6 Exit the provisioning mode.

`end Prov`

Connecting identical TTC CAS PBX trunks

The following provisioning procedure uses, as an example, vs/501 on lp/5 as one end of the connection. Provision the other end using lp/8 and vs/801 exactly the same as vs/501 on lp/5.

Procedure 3
Connecting identical TTC2M CAS PBX trunks

1 Follow the steps below. For more information on each of the steps, refer to the similar scenario described in procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119).

Note 1: In this scenario, 4-port MVP-E FPs cannot be used. These FPs do not interwork with TTC FPs.

Note: The TTC2M MVP-E FPs use an E1 port.

`start Prov`

`set shelf card/5 cardType <FP>`

where:
<FP> id 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. The cardType attribute must contain the same value at both ends of the connection.

add lp/5

set lp/5 mainCard shelf card/5

set lp/5 logicalProcessorType sw lpt/vtds

If you plan to use G.728, G.729, or G.729A voice encoding, tandem pass through, or fax demodulation (fax relay) on an MVP-E FP, you must add the corresponding features using the following command:

set sw lpt/vtds featureList g728 g729 tandemPassThrough faxRelay

add lp/5 e1/0

set lp/5 e1/0 lineType cas

set lp/5 e1/0 clockingSource module

d lp/5 e1/0

```
Lp/5 E1/0
lineType = CAS
clockingSource = module
crc4Mode = Off
customerIdentifier = 0
```

del lp/5 e1/0 chan/0

add lp/5 e1/0 chan/1

set lp/5 e1/0 chan/1 timeslots 1 ...

You can provision up to 30 channel and timeslot pairings. To simplify provisioning, define instances from 1 to 31 (do not use or define channel or timeslot 16).

add vs/501

set vs/501 plc remoteName "em/noder17 vs/501"

During end-to-end negotiation, if the *Vs Plc* component's *remoteName* attribute is set incorrectly (for example, because of improper syntax), a connection attempt fails even if the source and destination FPs share compatible provisioning data. If you do not specify a value for the *remoteName* attribute, the *Vs Plc* component accepts connection requests from any remote *Vs Plc* component. However, with no value specified for the *remoteName* attribute, the *Vs Plc* component cannot originate a connection request and the connection establishment process is slower.

`set vs/501 plc requiredTxBandwidth 32000`

`set vs/501 plc requiredRxBandwidth 32000`

Note: Bandwidth reservation is based on the provisioned encoding for the particular type of connection—voice, modem or facsimile. Review the default values offered by Voice Transport—particularly for connections using g728 or g729 encoding—and modify them if they do not meet the needs of your particular traffic and encoding type.

2 View the provisioned Permanent Logical Connection (PLC).

`d vs/501 plc`

3 Link this voice application with the desired voice channel.

`set vs/501 framer interfaceName lp/5 e1/0 chan/1`

4 Enable the CAS signalling. Transport the ABCD (for E1 CAS) signalling bits across the network.

`set vs/501 framer casSignalling transparent`

`set vs/501 framer signalBits a`

Note: Only A-bit signalling is allowed on the TTC2M MVP-E FPs.

5 View the provisioned frames.

`d vs/501 framer`

```
Vs/501 Framer
interfaceName = Lp/5 E1/0 Chan/1

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s  *1 (See
“Provisioning notes” (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on  *4
ecanBypassMode = g165
echoTailDelay = 64 msec
```

```
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec  *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion     = off  *5
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
transmitBusyYellow = no
transmitCasYellow = no
casSignalling = transparent  *7
invertBits = no
signalBits = A  *9
idleCode = A : 1
           B : 0
           C : 0
           D : 0
seizeCode = A : 0
            B : 1
            C : 1
             D : 1
```

6 Exit the provisioning mode.

`end Prov`

Connecting DS1 ESF CAS and DS1 SF (D4) CAS PBX trunks

In this procedure, vs/201 is on a DS1 D4 CAS trunk and vs/801 is on a DS1 ESF CAS trunk. Provision the logical processors and cards as before.

> ***Note:*** For applications where *casSignalling* is set to interpret, there can only be two valid signalling codes on the trunk. This means that wink start trunks must have identical idle and seize code values.

Procedure 4
Connecting DS1 ESF CAS and DS1 SF (D4) CAS PBX trunks

1 Follow the steps below. For more information on each of the steps, refer to the similar scenario described in procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119).

start Prov

add lp/2 ds1/0

add lp/8 ds1/0

set lp/8 ds1/0 lineType esfcas

set lp/8 ds1/0 zeroCoding b8zs

set lp/8 ds1/0 clockingSource module

set lp/8 ds1/0 raiAlarmType fdl

d lp/8 ds1/0

```
Lp/8 DS1/0
lineType = ESFCAS
zeroCoding = b8zs
clockingSource = module
raiAlarmType = fdl
lineLength = 0
customerIdentifier = 0
vendor = ""
commentText = ""
```

del lp/8 ds1/0 chan/0

add lp/8 ds1/0 chan/1

set lp/8 ds1/0 chan/1 timeslots 1

set lp/2 ds1/0 lineType d4cas

set lp/2 ds1/0 zeroCoding bit7stuffing

set lp/2 ds1/0 clockingSource module

set lp/2 ds1/0 raiAlarmType bit2

d lp/2 ds1/0

```
Lp/2 DS1/0
lineType = D4CAS
zeroCoding = Bit7Stuffing
```

```
clockingSource = module
raiAlarmType = Bit2
lineLength = 0
customerIdentifier = 0
vendor = ""
commentText = ""
```

del lp/2 ds1/0 chan/0

add lp/2 ds1/0 chan/1

set lp/2 ds1/0 chan/1 timeslots 1

add vs/201

add vs/801

set vs/201 framer interfaceName lp/2 ds1/0 chan/1

set vs/801 framer interfaceName lp/8 ds1/0 chan/1

set vs/201 plc remoteName "em/noder17 vs/801"

set vs/801 plc remoteName "em/noder32 vs/201"

During end-to-end negotiation, if the *Vs Plc* component's *remoteName* attribute is set incorrectly (for example, because of improper syntax), a connection attempt fails even if the source and destination FPs share compatible provisioning data. If you do not specify a value for the *remoteName* attribute, the *Vs Plc* component accepts connection requests from any remote *Vs Plc* component. However, with no value specified for the *remoteName* attribute, the *Vs Plc* component cannot originate a connection request and the connection establishment process is slower.

2 Enable the CAS signalling. Since both ends are using different signalling formats—AB (for D4) and ABCD (for ESF)—specify that signalling information be interpreted for idle and seize codes.

set vs/201 framer casSignalling interpret

set vs/201 framer signalBits ab

set vs/801 framer casSignalling interpret

set vs/801 framer signalBits abcd

3 Set *transmitBusyYellow* to yes if you want the function processor (FP) to send a busy ABCD-signalling state to the connected PBX for this voice channel whenever the voice path is down.

set vs/201 framer transmitBusyYellow yes

set vs/801 framer transmitBusyYellow yes

4 Set *transmitCasYellow* to yes if you want the FP to transmit a yellow alarm condition to the connected PBX whenever the voice path is down.

set vs/201 framer transmitCasYellow yes

set vs/801 framer transmitCasYellow yes

Note: Set this attribute to yes for all voice services interfacing to the same port. Otherwise, a yellow alarm could be transmitted when channels are still available.

5 Set the *idleCode* and *seizeCode* attributes on each FP. These are typical values for E&M 4-wire TIE trunk protocols.

set vs/201 framer idleCode a 0 b 0 c 0 d 0

set vs/201 framer seizeCode a 1 b 1 c 1 d 1

set vs/801 framer idleCode a 0 b 0 c 0 d 0

set vs/801 framer seizeCode a 1 b 1 c 1 d 1

Note 1: When *casSignalling* is set to interpret, each end's *idleCode* and *seizeCode* attributes must match the idle and seize codes of the connected PBX.

Note 2: Since *signalBits* for vs/201is set to AB, the C and D bits are ignored.

d vs/201 framer

```
Vs/201 Framer
interfaceName = Lp/2 DS1/0 Chan/1

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s  *1 (See
"Provisioning notes" (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on  *4
ecanBypassMode = g165
```

```
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec  *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion     = off  *5
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
transmitBusyYellow = yes  *6
transmitCasYellow = yes  *6a
casSignalling = interpret  *8
invertBits = no
signalBits = AB  *9b
idleCode = A : 0
           B : 0
           C : 0
           D : 0
seizeCode = A : 1
            B : 1
            C : 1
             D : 1
```

d vs/801 framer

```
Vs/801 Framer
interfaceName = Lp/8 DS1/0 Chan/1

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s  *1 (See
"Provisioning notes" (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on *4
ecanBypassMode = g165
```

```
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec  *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion     = off  *5
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
transmitBusyYellow = yes  *6
transmitCasYellow = yes  *6a
casSignalling = interpret  *8
invertBits = no
signalBits = ABCD  *9
idleCode = A : 0
           B : 0
           C : 0
           D : 0
seizeCode = A : 1
            B : 1
            C : 1
             D : 1
```

6 Exit the provisioning mode.

end Prov

Connecting DS1 ESF CAS and E1 CAS PBX trunks

In this procedure, vs/501 is on an E1 CAS trunk and vs/801 is on a DS1 ESF CAS trunk. Provision the logical processors and cards as before.

> ***Note:*** For applications where the *casSignalling* attribute is set to interpret, there can only be two valid signalling codes on the trunk. This means that wink start trunks must have identical idle and seize code values.

Procedure 5
Connecting DS1 ESF CAS and E1 CAS PBX trunks

1 Follow the steps below. For more information on each of the steps, refer to the similar scenario described in procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119).

start Prov

add lp/5 e1/0

set lp/5 e1/0 lineType cas

set lp/5 e1/0 clockingSource module

d lp/5 e1/0

```
Lp/5 E1/0
lineType = CAS
clockingSource = module
crc4Mode = Off
customerIdentifier = 0
vendor = ""
commentText = ""
```

del lp/5 e1/0 chan/0

add lp/5 e1/0 chan/1

set lp/5 e1/0 chan/1 timeslots 1

add lp/8 ds1/0

set lp/8 ds1/0 lineType esfcas

set lp/8 ds1/0 zeroCoding b8zs

set lp/8 ds1/0 clockingSource module

set lp/8 ds1/0 raiAlarmType fdl

d lp/8 ds1/0

```
Lp/8 DS1/0
lineType = ESFCAS
zeroCoding = b8zs
clockingSource = module
raiAlarmType = fdl
lineLength = 0
customerIdentifier = 0
vendor = ""
commentText = ""
```

del lp/8 ds1/0 chan/0

add lp/8 ds1/0 chan/1

set lp/8 ds1/0 chan/1 timeslots 1

```
add vs/501

add vs/801

set vs/501 framer interfaceName lp/5 e1/0 chan/1

set vs/801 framer interfaceName lp/8 ds1/0 chan/1

set vs/501 plc remoteName "em/noder17 vs/801"

set vs/801 plc remoteName "em/noder32 vs/501"
```

During end-to-end negotiation, if the *Vs Plc* component's *remoteName* attribute is set incorrectly (for example, because of improper syntax), a connection attempt fails even if the source and destination FPs share compatible provisioning data. If you do not specify a value for the *remoteName* attribute, the *Vs Plc* component accepts connection requests from any remote *Vs Plc* component. However, with no value specified for the *remoteName* attribute, the *Vs Plc* component cannot originate a connection request and the connection establishment process is slower.

2 Set *transmitBusyYellow* to yes if you want the function processor (FP) to send a busy ABCD-signalling state to the connected PBX for this voice channel whenever the voice path is down.

```
set vs/501 framer transmitBusyYellow yes

set vs/801 framer transmitBusyYellow yes
```

3 Set *transmitCasYellow* to yes if you want the FP to transmit a yellow alarm to the connected PBX whenever the voice path is down.

```
set vs/501 framer transmitCasYellow yes

set vs/801 framer transmitCasYellow yes
```

Note: Set this attribute to yes for all voice services interfacing to the same port. Otherwise, a yellow alarm could be transmitted when channels are still available.

4 Enable the CAS signalling. Since both ends are using different signalling formats—ABCD (for E1 CAS) and ABCD (for DS1 ESF)—specify that signalling information be interpreted for idle and seize codes.

```
set vs/501 framer casSignalling interpret

set vs/501 framer signalBits abcd

set vs/801 framer casSignalling interpret

set vs/801 framer signalBits abcd
```

5 Set the *idleCode* and *seizeCode* attributes on each FP. When *casSignalling* is set to interpret, each end's *idleCode* and *seizeCode* attributes must match the idle and seize codes of the connected PBX. Only two signalling codes—to identify idle, seize, dial pulse, or winks—are allowed.

set vs/501 framer idleCode a 1 b 1 c 0 d 1

set vs/501 framer seizeCode a 0 b 1 c 0 d 1

set vs/801 framer idleCode a 0 b 0 c 0 d 0

set vs/801 framer seizeCode a 1 b 1 c 1 d 1

6 Convert between DS1 mu-Law to E1 A-Law. The 4-port DS1 MVP-E FP does the conversion on both the ingress and egress sides. Therefore, only set this attribute on a DS1 *Vs* component.

Note: Set the E1 PBX trunk card to A-Law.

set vs/801 framer aLawConversion on

d vs/501 framer

```
Vs/501 Framer
interfaceName = Lp/5 E1/0 Chan/1

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s *1 (See "Provisioning
notes" (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on  *4
ecanBypassMode = g165
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec   *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion     = off  *5b
```

```
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
transmitBusyYellow = yes  *6
transmitCasYellow = yes  *6a
casSignalling = interpret  *8
invertBits = no
signalBits = ABCD   *9b
idleCode = A : 1
           B : 1
           C : 0
           D : 1
seizeCode = A : 0
            B : 1
            C : 0
             D : 1
```

d vs/801 framer

```
Vs/801 Framer
interfaceName = Lp/8 DS1/0 Chan/1

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s  *1
                             (See "Provisioning
notes" (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on  *4
ecanBypassMode = g165
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec   *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
```

```
aLawConversion     = on  *5b
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
transmitBusyYellow = yes  *6
transmitCasYellow = yes  *6a
casSignalling = interpret  *8
invertBits = no
signalBits = ABCD  *9
idleCode = A : 0
           B : 0
           C : 0
           D : 0
seizeCode = A : 1
            B : 1
            C : 1
             D : 1
```

7 Exit the provisioning mode.

end Prov

Connecting DS1 ESF CAS and TTC CAS PBX trunks

In this procedure, vs/501 is on an TTC CAS trunk and vs/801 is on a DS1 ESF CAS trunk. Provision the logical processors and cards as before.

> ***Note:*** For applications where the *casSignalling* attribute is set to interpret, there can only be two valid signalling codes on the trunk. This means that wink start trunks must have identical idle and seize code values.

Procedure 6
Connecting DS1 ESF CAS and TTC CAS PBX trunks

1 Follow the steps below. For more information on each of the steps, refer to the similar scenario described i n procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119).

start Prov

add lp/5 e1/0

set lp/5 e1/0 lineType cas

set lp/5 e1/0 clockingSource module

```
d lp/5 e1/0

Lp/5 E1/0
lineType = CAS
clockingSource = module
crc4Mode = Off
customerIdentifier = 0
vendor = ""
commentText = ""

del lp/5 e1/0 chan/0

add lp/5 e1/0 chan/1

set lp/5 e1/0 chan/1 timeslots 1

add lp/8 ds1/0

set lp/8 ds1/0 lineType esfcas

set lp/8 ds1/0 zeroCoding b8zs

set lp/8 ds1/0 clockingSource module

set lp/8 ds1/0 raiAlarmType fdl

d lp/8 ds1/0

Lp/8 DS1/0
lineType = ESFCAS
zeroCoding = b8zs
clockingSource = module
raiAlarmType = fdl
lineLength = 0
customerIdentifier = 0
vendor = ""
commentText = ""

del lp/8 ds1/0 chan/0

add lp/8 ds1/0 chan/1

set lp/8 ds1/0 chan/0 timeslots 1

add vs/501

add vs/801

set vs/501 framer interfaceName lp/5 e1/0 chan/1

set vs/801 framer interfaceName lp/8 ds1/0 chan/1
```

set vs/501 plc remoteName "em/noder17 vs/801"

set vs/801 plc remoteName "em/noder32 vs/501"

During end-to-end negotiation, if the *Vs Plc* component's *remoteName* attribute is set incorrectly (for example, because of improper syntax), a connection attempt fails even if the source and destination FPs share compatible provisioning data. If you do not specify a value for the *remoteName* attribute, the *Vs Plc* component accepts connection requests from any remote *Vs Plc* component. However, with no value specified for the *remoteName* attribute, the *Vs Plc* component cannot originate a connection request and the connection establishment process is slower.

2 Set *transmitBusyYellow* to yes if you want the function processor (FP) to send a busy ABCD-signalling state to the connected PBX for this voice channel whenever the voice path is down.

set vs/801 framer transmitBusyYellow yes

3 Set *transmitCasYellow* to yes if you want the FP to transmit a yellow alarm to the connected PBX whenever the voice path is down.

set vs/501 framer transmitCasYellow yes

set vs/801 framer transmitCasYellow yes

Note: Set this attribute to yes for all voice services interfacing to the same port. Otherwise, a yellow alarm could be transmitted when channels are still available.

4 Enable the CAS signalling. Since both ends are using different signalling formats—ABCD (for DS1 ESF) and A (for TTC CAS)—specify that signalling information be interpreted for idle and seize codes.

set vs/501 framer casSignalling interpret

set vs/501 framer signalBits a

set vs/801 framer casSignalling interpret

set vs/801 framer signalBits abcd

5 Set the *idleCode* and *seizeCode* attributes on each FP.

set vs/501 framer idleCode a 1

set vs/501 framer seizeCode a 0

set vs/801 framer idleCode a 0 b 0 c 0 d 0

set vs/801 framer seizeCode a 1 b 1 c 1 d 1

Note: When *casSignalling* is set to interpret, each end's *idleCode* and *seizeCode* attributes must match the idle and seize codes of the connected PBX.

6 Convert between DS1 mu-Law to A-Law. The 4-port DS1 MVP-E FP does the conversion on both ingress and egress. Therefore only set this attribute on the DS1 VS application. Setting this attribute on the J2M voice or TTC2M MVP FP has no effect.

Although TTC is mu-Law in the PBX network, it is converted to A-Law within the Passport 7400 network by the J2M voice or TTC2M MVP FP. Therefore, the DS1 MVP-E FP must have *aLawConversion* set to on to convert the TTC signal back to mu-Law.

`set vs/801 framer aLawConversion on`

`d vs/501 framer`

```
Vs/501 Framer
interfaceName = Lp/5 E1/0 Chan/0

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s *1 (See "Provisioning
notes" (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on  *4
ecanBypassMode = g165
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec   *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion     = off  *5b
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
```

```
transmitBusyYellow = no  *6
transmitCasYellow = yes  *6a
casSignalling = interpret  *8
invertBits = no
signalBits = A  *9c
idleCode = A : 1
seizeCode = A : 0
```

d vs/801 framer

```
Vs/801 Framer
interfaceName = Lp/8 DS1/0 Chan/0

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s  *1 (See
“Provisioning notes” (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on  *4
ecanBypassMode = g165
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec   *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion     = on  *5b
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
transmitBusyYellow = yes  *6
transmitCasYellow = yes  *6a
casSignalling = interpret  *8
invertBits = no
signalBits = ABCD  *9
idleCode = A : 0
           B : 0
```

```
           C : 0
           D : 0
seizeCode = A : 1
            B : 1
            C : 1
             D : 1
```

7 Exit the provisioning mode.

end Prov

Connecting DS1 CCS and E1 CCS PBX trunks

This procedure applies to connecting a DS1 MVP-E FP to an E1 MVP-E FP or a 4-port DS1 MVP-E FP to a 4-port E1 MVP-E FP. Set up logical processors and cards as in the previous procedures. Any voice service (*Vs* component) attributes not mentioned should be set up as before.

> ***Note:*** The following procedure requires you to configure a bit transparent data service (BTDS). For more information on BTDS, see 241-7401-775 *Passport 7400 Bit Transparent Data Service Guide*.

Procedure 7
Connecting DS1 CCS and E1 CCS PBX trunks

1 Follow the steps below. For more information on each of the steps, refer to the similar scenario described in procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119).

start Prov

set lp/8 ds1/0 linetype esf

set lp/5 e1/0 linetype ccs

Note: In this procedure, you can interconnect both 1-port and 4-port DS1 MVP-E FPs and 1-port and 4-port E1 MVP-E FPs.

2 Align the channels and timeslots for CCS.

set lp/8 ds1/0 chan/0 timeslot 24

set lp/5 e1/0 chan/16 timeslot 16

3 Add a BTDS for the CCS channel. Typically DS1 is timeslot 24 while E1 is timeslot 16.

add btds/824

```
Btds/824
The following components have been created:
Btds/824 Framer
Btds/824 Plc
Btds/824
```

`add btds/516`

`set btds/824 plc remoteName "em/noder17 btds/516"`

`set btds/824 framer interfaceName lp/8 ds1/0 chan/0`

`set btds/516 plc remoteName "em/noder32 btds/824"`

`set btds/516 framer interfaceName lp/5 e1/0 chan/16`

`add vs/801`

```
Vs/801
```

```
The following components have been created:
Vs/801 Framer
Vs/801 Plc
Vs/801
```

`add vs/501`

Note: Since the PBX CCS protocol sends out setup messages with the timeslot number, timeslots must match on either end of the Passport 7400 network. For example, timeslot 1 must go to timeslot 1.

`set vs/801 plc remoteName "em/noder17 vs/501"`

`set vs/801 framer interfaceName lp/8 ds1/0 chan/1`

`set vs/501 plc remoteName "em/noder32 vs/801"`

`set vs/501 framer interfaceName lp/5 e1/0 chan/1`

During end-to-end negotiation, if the *Vs Plc* component's *remoteName* attribute is set incorrectly (for example, because of improper syntax), a connection attempt fails even if the source and destination FPs share compatible provisioning data. If you do not specify a value for the *remoteName* attribute, the *Vs Plc* component accepts connection requests from any remote *Vs Plc* component. However, with no value specified for the *remoteName* attribute, the *Vs Plc* component cannot originate a connection request and the connection establishment process is slower.

4 Set up the VS for CCS signalling.

`set vs/801 framer casSignalling none`

`set vs/501 framer casSignalling none`

Note: Setting *casSignalling* to none makes the following attributes irrelevant: *invertBits*, *signalBits*, *idleCode* and *seizeCode*.

`set vs/801 framer transmitBusyYellow yes`

`set vs/501 framer transmitBusyYellow yes`

Note: Only set *transmitBusyYellow* to yes if you want the FP to send a yellow alarm state to the connected PBX whenever this voice path is down.

`d vs/801 framer`

```
Vs/801 Framer
interfaceName = Lp/8 DS1/0 Chan/1

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s  *1 (See
"Provisioning notes" (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on  *4
ecanBypassMode = g165
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec   *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion     = on  *5b
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
transmitBusyYellow = yes  *6
transmitCasYellow = no
```

```
casSignalling = none
invertBits = no
signalBits = A
idleCode = A : 0
           B : 0
           C : 0
           D : 0
seizeCode = A : 1
            B : 1
            C : 1
             D : 1
```

d vs/501 framer

```
Vs/501 Framer
interfaceName = Lp/5 E1/0 Chan/1

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s  *1 (See
"Provisioning notes" (page 155))
minVoiceG711G726Rate = 24 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s  *2
minModemFaxG711G726Rate = 32 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = on  *3
speechHangoverTime = 150 msec  *3a
echoCancellation   = on *4
ecanBypassMode = g165
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec   *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion     = off  *5c
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
transmitBusyYellow = yes  *6
```

```
transmitCasYellow = no
casSignalling = none
invertBits = no
signalBits = A
idleCode = A : 0
           B : 0
           C : 0
           D : 0
seizeCode = A : 1
            B : 1
            C : 1
             D : 1
```

5 Exit the provisioning mode.

`end Prov`

Video transmission

To conserve bandwidth while a video connection is idle, provision the video connection using CAS voice services. Set up logical processors, FPs, DS1 or E1 ports, and channels as before and add a *Vs* component. For information on which FPs support this feature, including support for TTC in Japan, refer to 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference*.

> ***Note:*** For applications where the *casSignalling* attribute is set to interpret, there can only be two valid signalling codes on the trunk. This means that wink start trunks must have identical idle and seize code values.

Procedure 8
Connecting DS1, E1, TTC PBX trunks for video transmission

1 Follow the steps below. For more information on each of the steps, refer to the similar scenario described in procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119).

`start Prov`

`add vs/801`

`set vs/801 framer interfaceName lp/8 ds1/0 chan/0`

`set vs/801 plc remoteName "em/noder17 vs/201"`

During end-to-end negotiation, if the *Vs Plc* component's *remoteName* attribute is set incorrectly (for example, because of improper syntax), a connection attempt fails even if the source and destination FPs share compatible provisioning data. If you do not specify a value for the *remoteName* attribute, the *Vs Plc* component accepts connection requests from any remote *Vs Plc* component. However, with no value specified for the *remoteName* attribute, the *Vs Plc* component cannot originate a connection request and the connection establishment process is slower.

2 Turn off voice-related functions.

set vs/801 framer voiceEncoding g711G726

set vs/801 framer modemFaxEncoding g711G726

set vs/801 framer maxVoiceG711G726Rate 64

set vs/801 framer minVoiceG711G726Rate 64

set vs/801 framer maxModemFaxG711G726Rate 64

set vs/801 framer minModemFaxG711G726Rate 64

set vs/801 framer echoCancellation off

set vs/801 framer silenceSuppression off

set vs/801 framer ingressAudioGain 0

set vs/801 framer egressAudioGain 0

3 Set up the *VoiceService* (*Vs*) component for CAS signalling.

set vs/801 framer casSignalling <value>

where:
<value> can be *transparent* if both ends of the connection use equivalent signalling formats; use *interpret* if both ends do not use the same signalling formats

set vs/801 framer signalBits abcd

Note: Set vs/801 Framer *signalBits* to *a* if using TTC on a TTC2M MVP-E function processor.

4 Set up idle suppression. Valid choices for *silenceSuppression* on voice FPs are off or casIdleCode. MVP-E FPs do not support the setting of casIdleCode. However, MVP-E FPs automatically stop sending frames into the subnet upon detection of a PBX's CAS idle code, thus overriding the setting of *silenceSuppression* while still conserving bandwidth.

If you set *silenceSuppression* to off, no further provisioning is required. If you choose to set silence suppression to casIdleCode on a connection involving voice FPs, then perform the following steps:

set vs/801 framer silenceSuppression casIdleCode

set vs/801 framer idleCode a 0 b 0 c 0 d 0

set vs/801 framer seizeCode a 1 b 1 c 1 d 1

Note: You must set *casSignalling* to interpret when *silenceSuppression* is set to casIdleCode.

d vs/801 framer

```
Vs/801 Framer
interfaceName = Lp/8 DS1/0 Chan/0

voiceEncoding = g711G726
maxVoiceG711G726Rate = 64 kbit/s
minVoiceG711G726Rate = 64 kbit/s
maxModemFaxG711G726Rate = 64 kbit/s
minModemFaxG711G726Rate = 64 kbit/s
ingressAudioGain = 0 dB
egressAudioGain = 0 dB
silenceSuppression = casIdleCode  *3b (See
“Provisioning notes” (page 155))
speechHangoverTime = 150 msec  *3a
echoCancellation   = off
ecanBypassMode = g165
echoTailDelay = 64 msec
echoReturnLoss = 6 dB
modemFaxSpeechDiscrim = on
modemFaxEncoding = g711G726
faxIdleSuppressionG711G726 = on
endOfCallPattern = standard  *4a
tandemPassThrough = disabled
insertedOutputDelay = default msec  *4b
faxHangoverTimeG711G726 = 1000 msec  *4c
aLawConversion     = off  *5c
dtmfRegeneration = off
comfortNoiseCap = -40 dBm0
v17EncodedAsG711G726 = no
transmitBusyYellow = yes  *6
casSignalling = yes  *8
invertBits = no
```

```
signalBits = ABCD  *9
idleCode = A : 0
           B : 0
           C : 0
           D : 0
seizeCode = A : 1
            B : 1
            C : 1
             D : 1
```

5 Exit the provisioning mode.

end Prov

Setting the overrideRemoteName attribute

The following provisioning procedure use, as examples, vs/301 and vs/1101. If needed, refer to procedure “Connecting identical DS1 ESF CAS PBX trunks” (page 119) for other provisioning information.

Procedure 9
Setting the overrideRemoteName attribute

1 Follow the steps below to provision vs/301.

start Prov

add vs/301

add lp/3 e1/1

set lp/3 e1/1 chan/0 timeSlots ! 1 2

set lp/3 e1/1 clockingSource module

set lp/3 e1/1 chan/0 applicationFramerName vs/301 framer

set vs/301 framer interface lp/3 e1/1 chan/0

set vs/301 plc remoteName “ “

2 Exit the provisioning mode.

3 Follow the steps below to provision vs/1101.

start Prov

add vs/1101

add lp/11 e1/1

set lp/11 e1/1 chan/0 timeSlots ! 1 2

set lp/11 e1/1 clockingSource module

set lp/11 e1/1 chan/0 applicationFramerName vs/1101 framer

set vs/1101 framer interface lp/11 e1/1 chan/0

set vs/1101 plc remoteName “ “

4 Exit the provisioning mode.

end Prov

5 Set the overrideRemoteName attribute.

start Prov

set vs/301 lc overrideRemoteName “em/noder1c vs/1101”

6 To display, the Lc components, type:

display vs/301 LCo

This command will display the *LCo* components. The setting of the *overrideRemoteName* attribute will override the setting for the *remoteName* attribute. The setting of this attribute will be the value provisioned above, in other words, *em/noder1c vs/1101*.

Provisioning notes

The note numbers in table “Provisioning notes for Framer component attributes” (page 156) apply to the display of Framer information listed in the preceding provisioning procedures. The appropriate note number—in bold type—appears to the right of the displayed Framer information. For example, a reference to note ***5c** below will appear in the text as follows:

d vs/801 framer

`aLawConversion = off` ***5c**

Table 14
Provisioning notes for Framer component attributes

Note number	Details
***1**	For information on how to set the *voiceEncoding*, *minVoiceG711G726Rate*, and *maxVoiceG711G726Rate* attributes, see “End-to-end negotiation provisioning guidelines and considerations” (page 41). For Voice Transport connections between voice and MVP/MVP-E FPs, see also “Interworking provisioning considerations” (page 46).
*2	For information on how to set the *modemFaxEncoding*, *minModemFaxG711G726Rate*, and *maxModemFaxG711G726Rate* attributes, see “End-to-end negotiation provisioning guidelines and considerations” (page 41). For Voice Transport connections between voice and MVP/MVP-E FPs, see also “Interworking provisioning considerations” (page 46).
*3	Delete periods of silence, for Voice Transport calls running on either voice or MVP/MVP-E FPs, to free up bandwidth on the internal Passport 7400 network.
*3a	When the negotiated value of the *silenceSuppression* attribute is on, congested, or slowAndCongested, you can adjust the default value of *speechHangoverTime* to a lower value if you require additional bandwidth savings or higher if the clipping of portions of voice conversations occurs in your network.
*3b	Frees up bandwidth during CAS idle periods (only on connections between voice FPs). Two minutes after the CAS idle code is detected by Passport 7400, the ingress channel on a voice FP goes into silence. MVP-E FPs do not support the value casIdleCode. When MVP-E FPs detect the CAS idle code (corresponding to the value provisioned under *idleBits*), they immediately stop sending frames into the subnet, thus conserving bandwidth and overriding the value set under *silenceSuppression*.
(Sheet 1 of 3)	

Table 14 (continued)
Provisioning notes for Framer component attributes

Note number	Details
*4	The default value on means that echo cancellation is enabled on voice and MVP-E FPs. The setting of the *ecanBypassMode* attribute applies only to MVP-E FPs when the *echoCancellation* attribute is set to on. The default value g165 specifies that the on-board echo canceller is placed in bypass mode according to ITU-T G.165 (that is, when receiving phase-reversed 2100 Hz tones). The other available values are g164 (places the on-board echo canceller in bypass mode upon receiving a 2100 Hz tone according to ITU-T G.164) and never (recommended for use only during debugging procedures). Also, the settings of the *echoTailDelay* attribute (for adjusting the echo delay coverage of the echo canceller) and the *echoReturnLoss* attribute (for improving echo cancellation performance when line conditions are poor and echo is a problem) only apply to MVP-E FPs when the *echoCancellation* attribute is set to on. Setting the *echoReturnLoss* attribute higher can improve echo cancellation performance, but reduce the signal level.
*4a	The value specified under the *endOfCallPattern* attribute must match the CCS idle pattern sent by the connected PBX. To verify the idle pattern, check the *recentIngressLineSamples* operational attribute on a channel with no calls currently running.
*4b	Only MVP/MVP-E FPs support the *insertedOutputDelay* attribute. The value default msec equates to 22 msec.
***4c**	Only MVP/MVP-E FPs support the *faxHangoverTimeG711G726* attribute. When the negotiated value of the *faxIdleSuppression* attribute is on, you can adjust the default value of *faxHangoverTimeG711G726* to a lower value if you require additional bandwidth savings or higher if the clipping of portions of facsimile transmissions occurs in your network. Higher values increase bandwidth usage. Available values are 300 up to 20000 msec.
*5	No conversion is required when going between identical FPs.
*5b	Only the DS1MVP-E FP does the conversion.
(Sheet 2 of 3)	

Table 14 (continued)
Provisioning notes for Framer component attributes

Note number	Details
*5c	No conversion between mu-Law and A-Law. The information coming into the Passport 7400 node is interpreted as non-voice data.
*6	Send busy signalling state when the voice path is down.
*6a	Transmit yellow alarm when the voice path is down for CAS links.
*7	Transport CAS signalling transparently to the far end. (Use when connection is between identical PBX trunks—DS1, E1 or TTC.)
*8	Interpret for idle and seize codes only. (Use when connection is between different PBX trunks—for example, DS1 to TTC).
*9	Transport all four signalling bits; number of signalling bits must be the same at both ends of a connection when *casSignalling* is set to transparent.
*9b	Transport only the AB signalling bits.
***9c**	Transport only the A signalling bit.
(Sheet 3 of 3)	

Chapter 9
Monitoring and troubleshooting Voice Transport

See the following sections for information related to monitoring and troubleshooting the Voice Transport service:

Monitoring

To determine how well your Voice Transport service is operating, use the list and display commands to monitor the service’s operational information and statistics. “Displaying Voice Transport traffic information” (page 160) provides an example of how to monitor the service to capture particular information. “Viewing end-to-end negotiation information” (page 160) provides information on the negotiated audio handling parameters for all Voice Transport connection types. For more details on interpreting Voice Transport operational information to calculate various network parameters such as bandwidth usage, contact your Nortel Networks account representative.

You can change the values of certain Voice Transport provisionable attributes as required to optimize your Passport 7400 network. However, before changing attribute values, ensure that you understand the impact of a change on the node and on the network. See 241-5701-060 *Passport 7400, 15000, 20000 Components* for descriptions of Voice Transport attributes and their associated values.

Displaying Voice Transport traffic information

To monitor the amount of voice, modem, and facsimile traffic being transmitted through the network, display the operational attributes under the *VoiceService* (*Vs*) *Framer* component's *Statistics* group. The operational attributes in the *Statistics* group count the number of cells received by the interface for each particular traffic type. Each cell represents 44 bytes of data.

Use the following command to display operational attributes containing Voice Transport statistical traffic information for all provisioned *Vs* components:

```
display VoiceService/* Framer Statistics
```

The *totalCells* operational attribute displays the total number of cells received by each particular *Vs* component instance. The following operational attributes provide a breakdown of the total number of cells according to type (for example, the traffic type, the encoding type used, or whether silence suppression was applied):

- *audioCells* applies to cells containing either voice or modem data
- *silenceCells* applies to cells without audio data
- *modemFaxCells* applies to cells containing modem and facsimile data
- *faxRelayCells* applies to cells sent to the network from each *Vs* component associated with an MVP FP when *modemFaxEncoding* is set to faxRelay

Viewing end-to-end negotiation information

Passport R5.1 and later software supports end-to-end negotiation between FPs. End-to-end negotiation can result in the successful establishment of a connection or the rejection of a connection request. The *Vs* component's *serviceFailureReason* operational attribute and the *Framer* component's *Negotiated* group of operational attributes indicate the results of end-to-end negotiation for both successful and failed connection attempts. End-to-end negotiation results include negotiated encoding choices and rates, negotiated audio handling parameters, and reasons why end-to-end negotiation either failed or succeeded. By viewing the results of end-to-end negotiation, you can isolate and correct provisioning problems.

The *Vs* component's *serviceFailureReason* operational attribute indicates the reasons a connection attempt failed or succeed. Use the following command to view the information contained under the *serviceFailureReason* attribute for a particular *Vs* component instance:

```
display VoiceService/<#> serviceFailureReason
```

For example, if end-to-end negotiation revealed incompatible voice encoding rates for a particular *Vs Framer* component, the *serviceFailureReason* attribute contains the value mismatchedVoiceRates. If this particular *Vs Framer* component uses g711G726 voice encoding, the failure can be attributed to the provisioned minimum and maximum voice rates. For more information on setting encoding choices and rates, see "End-to-end negotiation provisioning guidelines and considerations" (page 41) and "Interworking provisioning considerations" (page 46). More than one end-to-end negotiation can occur. If more than one end-to-end negotiation error occurs, only the *serviceFailureReason* attribute on the destination node records multiple error values.

The *Framer* component's *Negotiated* group of operational attributes indicate the audio handling parameters agreed upon by the source and destination FPs during end-to-end negotiation. Use the following command to view the operational attributes in the *Negotiated* group for a particular *Vs* component instance:

```
display VoiceService/<#> Framer Negotiated
```

For example, if the end-to-end negotiation process determines that tandem pass through is not supported by the destination FP, the *negotiatedTandemPassThrough* attribute value is disabled.

Alarms

Alarms are messages used to indicate faults or failure conditions on the node.

Alarms are generated asynchronously by Passport 7400 components. When a component generates an alarm, it does so to signal that it is in need of repair or that it has detected a fault elsewhere on the node.

Alarms contain a relatively large amount of information, all of which will assist you in the monitoring and surveillance of your network. Because alarms are such an important and integral part of Passport 7400 fault management, they are described separately in 241-5701-500 *Passport 6400, 7400, 15000, 20000 Alarms*. See the following sections for more information on alarms:

- “Causes of alarms” (page 162)
- “Voice Transport-related alarms” (page 162)

Causes of alarms

As a general rule, you can expect to see an alarm in the following situations:

- degradation/quality-of-service conditions (for example, if a threshold is reached)
- processing errors (for example, protocol violations)
- failures/out-of-service conditions (for example, hardware or facility failures)
- administrative conditions (for example, the lock command is issued)
- security violations

Voice Transport-related alarms

The alarms related to Voice Transport are as follows:

- 7018 0001 to 7018 0004 Path Administrator-related alarms
- 7018 1001 and 7018 1002 LCo-related alarms
- 7019 0001 *VoiceService* component-related alarm
- 7011 5006 port management-related alarm

Problem solving

The following sections contain information on the typical sources of problems and examples of how to deal with particular problems:

- “Hardware connection problems” (page 163)
- “Service connection problems” (page 163)
- “Checking service provisioning” (page 169)
- “Checking bandwidth” (page 169)
- “Troubleshooting example” (page 170)

Hardware connection problems

Some problems may be traced back to hardware connection problems. Check the cable, connections, and pins. See 241-7401-240 *Passport 7400 Hardware Installation, Maintenance and Upgrade* for more information.

For a description of function processors, see 241-7401-200 *Passport 7400 Hardware Description*. For function processor configuration information, see 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference* and 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide*.

Service connection problems

Problems with setting up connections may be due to errors or mismatches in setting up the system or provisioning. See “End-to-end negotiation provisioning guidelines and considerations” (page 41) and use the “Configuration checklists” (page 49) to check if there are any steps that you may have forgotten in the process. For information on failed connections, see the operational attribute *serviceFailureReason*.

In general, you should look for the following (see also figure “Flowchart: an example of troubleshooting using LCo” (page 164)):

- Is the path up or down?
- Where does it go down?
- Why did it go down?

Table “Handling problems” (page 165) contains information about how to handle problems.

Figure 23
Flowchart: an example of troubleshooting using LCo

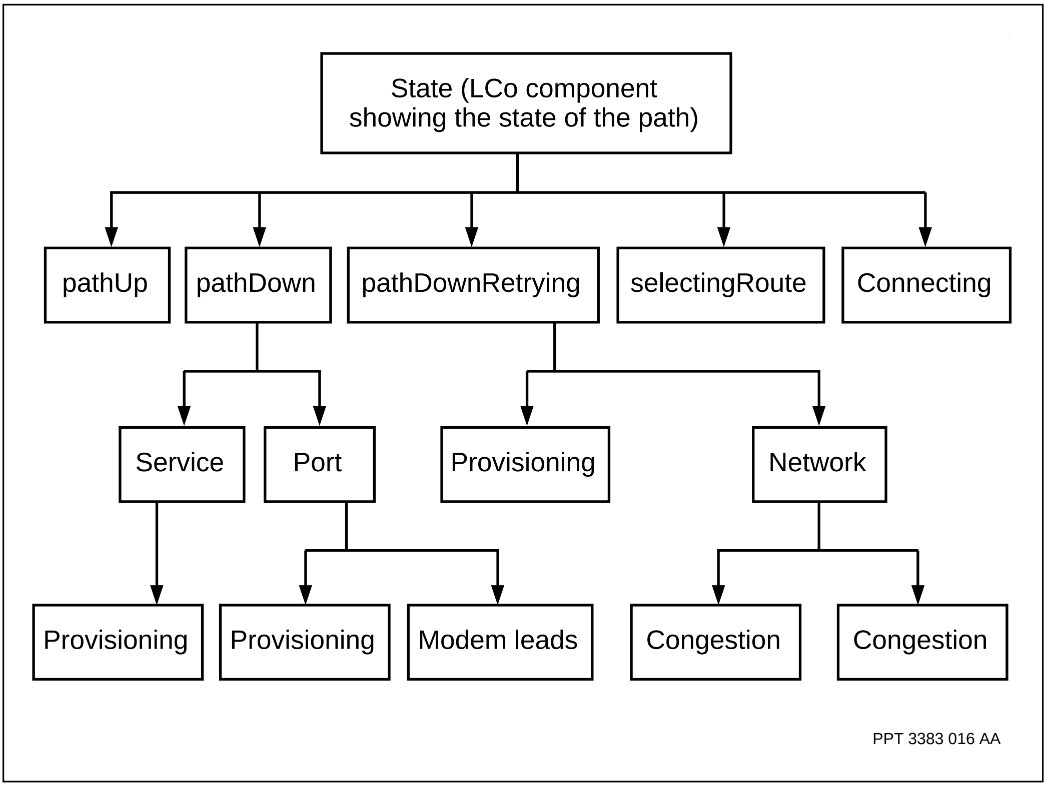


Table 15
Handling problems

Problems that may occur	Probable causes	Corrective measures
Unable to provision voice service.	Error in card provisioning.	Check the card. If an orange light is glowing, check the *logicalProcessor* and *logicalProcessorType* attribute provisioning. Reprovision using correct LP and LPT information. See 241-5701-600 *Passport 7400, 15000, 20000 Configuration Guide* or 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference*.
The connection doesn't come up (see also Note:).	Error in provisioning the service.	Check the service provisioning for both ends of the connection and the serviceFailureReason operational attribute. See "End-to-end negotiation provisioning guidelines and considerations" (page 41), "Interworking provisioning considerations" (page 46), and "Configuration checklists" (page 49) for more information. Passport trunk provisioning information must be identical for both ends and consistent with the PLC provisioning of the connection. Reprovision with correct information.
	Specified bandwidth is not available.	Check the available bandwidth of Passport trunks in the path. Reprovision using less bandwidth if it is not needed or re-engineer the network to make bandwidth available.
(Sheet 1 of 4)		

Table 15 (continued)
Handling problems

Problems that may occur	Probable causes	Corrective measures
The connection doesn’t come up (continued)	If *manualPath* has been used, one of the nodes or Passport trunks used may have failed.	Check nodes/Passport trunks for failure. Reprovision using a path that does not include failed nodes or Passport trunks.
	The port is disabled. The port is in an alarmed state.	Check the operational attributes of the port. Refer to the DS1 and E1 components in 241-5701-060 *Passport 7400, 15000, 20000 Components*. Check for any outstanding alarms. Refer to 241-5701-500 *Passport 6400, 7400, 15000, 20000 Alarms*.
Change in provisioned rate or value of one of the following attributes: • *minVoiceG711G726Rate* • *minModemFaxG711G726Rate* • *silenceSuppression* • *faxIdleSuppressionG711-G726* • *tandemPassThrough* • *dtmfRegeneration* • *v17EncodedAsG711G726*	End-to-end negotiation modified the provisioned rate or value.	Check the actual rate or value used during end-to-end negotiation for this connection. Use the display command to view the operational attributes under the *VoiceService Framer* component’s *Negotiated* group. Refer to “End-to-end negotiation provisioning guidelines and considerations” (page 41) and “Interworking provisioning considerations” (page 46) for more information.
(Sheet 2 of 4)		

Table 15 (continued)
Handling problems

Problems that may occur	Probable causes	Corrective measures
The PLC is up and running, but no data is being sent.	DCE–DTE is not provisioned properly on the subscriber's end.	Check the subscriber's-end DCE–DTE provisioning.
	User's-end terminal may be experiencing problems.	Check the terminal. Take appropriate action to rectify the problem.
	Access line to Passport 7400 may not be transmitting data.	Check the access line. Rectify any problems encountered.
Connection goes down and does not reset.	Under extreme circumstances (For example, no suitable Passport trunk is available), a path may take 1–2 minutes to reroute.	Wait 1–2 minutes and check to see if rerouting has occurred.
	If the security option is being used, no sufficiently secure Passport trunk may be available.	Check Passport trunk provisioning. Take appropriate action to see that a secure Passport trunk is made available. Refer to 241-5701-420 *Passport 7400, 15000, 20000 Trunking Guide*, if necessary.
Unexpected data loss.	Network clock synchronization is not setup properly.	Check that all of the Passport trunks in the path have properly provisioned network clock synchronization.
	Congestion.	Check bandwidth utilization. Take steps to reduce congestion if the problem continues.
	Poor Passport trunk-error performance.	See document 241-5701-420 *Passport 7400, 15000, 20000 Trunking Guide*.
(Sheet 3 of 4)		

Table 15 (continued)
Handling problems

Problems that may occur	Probable causes	Corrective measures
Network clock synchronization remains in *coarseAquisition* for more than one minute.	Improper provisioning of the master.	Network clock synchronization may be provisioned for line where module is the proper value. Check and reprovision with correct value.
	Improper provisioning of references.	Carefully verify provisioning of references.
		As a last resort, quickly lock and unlock the port to which network clock synchronization is attempting to sync (where it is in coarse acquisition).
Note: If a problem has occurred before with this connection, check the *lastTeardownReason* attribute under the *Plc* component.		
(Sheet 4 of 4)		

Checking service provisioning

Use the following procedure to check service provisioning.

Procedure steps

1 To obtain information about the settings of the provisioned view of the *Plc* component from the provisioning mode, type:

`d -p vs/<voice_service_number> Plc`

2 Print this information, if possible, as you will be comparing it to other provisioning data.

3 To obtain information about the settings of the current view of the Trunk Path Administrator component, from the provisioning mode, type:

`d trunk/<trunk_number> Pa`

4 Use the information in 241-5701-060 *Passport 7400, 15000, 20000 Components* to aid you in determining how the attributes should relate.

Variable definitions

Variable	Definition
<trunk_number>	The instance of Passport trunk that you used in the view.
<voice_service_number>	The instance of the *Vs* component you want to view.

Checking bandwidth

Use the following procedure to obtain information about the bandwidth used and the bandwidth available.

Procedure steps

1 Display bandwidth information from Trunk Path Administrator.

`d trk/<trunk_number> pa`

2 Determine if the amount of bandwidth requested is greater than the amount available in the network. To display the information, type:

`d vs/<voice_service_number> ls reasonForNoRoute`

There may be lack of bandwidth if the response is

`plcAttributesNotMet`

Variable definitions

Variable	Definition
<trunk_number>	The instance of Passport trunk that you used in the view.
<voice_service_number>	The instance of the *Vs* component you want to view.

Troubleshooting example

In the case given here, the operator has provisioned a route that has failed to come up. The operator looks at the Routing RouteSelector component's *reasonForNoRoute* attribute to determine the reason.

d rtg rs

```
Rtg Rs
selectedRouteDescription =
routeCostMetric          =  0
routeDelayMetric         = 0 ms
reasonForNoRoute        = unknownRemoteNodeName<<<====
routeSelectionAttributes = fromOperator
sourceId                 = 1157
remoteName               = /NOdeR2b<<<====
setupPriority            = 3
requiredTxBandwidth      = 2084 bit/s
requiredRxBandwidth      = 2084 bit/s
maximumTransmissionUnit = 0
security                 = 4
trafficType              = data
permittedTrunkTypes      = terrestrial satellite tt1
tt2 tt3 ~tt4 ~tt5 ~tt6
customerParameter        = 4
minimizationCriterion    = cost
maximumAcceptableCost    = 1280
maximumAcceptableDelay   = 100000 ms
routeStatistics          = statisticsTable

                     setupPriority|01234
-------------+ ---- ---- --- ---- ----
              routesRequested|      150 01135511511400
             routesSelectedAtBp_0|   0 0      0      0 0
             routesSelectedAtBp_1|   1 0      0      0 0
```

```
        routesSelectedAtBp_2|  80 0   2965      0 0
        routesSelectedAtBp_3|   0 0  24772  10087 0
        routesSelectedAtBp_4|  46 0 73777 111989  0

ok
```

The *reasonForNoRoute* attribute indicates that the *remoteName* is unknown. In this case the *remoteName* used is also displayed. For other types of problems, the operator may need to display another attribute to show the required information.

In this case, the problem can be identified as a typographic error in the *remoteName* provisioning. The `/` mark at the beginning of the node name is incorrect. To correct the problem, the operator would reprovision the *remoteName* with the correct name of the other end of the connection.

Appendix A
Data stream refresher

A voice connection between Passport nodes is a stream of frames. Each frame is composed of numerous time slots. One time slot is one voice channel, or in other words, one voice service. Each voice channel is composed of eight data bits per frame, or 8000 x 8 bits per second, without compression.

In synchronous systems, such as Voice Transport, each voice channel is allocated to a permanent time slot in each frame of the data stream. In other words, the timing between adjacent Passport 7400 nodes is synchronized. Voice channel 1 (for example) is always transferred in time slot 1 (for example) of successive frames. Synchronizing the clocks prevents data frame slips that cause data to be lost or duplicated.

In North America, the data stream is a series of frames composed of 24 time slots. This data stream format is called the DS1 signal, depicted in figure "DS1 signal data frame" (page 174). In Europe, the data stream is a series of frames composed of 32 time slots. This data stream format is called the E1 signal, depicted in figure "E1 signal data frame" (page 175).

Figure 24
DS1 signal data frame

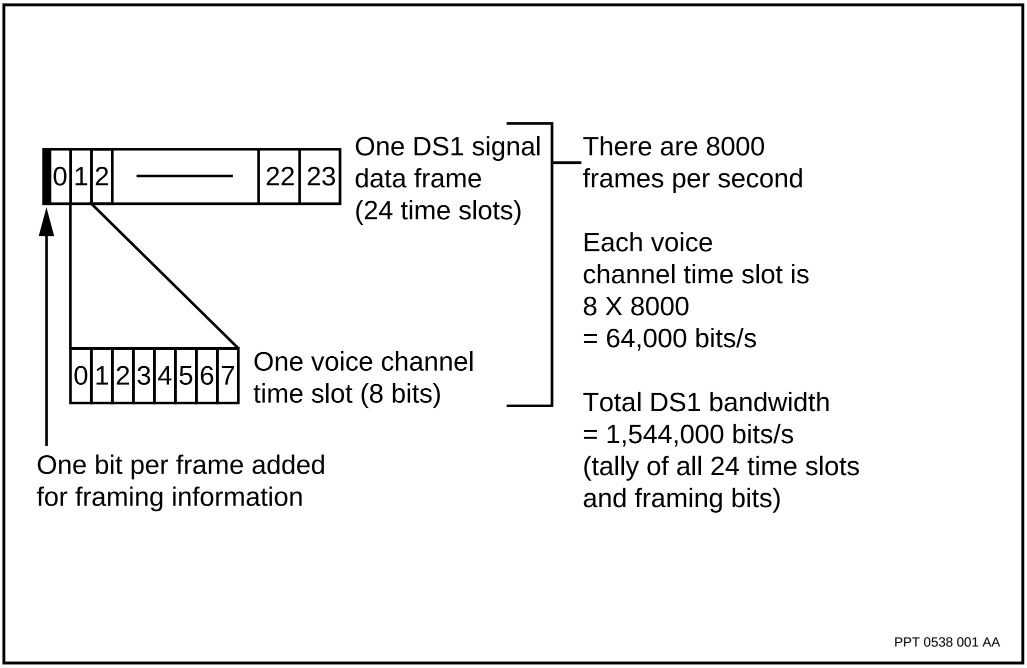
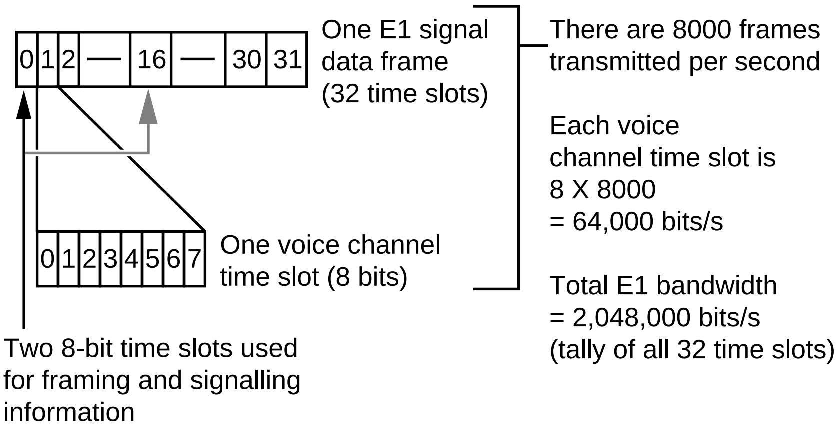


Figure 25
E1 signal data frame



Note: This example shows an E1 channel associated signalling (CAS) signal format where time slot 16 is used for signalling information.
Timeslot 16 can be used for common channel signalling (CCS) data.

Note: For the J2MV function processor, slot 0 is used for signalling information. Time slot 16 is unused.

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Appendix B
Signalling refresher

There are two types of signalling formats:

- "Common channel signalling (CCS)" (page 177)
- "Channel associated signalling (CAS)" (page 178)

Both types of signalling formats are similar to each other except for their type of channel control. CAS uses 1, 2, or 4 bits, called ABCD bits, to perform all or part of channel control. CCS uses High-Level Data Link Control (HDLC) frames for its channel control. Typically a trunk (a full DS1 or E1 pipe) is either all CAS or CCS, though it is possible to split a trunk to support both signalling formats.

See the following sections for information about each interface's framing format:

- "DS1 framing format" (page 178)
- "E1 framing format" (page 179)
- "TTC2M framing format" (page 181)

Common channel signalling (CCS)

A typical CCS setup would require one CCS link (often called the D-channel) per trunk group (a full DS1 or E1 pipe). The D-channel is carried by the Bit Transparent Data Service (BTDS) and is not interpreted. CCS messages can be handled if the *Dpnss1* or *Mcdn* component is provisioned.

Channel associated signalling (CAS)

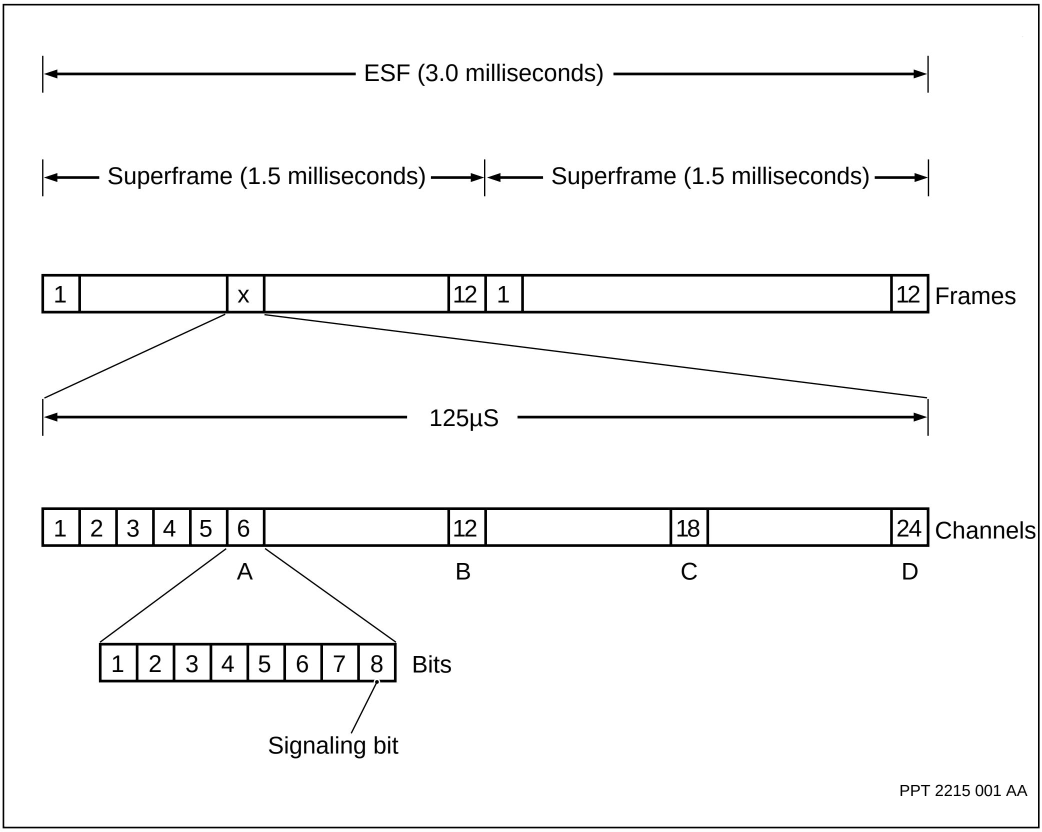
CAS is associated with one of the following three modes: DS1 Super Frame (SF or d4) mode (the provisionable attribute *linetype* is set to d4cas); DS1 Extended Super Frame (ESF) mode (the provisionable attribute *linetype* is set to esfcas); and E1 CAS mode (the provisionable attribute *linetype* is set to cas). The main difference between DS1 and the E1 implementation is that DS1 employs a bit robbing scheme (for example, one bit is robbed from every timeslot every six frames), and E1 allocates one timeslot (time slot 16) to carry all the signalling bits.

DS1 framing format

A DS1 frame is composed of 24 8-bit time slots and one framing bit. Therefore one DS1 frame consists of 193 bits. Each frame is 125 microseconds long which translates to a raw data rate of 1.544Mbps (refer to appendix “Data stream refresher” (page 173) for more information). The main difference between SF and ESF is that SF has 12 frames per super frame while ESF has 24 frames per super frame. Refer to figure “DS1 super frame” (page 179). This difference in the number of frames per super frame ensures that SF can support only two signalling bits (for example, the A-bit is in the 6th frame and the B-bit is in the 12th frame) while ESF can support all four signalling bits (i.e. ABCD).

The valid signalling bits can be either the full range or a subset of the range supported by the framing type. This means that both SF and ESF can support A or AB bit signalling, but only ESF can support ABCD bit signalling. The only valid signalling bit combinations are A, AB, or ABCD.

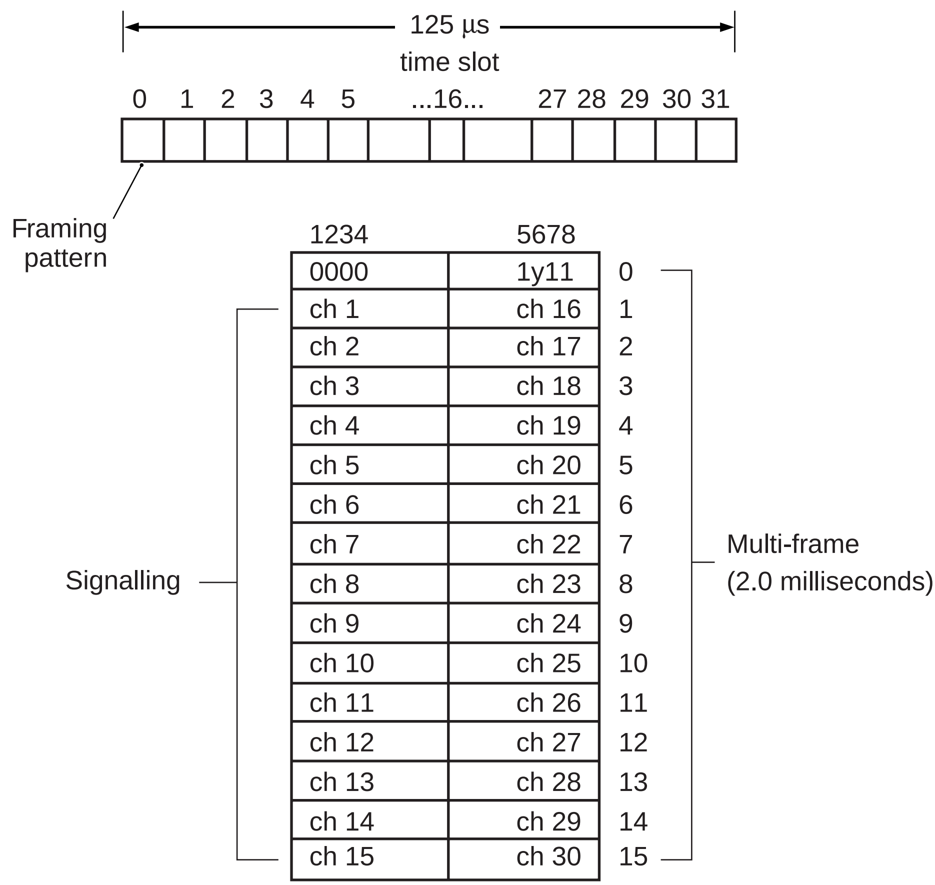
Figure 26
DS1 super frame



E1 framing format

Like DS1, each E1 frame is 125 microseconds long. Unlike DS1, E1 has 32 8-bit time slots per frame. Refer to figure “E1 multi-frame” (page 180). Time slot 0 is reserved for carrying framing information and time slot 16 is reserved for carrying the ABCD signalling bits (refer to appendix “Data stream refresher” (page 173) for more information). The only valid signalling bit combinations are A, AB, or ABCD.

Figure 27
E1 multi-frame



Notes:
1) '0000' in frame 0 is the multi-frame pattern.
2) 'y' in frame 0 is the remote end alarm indication; 0 is off, 1 is on.
3) ch1 to 15 <-> timeslot 1 to 15 & ch 16 to 30 <-> timeslot 17 to 31.

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TTC2M framing format

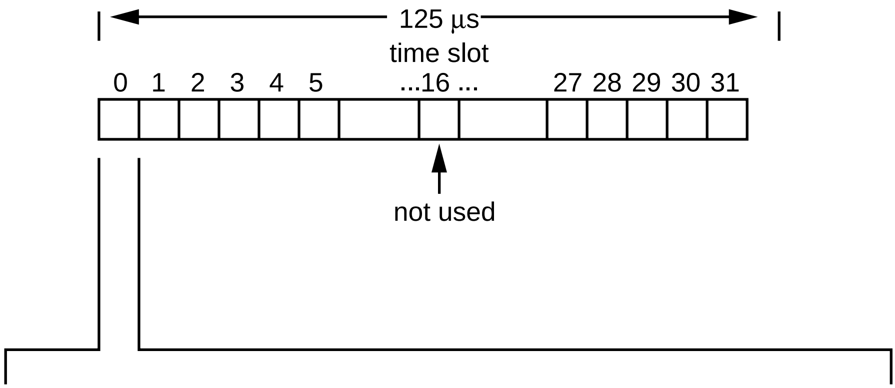
TTC2M is the framing format used in Japan. TTC2M is a communication protocol which is based on the JJ-20.10, JJ-20.11, and JJ-20.12 standards in Japan.

Each TTC2M frame is 125 microseconds long. Each TTC2M frame has thirty-two 8-bit timeslots which are numbered from 0 to 31. Refer to figure “TTC2M multi-frame” (page 182). Unlike E1 framing, timeslot 0 is used for both framing information and A-bit signalling. Timeslot 16 is not defined.

> ***Note:*** Unlike DS1 and E1 framing, TTC2M does not have BCD signalling bits.

For information on which FPs support TTC2 FM, refer to 241-5701-615 *Passport 7400, 15000, 20000 FP Configuration Reference*.

Figure 28
TTC2M multi-frame



Frame	1	2	3	4	5	6	7	8
1	F	0	S	X	X	X	X	X
2	F	1	S	A1	A2	A3	A4	A5
3	F	1	S	A6	A7	A8	A9	A10
4	F	1	S	A11	A12	A13	A14	A15
5	F	1	S	A16	A17	A18	A19	A20
6	F	1	S	A21	A22	A23	A24	A25
7	F	1	S	A26	A27	A28	A29	A30
8	F	1	S	X	X	X	X	X

Note: Timeslot 0 carries eight bites, with the following designation:

- X = not used

  F = CMI violation to mark start of frame
- Bit 2 is used for multiframe alignment
- Bit 3 (S bit) is used for remote alarm indication, 0 is an error, 1 is normal
- Bit 4 through to Bit 8 carries A-bit signaling, one per channel

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Appendix C
Voice Transport function processor migration information

When you migrate from existing MVP-E FPs to the 4-port MVP-E FPs, you must consider the differences in features supported on each of the FP types.

The table, “Feature support differences between 4-port MVP-E FP and other voice and MVP-E FPs” (page 183) compares the differences in the features supported on the 4-port MVP-E FP and those supported on the MVP-E FPs.

Refer to the following sections, for specific migration procedures for each of the features that are not supported by 4-port MVP-E FPs:

- “Migrating from 1-port MVP-E FPs to 4-port MVP-E FPs” (page 184)

  Note: In addition, consider all new features that require additional provisioning during the migration to the 4-port MVP-E FP.

Table 16
Feature support differences between 4-port MVP-E FP and other voice and MVP-E FPs

	1-port MVP-E
Unsupported features in 4-port MVP-E	• V.17 fax call encoded using G.711 and/or G726 while other fax calls encoded using fax relay • payload channel link loopback

Note: For the 4-port E1 MVP-E FP, channel test on timeslot 16 is not supported.

Migrating from 1-port MVP-E FPs to 4-port MVP-E FPs

Refer to the following specific migration procedures for each of the features that are not supported by 4-port MVP-E FPs.

- "Handling of V.17 fax calls" (page 184)
- "Payload channel link loopback" (page 184)

Handling of V.17 fax calls

4-port MVP-E FPs do not support handling V.17 fax calls differently than V.29 fax calls. On four-port MVP-E FPs, V.29 and V.17 fax calls are always handled in the same way; both are encoded using fax relay or both are encoded according to ITU-T G.711and/or G.726. When you migrate from Voice FPs to 4-port MVP-E FPs, it is not necessary to disable this feature. The feature is automatically disabled through end-to-end negotiation.

Payload channel link loopback

The 4-port MVP-E FP does not support payload channel link loopback. This FP supports external loopback on a per-port basis.

Passport 7400

Voice Transport Guide

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