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Nortel Multiservice Switch 7400

Operations: Voice Networking

NN10600-755

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Contents

What's new

The following feature was added to this document:

- Nortel Multiservice Switch 7400 4pDS1 and 4pE1 MVPe multiprotocol on different ports (page 7)

Attention: To ensure that you are using the most current version of an NTP, check the NTP list in NN10600-000 *Nortel Multiservice Switch 7400/15000/20000 What's New*.

Nortel Multiservice Switch 7400 4pDS1 and 4pE1 MVPe multiprotocol on different ports

The following section was updated for this feature:

- Installation (page 72)

Voice Networking configuration

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

Prerequisites to Voice Networking configuration

- Use the task flows and procedures in NN10600-270 *Nortel Multiservice Switch 7400/15000/20000 Software Installation* to install Voice Networking and related software.
- Install and configure path-oriented routing system (PORS) and the dynamic packet routing system (DPRS). PORS and DPRS enable the Voice Networking service connections between nodes and route the audio traffic. Ensure that your version of PORS and DPRS are compatible with the version of Voice Networking software you plan to deploy.

  For information on configuring PORS, see NN10600-435 *Nortel Multiservice Switch 7400/15000/20000 Operations: Path-Oriented Routing System* and NN10600-420 *Nortel Multiservice Switch 7400/15000/20000 Operations: Trunking*. For information on configuring DPRS and the call server, see NN10600-425 *Nortel Multiservice Switch 7400/15000/20000 Operations: Dynamic Packet Routing System* and NN10600-450 *Nortel Multiservice Switch 7400: Operations: DPN-100 Interworking*, and NN10600-405 *Nortel Multiservice Switch 7400/15000/20000 Operations: Call Server*, respectively.
- To use PORS with DNA-based routing, you must configure the *Routing DpnAddressPlan* components and the *ModuleData VirtualCircuitSystem* components. See Voice Networking based on DNAs and DPRS (page 109).

  For more information about configuring the *Routing DpnAddressPlan* components, see NN10600-450 *Nortel Multiservice Switch 7400: Operations: DPN-100 Interworking*. For more information about configuring the *ModuleData VirtualCircuitSystem* components, see NN10600-901 *Nortel Multiservice Switch 7400/15000/20000 Frame Relay Configuration Management*.

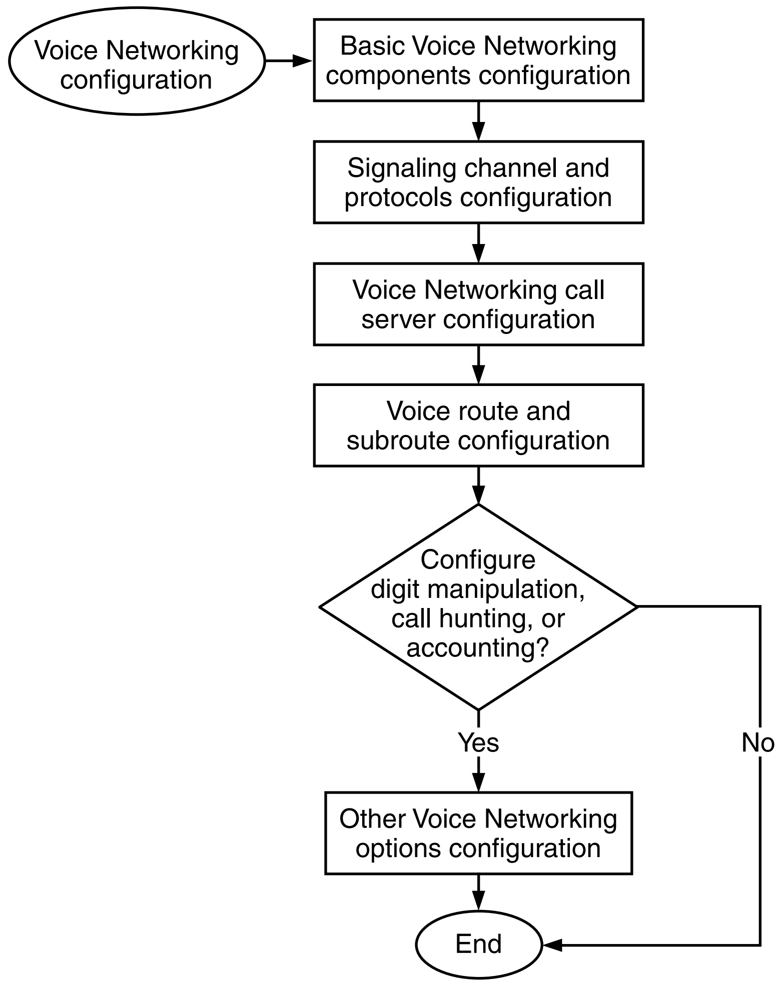
- Install network clock synchronization (NCS). NCS is described in NN10600-550 *Nortel Multiservice Switch 7400/15000/20000 Common Configuration Procedures*.
- Install and configure hunt group software on each node that hosts a hunt group server. For information on installing and configuring hunt group software, see NN10600-415 *Nortel Multiservice Switch 7400/15000/20000 Operations: Hunt Group Server*.
- Install the required hardware. For more information, see Hardware requirements for Voice Networking (page 68).
- To simplify the configuration process, relate like-numbered component and subcomponent instances.

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

Voice Networking configuration tasks

This task flow shows you the sequence of tasks you perform to configure Voice Networking. To link to any task, go to Voice Networking task navigation (page 10).

Voice Networking configuration tasks



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Voice Networking task navigation

- Basic Voice Networking components configuration (page 11)
- Signaling channel and protocols configuration (page 18)
- Voice Networking call server configuration (page 37)
- Voice route and subroute configuration (page 44)
- Other Voice Networking options configuration (page 57)

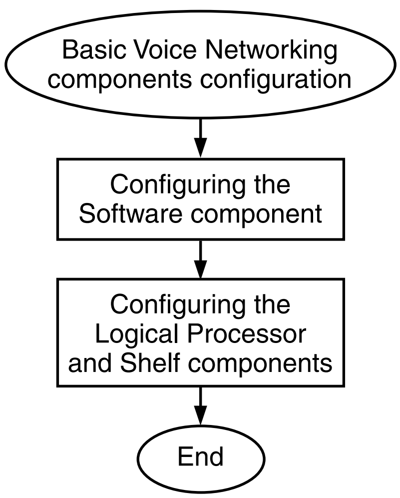
Basic Voice Networking components configuration

Configure the basic components of the Voice Networking service.

Basic Voice Networking components configuration procedures

This task flow shows you the sequence of procedures you perform to configure the basic Voice Networking components. To link to any procedure, go to Basic Voice Networking components procedure navigation (page 11).

Basic voice networking components configuration procedures



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Basic Voice Networking components procedure navigation

- Configuring the Software component (page 12)
- Configuring the LogicalProcessor and Shelf components (page 14)

Configuring the Software component

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

Prerequisites

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

Procedure steps

Step	Action
1	Add voice networking call server software to the *LogicalProcessorType* component instance for the control processor (CP). **`set sw lpt/CP featureList vncsCallServer`**
2	Add an *Lpt* component instance for the Voice Networking service. **`add sw lpt/<name>`**
3	Set the *featureList* attribute by specifying the Voice Networking software package, defining the signaling protocol, and if necessary, adding MVP-E function processor (FP) specific software features. **`set sw lpt/<name> featureList voiceNetworking <protocol> <optional_feature>`**
4	Repeat this procedure to configure Voice Networking software on each node in the network that connects to a PBX.

--End--

Variable definitions

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

Procedure job aid

Software component hierarchy

```
Root
 └ Software
    └ LogicalProcessorType/<name>
       featureList
```

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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 Nortel Multiservice Switch node.

Procedure steps

Step	Action
1	Add logical processors. **`add lp/<lp_number>`**
2	Link the LPs to the configured Voice Networking software. **`set lp/<lp_number> logicalProcessorType sw lpt/<name>`**
3	For each FP, define its slot number. **`add shelf card/<card_number>`**
4	For each FP, define its card type. **`set shelf card/<card_number> cardType <FP>`**
5	Link each LP to an FP. **`set lp/<lp_number> mainCard shelf card/<card_number>`**
6	Perform a semantic check of your changes. **`check Prov`**
7	Add a port. **`add lp/<lp_number> <port>/<port_number>`**
8	Specify the type of framing format to use for the port. **`set lp/<lp_number> <port>/<port_number> linetype <framing_format>`**
9	Specify the clocking source to use for each port. **`set lp/<lp_number> <port>/<port_number> clockingSource <value>`**
10	Add channels and timeslots to each port. The type of signaling protocol and port determines how to provision channels and timeslots. For both CAS and CCS protocols interfacing to an E1 port, define 30 channels to process calls and one to carry signaling information. For CCS protocols interfacing to a

DS1 port, define 23 channels for bearer services and one to carry signaling information. For CAS protocols interfacing to a DS1 port, define 24 channels to carry traffic.

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

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

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

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

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

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

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

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

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

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

15 Repeat this procedure to configure logical and function processors on each node in the network that connects to a PBX.

--End--

Variable definitions

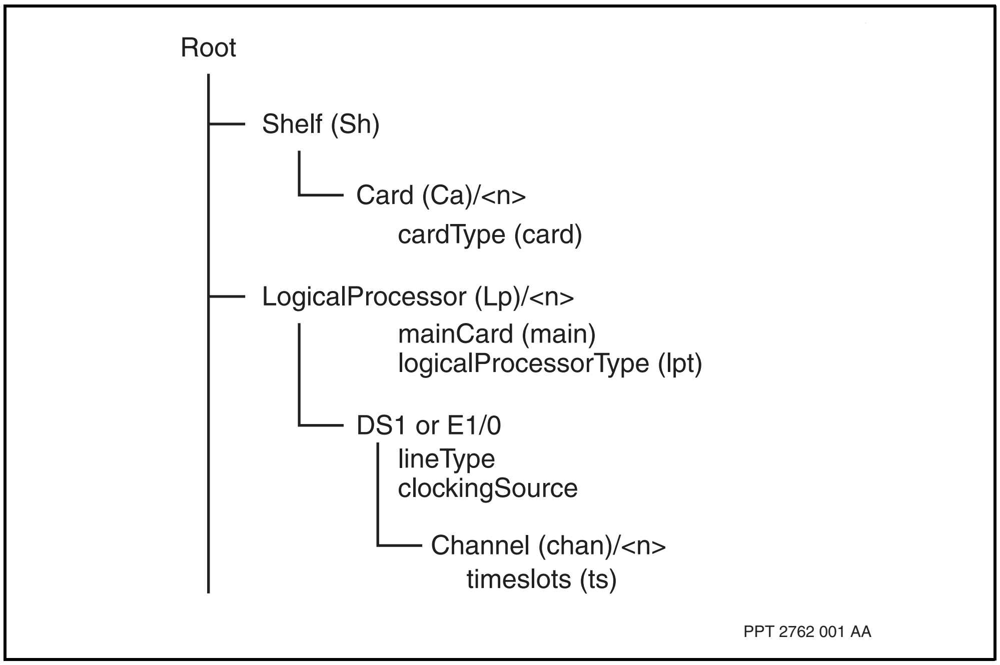
Variable	Value
<card number>	is instance number of the card.
<FP>	is the card type. To determine the value for the FP you are configuring, refer to NN10600-551 *Nortel Multiservice Switch 7400/15000/20000 FP Configuration Reference*. Ensure that the FP you specify supports the protocol and features you configured under the *Software LogicalProcessorTyp*e component.
<framing_format>	is 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>	is the instance number of the LP.
<name>	is the title given to the *Software LogicalProcessorType* component instance.
<port>	is either e1 or ds1. The FP you specify under the *Card* component determines the type of port you provision. To determine the value for the FP you are configuring, refer to NN10600-551 *Nortel Multiservice Switch 7400/15000/20000 FP Configuration Reference*. When you add an *E1* or *Ds1* component, the system automatically creates the component instance Channel/0.
<port_number>	is instance number of the port.
<signaling_channel>	is 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 (for example, ETSI QSIG) define channels 1 to 30 for bearer services and assign timeslot 16 to channel 31.

(1 of 2)

Variable	Value
<timeslot_number>	is the timeslot number.
<value>	is Local, line or module.
	If you configured the *NetworkSynchronization* component, the *clockingSource* attribute defaults to module, otherwise it defaults to line. A value of module specifies that the E1 or DS1 port synchronizes to the Stratum-3 clock on the active CP.
	(2 of 2)

Procedure job aid

LogicalProcessor and Shelf component hierarchy



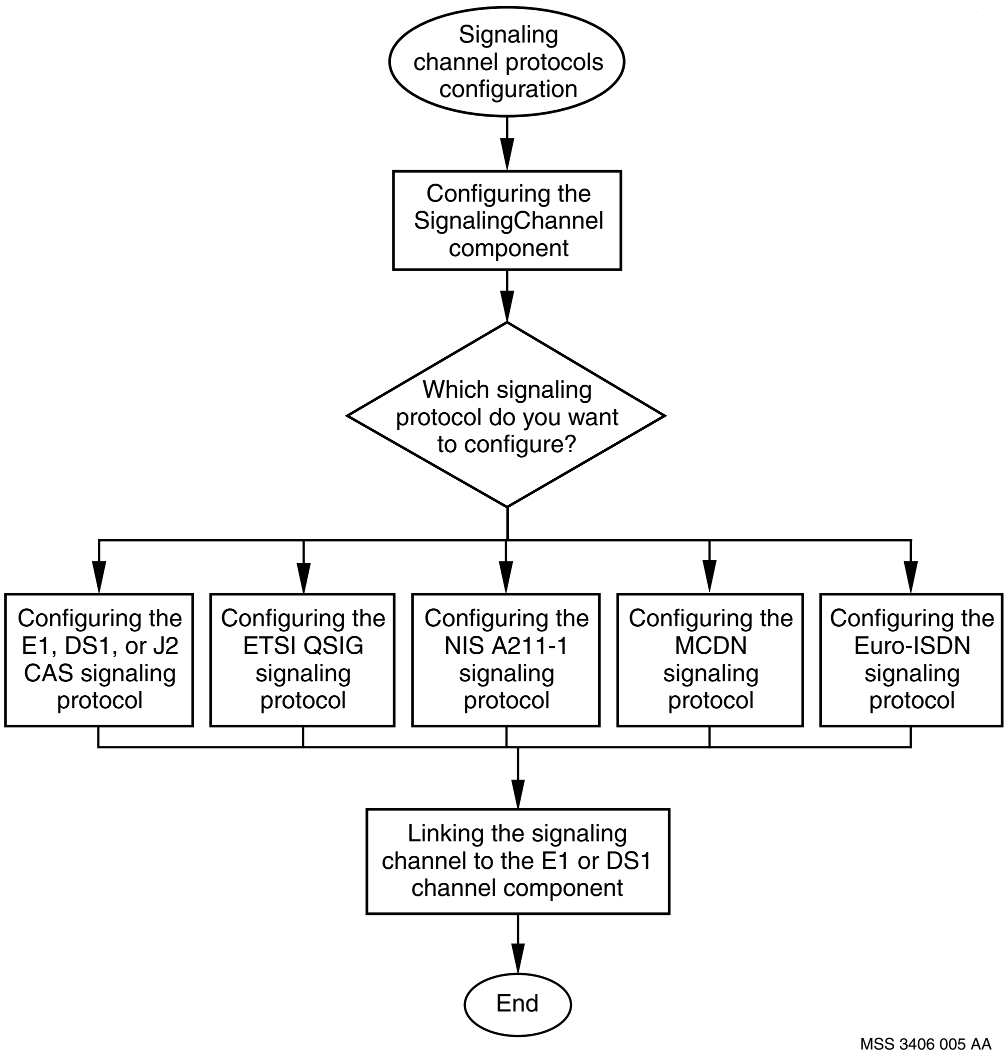
Signaling channel and protocols configuration

Configure the signaling channels and their associated protocols.

Signaling channel and protocols configuration procedures

This task flow shows you the sequence of procedures you perform to configure the signaling channel and protocols. To link to a procedure, go to Signaling channels and protocols configuration navigation (page 19).

Signaling channels and protocols configuration procedures



Signaling channels and protocols configuration navigation

- Configuring the SignalingChannel component (page 20)
- Configuring the E1, DS1, or J2 CAS signaling protocol (page 23)
- Configuring the ETSI QSIG signaling protocol (page 26)
- Configuring the NIS A211-1 signaling protocol (page 28)
- Configuring the MCDN signaling protocol (page 30)
- Configuring the Euro-ISDN signaling protocol (page 33)
- Linking the signaling channel to the DS1 or E1 channel component (page 36)

Configuring the SignalingChannel component

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

Prerequisites

- For more reference and conceptual information, refer to Voice Networking fundamentals (page 65) and Signaling protocols fundamentals (page 78).
- To automatically enable CCS-to-CAS protocol gateways, add the MCDN, etsiQsig or nisSig and casSig protocol software features on the appropriate Nortel Multiservice Switch nodes. Make sure to add the protocol that matches the protocol running on the connected PBX and configure the *SigChan* component on each node accordingly.
- To automatically enable interworking between NIS A211-1 and MCDN, add the nisSig and mcdnSig protocol software features on the appropriate nodes. Make sure to add the protocol that matches the protocol running on the connected PBX and configure the *SigChan* component on each node accordingly.

Procedure steps

Step	Action
1	Add a a signaling channel. **`add sigchan/<signaling_channel>`**
2	Specify the signaling protocol to use. **`add sigchan/<signaling_channel> <protocol>`**
3	Optionally, override or replace missing or possibly corrupted numbering plan indicator (NPI) information supplied by the calling PBX. **`set sigchan/<signaling_channel> forceNpiTon <decision>`**
4	Optionally, override or replace missing or possibly corrupted type of number (TON) information supplied by the calling PBX. **`set sigchan/<signaling_channel> defaultNpiTon <value>`**
5	Optionally, add the *InternalCauseMap* component. **`add sigchan/<signaling_channel> InternalCauseMap`**
6	Optionally, verify that the default settings of the *InternalCauseMap* component's provisionable attributes meet network requirements, and make changes as required. **`display sigchan/<signaling_channel> InternalCauseMap`**
7	Repeat this procedure to configure the signaling channel on each node that connects to a PBX.

--End--

Variable definitions

Variable	Value
<decision>	is yes or no. No is the default. If set to yes, Voice Networking ignores NPI and TON information sent by the calling PBX and uses the values configured under the *defaultNpiTon* attribute. Even when the *forceNpiTon* attribute is set to no, Voice Networking can use the *defaultNpiTon* attribute's default value, casUnknown, when NPI and TON information in the call setup message received from the calling PBX is missing or cannot be interpreted.
<port>	is E1 or Ds1.
<protocol>	is E1Cas, Ds1Cas, J2Cas, EtsiQsig, NisSig, McdnSig, or EuroIsdn. The protocol you specify corresponds to the protocol software you configured under the *Software* component and matches the signaling format—CAS or CCS—you specified under the *E1* or *DS1* port component.
<signaling_channel>	is the channel you defined to carry timeslot 16 signaling information for CAS or CCS protocols interfacing to an E1 port or 0 for CAS or CCS protocols interfacing to a Ds1 port.
<value>	is one or more of the following: . • casUnknown (default) • unknown • international • national • subscriber • p0 up to p7

Procedure job aid

SignalingChannel component hierarchy

- **Root**
  - **SignallingChannel/1-153**
    - subroutes
    - octothorpeEod
    - force/NpiTon
    - defaultNpiTon
    - **InternalCauseMap**
    - **E1Cas or Ds1Cas or J2Cas**
      - **Framer**
        - interfaceName
    - **EtsiQsig**
      - **Framer**
        - interfaceName
    - **NisSig**
      - **Framer**
        - interfaceName
    - **McdnSig**
      - **Framer**
        - interfaceName
    - **EuroIsdn**
      - **Framer**
        - interfaceName

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internal cause codes component hierarchy

- **SignallingChannel**
  - **InternalCauseMap**
    - egressLinkOutOfServCause
    - chanOrCircNotAvailCause
    - tempFailureCause
    - switchCongestCause
    - reqChanOrCircNotAvailCause
    - resourceUnavailCause
    - servNotAllowedCause
    - noSuchChannelCause
    - incompatDestCause

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Configuring the E1, DS1, or J2 CAS signaling protocol

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

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

Prerequisites

- See NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference* for attribute descriptions and values related to configuring the E1, DS1, or J2 CAS signaling protocol (see E1Cas, Ds1Cas, or J2Cas signaling protocol component hierarchy (page 25)).

Procedure steps

Step	Action
1	Select the appropriate tone table to use according to country. **`set sigchan/<signaling_channel> <signaling_protocol> tonetableselection <country_number>`**
2	Verify that the default settings of the AbcdProv group of provisionable attributes meet network requirements. **`display -p sigchan/<signaling_channel> <signaling_protocol> abcdprov`**
3	Verify that the default settings of the TimerProv group of provisionable attributes meet network requirements. **`display -p sigchan/<signaling_channel> <signaling_protocol> timerprov`**
4	Add a *DTMF* component. **`add sigchan/<signaling_channel> <signaling_protocol> DTMF`** **`set sigchan/<signaling_channel> <signaling_protocol> DTMF burstTime <value>`**
5	Repeat this procedure for each signaling channel on each node connected to a PBX that uses a CAS protocol.

--End--

Variable definitions

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

Procedure job aid

E1Cas, Ds1Cas, or J2Cas signaling protocol component hierarchy

- **SignallingChannel**
  - **E1Cas or Ds1Cas or J2Cal**
    - toneTableSelection
    - AbcdProv (group)
      - signallingBits
      - idleTx
      - idleRx
      - faultTx
      - faultRx
      - seizeTx
      - seizeRx
      - seizeVoiceTx
      - seizeVoiceRx
      - seizeDataTx
      - seizeDataRx
      - seizeAckTx
      - seizeAckRx
      - winkTx
      - winkRx
      - connectTx
      - connectRx
      - clearBackTx
      - clearBackRx
      - clearFwdTx
      - clearFwdRx
    - TimerProv (group
      - guardTime
      - winkTime
      - winkMinTime
      - winkMaxTime
      - seizeAckDelayTime
      - seizeAckSupvTime
      - outpulseDelayTime
      - endofDialtime
      - disconnectTime
      - disconnectSupvTime
    - **Dtmf**
      - burstTime

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Configuring the ETSI QSIG signaling protocol

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

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

Prerequisites

- See NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference* for attribute descriptions and values related to configuring the ETSI QSIG signaling protocol (see ETSI QSIG signaling protocol component hierarchy (page 27)).

Procedure steps

Step	Action
1	Verify that the default settings of the *EtsiQsig* component's provisionable attributes meet network requirements, and make changes as required. **`display sigchan/<signaling_channel> EtsiQsig`**
2	If necessary, enable the *EtsiQsig* component to detect, segment, and reassemble large-sized ISDN messages. **`set sigchan/<signaling_channel> EtsiQsig msgSegmentation enabled`**
3	If the connected PBX defines channels 1 to 30 for bearer services and the node defines channels 1 to 15 and 17 to 31, map the PBX's channel assignments to those configured on the node. **`set sigchan/<signaling_channel> EtsiQsig e1ChannelNumbers contiguous`**
4	Repeat step 1 to step 3 for each signaling channel on each node connected to a PBX that uses the ETSI QSIG signaling protocol.

--End--

Procedure job aid

ETSI QSIG signaling protocol component hierarchy

SignallingChannel

- **EtsiQsig**
  - side
  - maxNonCallConcurrent
  - overlapSending
  - overlapRecieving
  - Layer2Options (group)
    - t23
    - t200
    - n200
    - t203
    - circuitSwitchedK
  - Layer3Options (group)
    - t309
    - t310
  - EQsigSpecificProv (group)
    - msgSegmentation
    - e1ChannelNumbers

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Configuring the NIS A211-1 signaling protocol

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

Prerequisites

- See NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference* for attribute descriptions and values related to configuring the NIS A211-1 signaling protocol (see NIS A211-1 signaling protocol component hierarchy (page 29)).

Procedure steps

Step	Action
1	Verify that the default settings of the *NisSig* component's provisionable attributes meet network requirements, and make changes as required. **`display sigchan/<signaling_channel> NisSig`**
2	Repeat step 1 for each signaling channel on each node connected to a PBX that uses the NIS A211-1 signaling protocol.

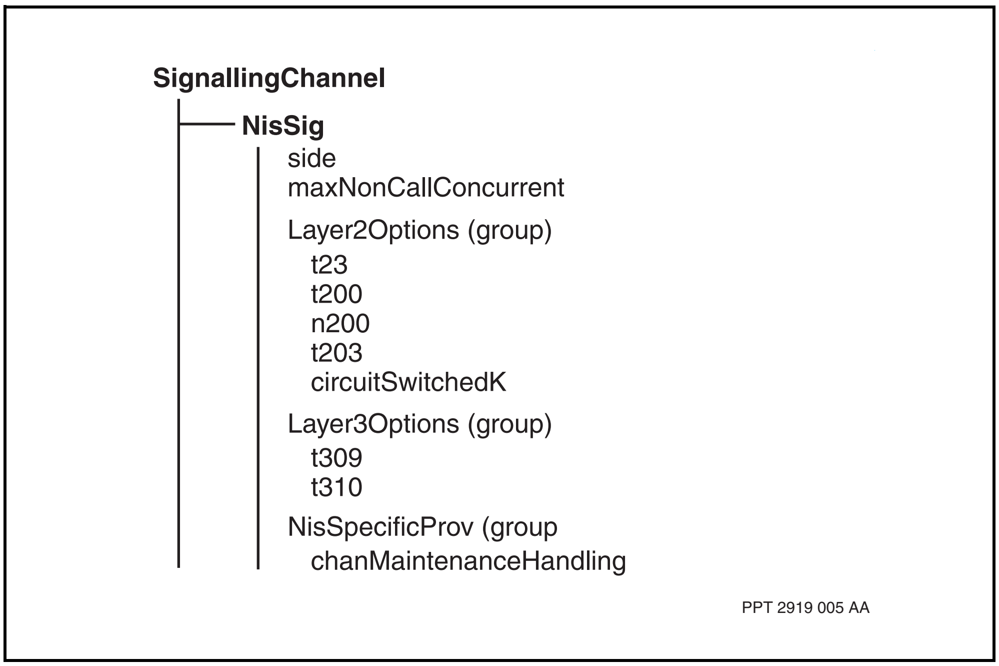
--End--

Variable definitions

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

Procedure job aid

NIS A211-1 signaling protocol component hierarchy



Configuring the MCDN signaling protocol

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

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

Prerequisites

- Review NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference* for attribute descriptions and values related to configuring the MCDN signaling protocol.

Procedure steps

Step	Action
1	Verify that the default settings of the *McdnSig* component's provisionable attributes meet network requirements, and make changes as required. **display -p sigchan/<signaling_channel> McdnSig <provisionable_attribute>**
2	If, in your network, certain nodes must have access restrictions (that is, calls must flow through certain tandem nodes), disable anti-tromboning on the minimum-access node. **set sigchan/<signaling_channel> McdnSig conOrFeaturesSupported ~antiTromboning**
3	Optionally, make changes to the drop back busy service for congestion handling. **set sigchan/<signaling_channel> McdnSig dropBackCongestion <dropback_value>**
4	If required, configure a particular type of network name display service. By default, Voice Networking does not enable MCDN network name display services. **set sigchan/<signaling_channel> McdnSig networkNameDisplay <display_value>**
5	Repeat step 1 to step 4 for each signaling channel on each node connected to a Meridian 1 PBX that uses the MCDN signaling protocol.

--End--

Variable definitions

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

Procedure job aid

MCDN signaling protocol component hierarchy

SignallingChannel

- **McdnSig**
  - side
  - maxNonCallConcurrent
  - overlapSending
  - overlapReceiving
  - chanMaintenanceHandling
  - Layer2Options (group)
    - t23
    - t203
    - t200
    - circuitSwitchedK
    - n200
  - Layer3Options (group)
    - t302
    - t310
    - t304
    - t316
    - t309
  - ConnectionlessFeatures (group)
    - clsFeaturesSupported
  - ConnectionOrientedFeatures (group)
    - dropBackCongestion
    - networkNameDisplay
    - multisiteBusinessGroup
    - conOrFeaturesSupported

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Configuring the Euro-ISDN signaling protocol

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

- OSI Layer 2 and 3 timer settings
- various timer settings for non call-associated sessions (also referred to as virtual calls). Virtual calls provide bearer-independent connection-oriented transport for Euro-ISDN supplementary services.
- to enable or disable support for overlap dialing (in both the sending and receiving directions).
- the particular Euro-ISDN protocol variant to use. T309 for Euro-ISDN is calculated according to the specification as follows: T309: 6 to 12 seconds, according to the formula (N200 + 1) x T200 + 2 seconds

Prerequisites

- Review NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference* for attribute descriptions and values related to configuring the Euro-ISDN signaling protocol (see Euro-ISDN signaling protocol component hierarchy (page 35)).
- For the 4-port E1 MVP-E FP, you must provision the protocol variant identically on all four ports.

Procedure steps

Step	Action
1	Verify that the default settings of the *EuroIsdn* component's provisionable attributes meet network requirements. **`display SigChan/<signaling_channel> EuroIsdn`**
2	Define the protocol variant. **`set sigchan/<signaling_channel> euroisdn variant <type>`**
3	If you specified either *austria* or *germany* under the variant attribute, set the side attribute to user (the default value network applies if you set the variant attribute to etsiGeneric). **`set sigchan/<signaling_channel> euroisdn side user`**
4	If necessary, change the default settings for tracking the amount of time involved in setting up and monitoring non call-associated sessions or virtual calls. **`set sigchan/<signaling_channel> euroisdn connectservicetimer <seconds>`** **`set sigchan/<signaling_channel> euroisdn responseservicetimer <seconds>`**


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

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

--End--

Variable definitions

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

Procedure job aid

Euro-ISDN signaling protocol component hierarchy

SignallingChannel

- **EuroIsdn**
  - side
  - maxNonCallConcurrent
  - overlapSending
  - overlapReceiving
  - Options (group)
    - variant
    - connectServiceTimer
    - responseServiceTimer
    - lifetimeServerTimer
  - Layer2Options (group)
    - t23
    - t200
    - n200
    - t203
    - circuitSwitchedK
  - Layer3Options (group)
    - t310

PPT 2919 007 AA

Linking the signaling channel to the DS1 or E1 channel component

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

Procedure steps

Step	Action
1	Link each signaling channel to the *E1* or *DS1* Channel component instance that carries signaling information for the particular signaling protocol. **set sigchan/<signaling_channel> <protocol> framer interfacename lp/<lp_number> <port>/<port_number> chan/<signaling_channel>**

--End--

Variable definitions

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

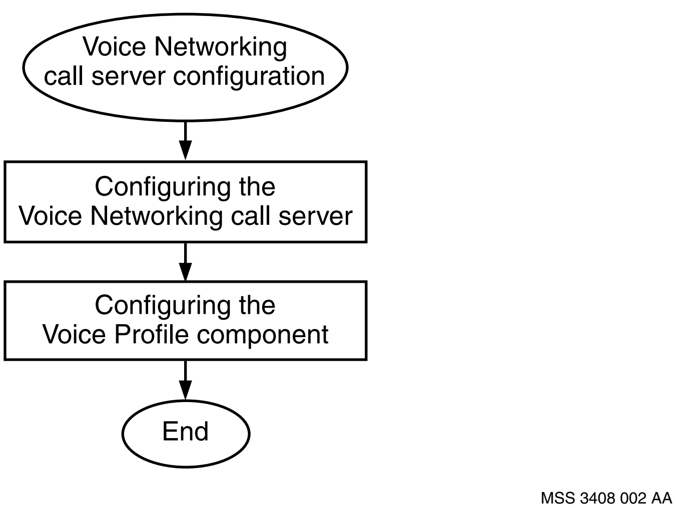
Voice Networking call server configuration

Configure the Voice Networking call server and its *VoiceProfile* component.

Voice Networking call server configuration procedures

This task flows shows the sequence of procedures you perform to configure the Voice Networking call server and the *VoiceProfile* component. To link to a procedure, go to Voice Networking call server configuration procedure navigation (page 37).

Voice Networking call server configuration procedures



Voice Networking call server configuration procedure navigation

- Configuring the Voice Networking call server (page 38)
- Configuring the VoiceProfile component (page 42)

Configuring the Voice Networking call server

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

Prerequisites

- For more information about the *CallRouter* component, refer to NN10600-405 *Nortel Multiservice Switch 7400/15000/20000 Operations: Call Server*.
- For more information about the DPN CSRM, refer to NN10600-450 *Nortel Multiservice Switch 7400: Operations: DPN-100 Interworking*.
- For more information about DPRS, see NN10600-425 *Nortel Multiservice Switch 7400/15000/20000 Operations: Dynamic Packet Routing System*.
- For more information on introducing DNA support, see Voice Networking based on DNAs and DPRS (page 109).

Procedure steps

Step	Action
1	Add a voice networking call server (VNCS) for each numbering plan. **`add vncs/<vncs_number>`**
2	Add a directory number (DN). **`add vncs/<vncs_number> dp/0 dn/<prefix>`**
3	For Voice Networking without DNA support (where the migration to DNA-based addressing support is in the planning stages or in progress) configure the VNCS's directory number to use address resolution based on PORS destination node and component identification. **`set vncs/<vncs_number> dp/<dp_number> dn/<prefix> destinationNodeId <nodeId>`** **`set vncs/<vncs_number> dp/<dp_number> dn/<prefix> destinationComponentId <Vroute>`**
4	For Voice Networking with DNA support, configure the VNCS's directory number to use address resolution based on data network address (DNA) parameters. **`set vncs/<vncs_number> dp/<dp_number> dn/<prefix> npi <npi>`** **`set vncs/<vncs_number> dp/<dp_number> dn/<prefix> dna <dna>`**
5	For a voice route on a remote node, ensure that the Nortel Multiservice Switch *CallRouter* component or Data Packet Network (DPN) call server

resource module (CSRM), is configured to recognize the DNAs that you added to the VNCS database.

6 For a hunt group server, see Configuring call hunting through a hunt group server (page 61) for the configuration procedure.

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

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

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

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

display vncs/<vncs_number> vp/<vp_number>

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

--End--

Variable definitions

Variable	Value
<dna>	is a binary coded decimal number between 0 and 15 digits in length that uniquely represents either the address of a voice route on a remote Multiservice Switch node or a hunt group server that supports call hunting. You can configure destination node and component identification and NPI and DNA values under directory numbers. If you do plan to migrate to DNA-based routing, having both sets of values configured makes the migration process easier.
<dp_number>	is the dialing plan number.
<nodeId>	is a decimal number between 0 and 4095 that corresponds to the value configured under the *ModuleData* component's *nodeId* attribute on a remote Multiservice Switch node.
<npi>	is either x121or the default value e164.
<prefix>	is 1 to 40 digits in length; can include wildcards (represented by the ? character). You can provision up to 10 000 *DirectoryNumber* component instances for each DiallingPlan component instance. For more information, see Directory numbers (page 95).

(1 of 2)

Variable	Value
<vncs_number>	is a VoiceNetworkingCallServer component instance. You can add up to 16 *VoiceNetworkingCallServer* component instances.
<vp_number>	is the voice profile number. The system provides the *VoiceProfile* component instance Vp/1 by default.
<Vroute>	is a particular *VoiceRoute* component instance on a remote Multiservice Switch node.
(2 of 2)	

Procedure job aid

VoiceNetworkingCallServer component hierarchy

❷ Multiservice Switch Subnet Routing

- **Root**
  - **VoiceNetworkingCallServer (Vncs)**/1-16
    - commentText
    - voiceRoutes (vRoutes)
    - **DiallingPlan (DP)**/0-2
      - **DirectoryNumber (Dn)**/xxx
        - destinationNodeId (nodeId)
        - destinationComponentId (dci)
        - voiceProfileNumber (vpn)
        - numberingPlanIndicator (npi)
        - dataNetworkAddress (dna)
      - **DirectoryNumber (Dn)**/yyy
    - **VoiceProfile (Vp)**/1-16
      - setupPriority (spri)
      - holdingPriority (hpri)
      - bumpPreference (bpref)
      - requiredTrafficType (rttyp)
      - permittedTrunkTypes (pttyp)
      - requiredSecurity (rsec)
      - requiredCustomerParm (rcparm)
      - pathAttributeToMinimize (pamin)
      - maximumAcceptableCost (maxCost)
      - maximumAcceptableDelay (maxDelay)
      - emissionPriority (epri)
      - discardPriority (dpri)
      - pathFailureAction (pfa)
      - optimization (opt)
      - voiceRates
      - voiceEncodingChoice
      - silenceSuppression
      - silenceSuppressionFactor
      - dtmfRegeneration
      - dataCallsAccepted (dca)
      - modemFaxRates
      - modemFaxEncodingChoice
      - faxIdleSuppressionG711G726
      - insertedOutputDelay
      - v17EncodedAsG711G726

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Configuring the VoiceProfile component

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

Prerequisites

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

- For information on defining encoding choices and rates for voice, modem/fax, and fax traffic, see Voice, modem, and facsimile encoding choices and rates (page 123) and Defining traffic handling options (page 128).

Procedure steps

Step **Action**

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

--End--

Variable definitions

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

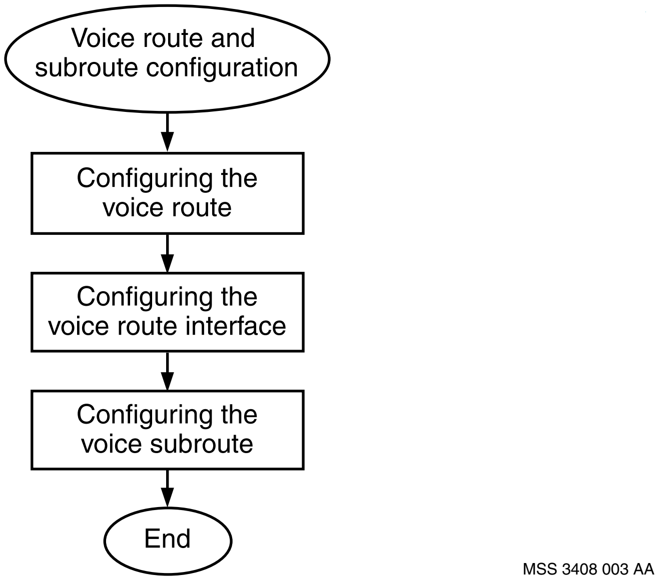
Voice route and subroute configuration

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

Voice route and subroute configuration procedures

This task flow shows you the sequence of tasks and procedures you perform to configure the voice route and subroute. To link to a procedure, go to Voice route and subroute configuration procedure navigation (page 45).

Voice route and subroute configuration procedures



Voice route and subroute configuration procedure navigation

Configuring the voice route

Configure the voice routes. to define common characteristics.

Prerequisites

- See NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference* for attribute descriptions and values related to configuring the voice route (see Voice route component hierarchy (page 48)).

Procedure steps

Step	Action
1	Add one *VoiceRoute* component for each customer group or connected PBX. **`add voiceroute/<vroute_instance>`**
2	Verify that the default settings of the provisionable attributes under the *VoiceRoute* component meet network requirements, and make changes as required. **`display vroute/<vroute_instance>`**
3	Optionally, define values for the *diallingPlan* attributes. **`set vroute/<vroute_instance> diallingPlan<0|1|2> <type>`**
4	Define the minimum number of digits that this voice route needs to receive before initializing an address resolution request to the VNCS. **`set vroute/<vroute_instance> minimumDigitsToRoute <minimum_digits>`**
5	Link the voice route to the Voice Networking call server. **`set vroute/<vroute_instance> voiceNetworkingCallServer vncs/<vncs_number>`**
6	Specify the private network identifier (PNI) number to enable MCDN connectionless services. **`set vroute/<vroute_instance> privateNetworkIdentifier <pni_number>`**
7	For Voice Networking with DNA support, specify DNA parameters to uniquely identify this voice route. **`set vroute/<vroute_instance> Dna npi <npi>`** **`set vroute/<vroute_instance> Dna dna <dna_number>`**
8	Repeat step 1 to step 7 to configure each *VoiceRoute* and *VoiceRoute Dna* component on each node connected to a PBX.

--End--

Variable definitions

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

Procedure job aid

Voice route component hierarchy

- **Root**
  - **VoiceRoute (VRoute)/1-16**
    - **DataNetworkAddress**
      - numberingPlanIndicator
      - dataNetworkAddress
      - **HuntGroupMember**
        - usageDeltaUpdateThreshold
        - **DataNetworkAddress**
          - numberingPlanIndicator
          - dataNetworkAddress

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Configuring the voice route interface

Configure the characteristics of the voice route interface.

Prerequisites

- See NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference* for attribute descriptions and values related to configuring the voice route interface (see Voice route interface component hierarchy (page 54)).
- See Voice Networking call server configuration (page 37) for more information on configuring speech activity detection and about enabling fax idle suppression.

Procedure steps

Step	Action
1	Verify that the default settings of the provisionable attributes under the *VoiceRoute Interface* component meet network requirements, and make changes as required. **display Vroute/<vroute_instance> Interface**
2	If necessary (for example, when external echo cancellation devices exist in the network), disable internal echo cancellation on voice FPs, **set Vroute/<vroute_instance> echoCancellation v1 off**
3	Optionally (for example, when external echo cancellation devices exist in the network), disable internal echo cancellation on MVP-E FPs. **set Vroute/<vroute_instance> echoCancellation v2 off**
4	On MVP-E FPs when the *echoCancellation* attribute's v2 row is set to on, define when to bypass the on-board echo canceller on MVP-E FPs. **set Vroute/<vroute_instance> Interface ecanBypassMode <type>**
5	On MVP-E FPs when the *echoCancellation* attribute's v2 row is set to on, if necessary, adjust the echo delay coverage on MVP-E FPs. **set Vroute/<vroute_instance> Interface echoTailDelay <echodelay_msec>**
6	On MVP-E FPs when the *echoCancellation* attribute's v2 row is set to on, if necessary, adjust the signal attenuation parameters on MVP-E FPs. **set Vroute/<vroute_instance> Interface echoReturnLoss <echoloss_dB>**
7	On MVP-E FPs when the *echoCancellation* attribute's v2 row is set to on, set the residual echo control to determine how to treat echo that remains after cancellation.

set Vroute/<vroute_instance> Interface residualEchoControl <control>

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

set Vroute/<vroute_instance> Interface reenableEcan <condition>

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

set Vroute/<vroute_instance> Interface echoPathLength <path_length>

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

set Vroute/<vroute_instance> Interface freezeHRegister <freeze_value>

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

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

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

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

set Vroute/<vroute_instance> Interface tandemPassThrough enabled

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

set Vroute/<vroute_instance> Interface speechHangoverTime <speechtime_msec>

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

set Vroute/<vroute_instance> Interface comfortNoiseCap <comfortnoise_dBm0>

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

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

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

--End--

Variable definitions

Variable	Value
<audiogain_+/-dB>	is ingressAudioGain, can be +12 to -12 dB, in 1 dB increments, on MVP-E FPs. For egressAudioGain, the value can be +12 to -12 dB, in 1 dB increments, on MVP-E FPs only.
<comfortnoise_dBm0>	is -40 to -78 dBm0. The default value -40 dBm0 effectively disables the cap as it represents the maximum level of comfort noise that can be generated.
<condition>	is modemDataGone or endOfCall. The default value of modemDataGone value reenables echo cancellation when the energy level is below -36 dBm0 for 300ms.
<control>	is cancelOnly, suppressResidual, or comfortNoise. The default for this value is comfortNoise. The comfortNoise value replaces the residual echo with noise at the same level as the ambient noise at the near end. The values cancelOnly and suppressResidual should be used only for testing.
<echodelay_msec>	is 32 or 64 milliseconds. The default value of 64 milliseconds is the recommended setting as it provides the greater amount of coverage.
<echoloss_dB>	is 0, 3, or 6 decibels. You can set the echoReturnLoss attribute to one of the higher decibel values when line conditions are poor and echo is a problem. Setting the *echoReturnLoss* attribute higher can improve echo cancellation performance, but reduces the signal level.
<faxtime_msec>	is any value between 300 and 20000 milliseconds (the default value is 1000 milliseconds). Fax idle suppression must be enabled (that is, the negotiated value of the *faxIdleSuppressionG711G726* attribute must be on) and the negotiated modem/fax encoding choice must be g711G726 or g726 for the *faxHangoverTimeG711G726* attribute to take effect. Setting the *faxHangoverTimeG711G726* attribute higher reduces clipping but increases bandwidth usage.
(1 of 2)	

Variable	Value
<freeze_value>	is yes or no
	Use the default value of no during normal operation. Use yes during testing.
<path_length>	is 0, 16, 32, 48, or 64 msec.
	The default for this value is 32 msec.
<speechtime_msec>	is 10 to 500 milliseconds, in 10 millisecond increments.
	The default value is 150 milliseconds. Setting the *speechHangoverTime* attribute higher reduces clipping but increases bandwidth usage. The negotiated value of the *silenceSuppression* attribute must be the value on for the *speechHangoverTime* attribute to take effect.
<type>	is one of g165 (default), g164, or never.
	The values g164 and g165 specify that the on-board echo canceller be placed in bypass mode according to ITU-T G.164 (receiving a 2100 Hz tone) and G.165 (receiving a phase-reversed 2100 Hz tone), respectively.
	Use never during debugging procedures only.
<vroute_instance>	is a value from 1 to 153.

(2 of 2)

Procedure job aid

VoiceRoute component hierarchy

Root
- **VoiceRoute (VRoute)/1-16**
  - commentText
  - subRoutes
  - typeOfRoute
  - diallingPlan0
  - diallingPlan1
  - diallingPlan2
  - huntingAlgorithm
  - minimumDigitsToRoute
  - voiceNetworkingCallServer
  - overrideDirectoryNumber
  - privateNetworkIdentifier
  - **Interface**
  - **IgDigitManipulation**
  - **EgDigitManipulation**
  - **DataNetworkAddress**
  - **VoiceAccounting**

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Voice route interface component hierarchy

```
Root
└── VoiceRoute (VRoute)/1-16
    └── Interface
          speechHangoverTime
          comfortNoiseCap
          echoCancellation
          ecanBypassMode
          echoTailDelay
          echoReturnLoss
          residualEchoControl
          freezeHRegister
          reenableEcan
          echoPathLength
          faxHangoverTimeG711G726
          modemFaxSpeechDiscrim
          ingressAudioGain
          egressAudioGain
          tandemPassThrough
```

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Configuring the voice subroute

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

Prerequisites

- See Logical and function processors (page 73) for more information about assigning channel and timeslot numbers for CAS and CCS protocols interfacing to E1 or DS1 ports.

Procedure steps

Step	Action
1	Add a voice subroute. **`add VoiceSubroute/<vsr_instance>`**
2	Add a switched voice service. **`add VoiceSubroute/<vsr_instance> SwitchedVoiceService/ <svs_number>`**
3	Link the voice subroute to the signaling channel. **`set vsr/<vsr_instance> sigChan <sigchan_number>`**
4	Link the voice subroute to the voice route. **`set vsr/<vsr_instance> vRoute/<vroute_instance>`**
5	Link each switched voice service to the bearer services configured under the E1 or DS1 port. **`set vsr/<vsr_instance> svs/<svs_number> framer interfacename lp/<lp_number> <port>/0 chan/ <channel_number>`**
6	Repeat step 5 for each *SwitchedVoiceService* component instance you provision.
7	Repeat step 1 to step 6 for each *VoiceSubroute* component instance on each node connected to a PBX.

--End--

Variable definitions

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

Procedure job aid

Voice subroute component hierarchy

```
Root
└ VoiceSubroute
    │ signallingChannel
    │ voiceRoute
    └ SwitchedVoiceService
        └ Framer
            interfaceName

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```

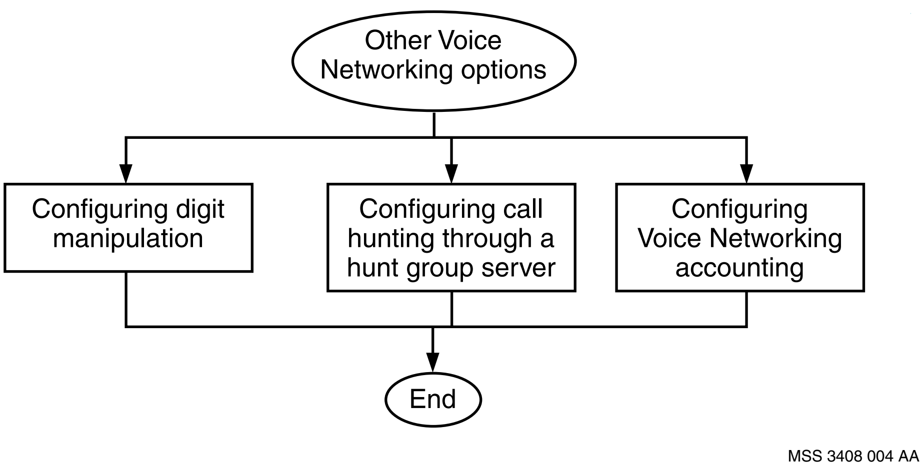
Other Voice Networking options configuration

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

Other Voice Networking options configuration procedures

This task flow shows you the sequence of procedures you perform to configure other Voice Networking options. To link to a procedure, go to Other Voice Networking options configuration procedures navigation (page 57).

Other Voice Networking options configuration procedures



Other Voice Networking options configuration procedures navigation

Configuring digit manipulation

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

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

Procedure steps

Step	Action
1	Add the digit manipulation component for either ingress or egress. **add VoiceRoute/<vroute_instance> <dm_component>**
2	Define the application of digit manipulation. **set VRoute/<vroute_instance> <dm_component> variant <var>**
3	Define the digits to insert into the called number. **set VRoute/<vroute_instance> <dm_component> digitsToInsert <digits>**
4	Define the number of digits to delete from the called number. **set VRoute/<vroute_instance> <dm_component> numDigitsToDelete <digits_delete>**
5	For digit manipulation on ingress, define which number call routing acts on. **set VRoute/<vroute_instance> <dm_component> callRouting <cr_point>**
6	For digit manipulation on egress, define how voice networking handles calls for which no digits remain after digit manipulation. **set VRoute/<vroute_instance> <dm_component> callHandlingWhenNoDigits <choice>**

--End--

Variable definitions

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

Procedure job aid

Digit manipulation component hierarchy

- **Root**
  - **VoiceRoute (VRoute)/1-16**
    - **IgDigitManipulation**
      - variant
      - digitsToInsert
      - numDigitsToDelete
      - callRouting
    - **EgDigitManipulation**
      - variant
      - digitsToInsert
      - numDigitsToDelete
      - callHandlingWhenNoDigits

PPT 2984 004 AA

Configuring call hunting through a hunt group server

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

Prerequisites

- See NN10600-415 *Nortel Multiservice Switch 7400/15000/20000 Operations: Hunt Group Server* for descriptions and procedures on setting up a hunt group server.
- See NN10600-410 *Nortel Multiservice Switch 7400/15000/20000 Operations: Call Redirection Server* for descriptions and procedures on the CRS.

Procedure steps

Step	Action
1	Configure the primary hunt group server.
2	Optionally, configure a backup hunt group server.
3	Provision the call redirection server (CRS) to associate the backup server with the primary server.
4	Identify the DNA of the primary hunt group server in the *dataNetworkAddress* attribute under the *VoiceNetworkingCallServer DiallingPlan DirectoryNumber* component.
5	Define the voice route as a hunt group member and add the primary *VoiceRoute DataNetworkAddress HuntGroupMember HuntGroupAddress/0* component instance. **add Vroute/<vroute_instance> DataNetworkAddress HuntGroupMember**
6	Optionally, set the number of channels that must be free or occupied before the availability message packet (AMP) is sent to the primary and backup hunt group servers. **set Vroute/<vroute_instance> DataNetworkAddress HuntGroupMember usageDeltaUpdateThreshold <udut>**
7	If a backup hunt group server is configured, add the component for this backup under the voice route DNA. **add Vroute/<vroute_instance> DataNetworkAddress HuntGroupMember HuntGroupAddress/1**
8	Set the address of the primary hunt group server to which the *VoiceRoute* DNA belongs. **set Vroute/<vroute_instance> DataNetworkAddress HuntGroupMember HuntGroupAddress/0 numberingPlanIndicator <npi>**

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

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

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

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

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

--End--

Variable definitions

Variable	Value
<dna>	is an address string of between 1 and 15 characters, which identifies the DNA of the hunt group server.
<npi>	is either x121 or e164. The default is x121.
<udut>	is a value between 1 and 4096. The default is 1.
<vroute_instance>	is the voice route instance.

Configuring Voice Networking accounting

Set up Nortel Multiservice Switch nodes to collect accounting statistics for calls processed by the VoiceRoute component.

Prerequisites

- Configure basic accounting using the procedures in NN10600-560 *Nortel Multiservice Switch 7400/15000/20000 Accounting*.

Procedure steps

Step	Action
1	Configure DNA parameters for the *VoiceRoute* component (if you have not already done so). **`set VRoute/<vroute_instance> Dna dna <dna#>`** **`set VRoute/<vroute_instance> Dna npi <type>`**
2	Specify the reason for generating accounting records. **`set VRoute/<vroute_instance> VoiceAccounting accountCollection <reason>`**
3	Specify an accounting class. **`set VRoute/<vroute_instance> VoiceAccounting accountClass <accountclass#>`**
4	Specify a data service exchange value. **`set VRoute/<vroute_instance> VoiceAccounting serviceExchange <serviceexchange#>`**
5	Specify the number of trailing digits to be removed from the accounting records. **`set VRoute/<vroute_instance> VoiceAccounting digitsSuppressed <digitssuppressed#>`**
6	Specify accounting options. The value suppressTerminatingEndRecords stops all accounting currently in progress for this VoiceRoute component instance. **`set VRoute/<vroute_instance> VoiceAccounting accountingOptions suppressTerminatingEndRecords`**
7	Repeat steps 1 to 6 to configure Voice Networking accounting on each candidate node.

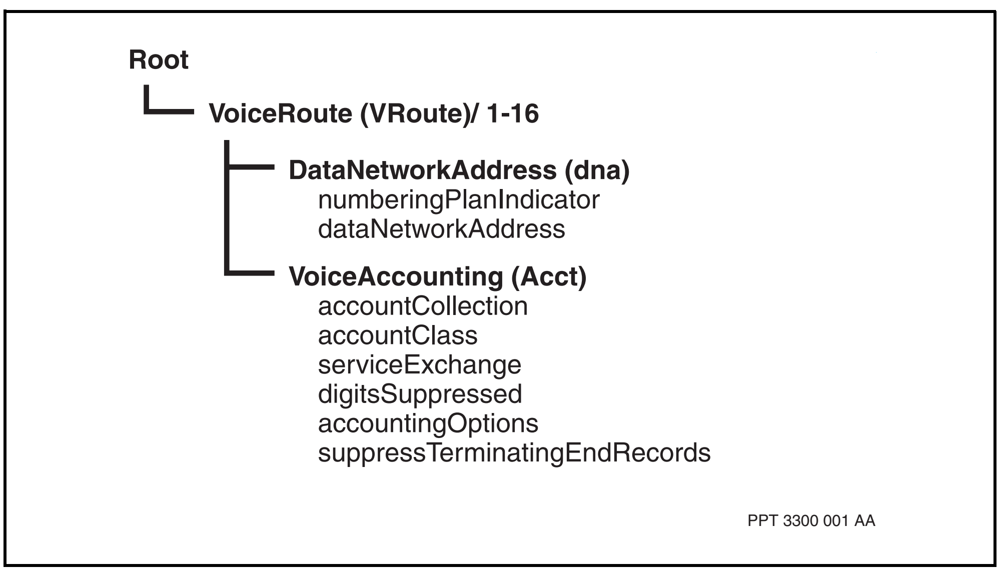
--End--

Variable definitions

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

Procedure job aid

Voice Networking accounting component hierarchy



Voice Networking fundamentals

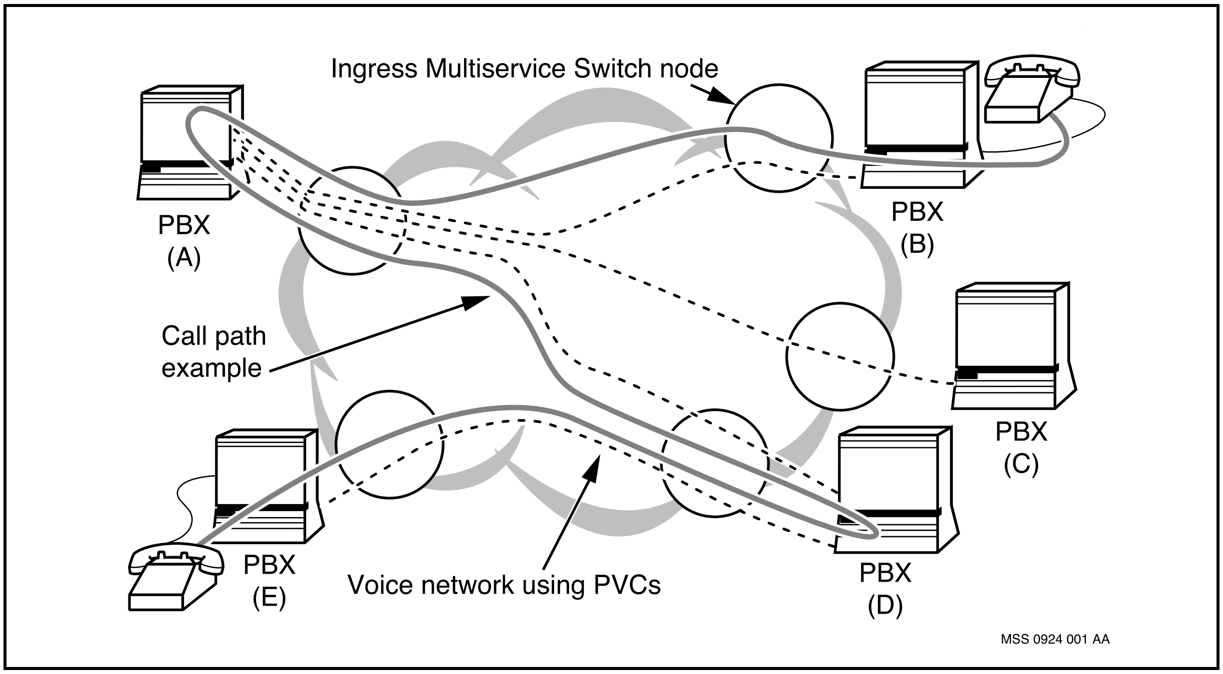
The Voice Networking service dynamically routes speech, modem, facsimile, and data calls through a Nortel Multiservice Switch network to addresses (dialed numbers) provided by the calling PBX.

In a Multiservice Switch network without Voice Networking, see the figure Multiservice Switch network without Voice Networking (page 66), permanent virtual circuits (PVC) transport calls between the ingress and egress nodes. PVCs, although continuously available:

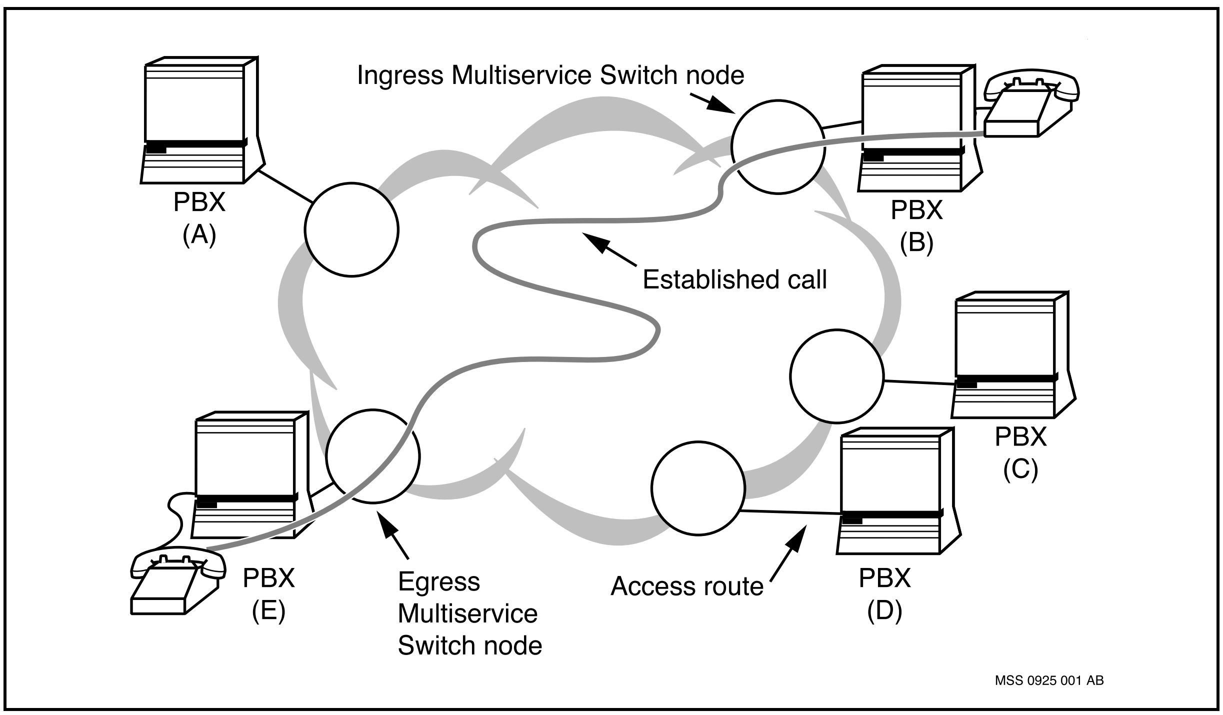
- do not always provide the most direct or efficient route
- reserve a certain amount of bandwidth even when idle
- require that you define and configure both endpoints of a connection
- can increase the potential for signal delays and distortions. For example, a call from PBX B to PBX E, as shown in the figure Multiservice Switch network without Voice Networking (page 66), requires routing through two tandem PBXs.

In a Multiservice Switch network with Voice Networking, the best path between the ingress and egress nodes is selected on a call-by-call basis by means of a switched virtual circuit (SVC). As shown in the figure Multiservice Switch network with Voice Networking (page 66), Voice Networking routes a call from PBX B to PBX E over an SVC that represents the most efficient path and accounts for configured parameters and bandwidth availability at the time of the call. When a Voice Networking call clears, the SVC is torn down, freeing up bandwidth in the subnet for subsequent calls. With Voice Networking, you only provision one endpoint of a connection—the dialed number.

Multiservice Switch network without Voice Networking



Multiservice Switch network with Voice Networking



Navigation

Software requirements for Voice Networking

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

- path-oriented routing system (PORS). The following guides contain information about PORS:
  - — NN10600-750 *Nortel Multiservice Switch 7400 Operations: Voice Transport*
  - — NN10600-030 *Nortel Multiservice Switch 7400/15000/20000 Overview*
  - — NN10600-435 *Nortel Multiservice Switch 7400/15000/20000 Operations: Path-Oriented Routing System*
- network clock synchronization (NCS). NCS is described in NN10600-550 *Nortel Multiservice Switch 7400/15000/20000 Common Configuration Procedures*.
- Dynamic Packet Routing System (DPRS). You require DPRS only if you employ data network address (DNA) based routing. See Routing fundamentals (page 93) for more details. The following guides contain information about DPRS:
  - — NN10600-405 *Nortel Multiservice Switch 7400/15000/20000 Operations: Call Server*
  - — NN10600-425 *Nortel Multiservice Switch 7400/15000/20000 Operations: Dynamic Packet Routing System*
  - — NN10600-450 *Nortel Multiservice Switch 7400: Operations: DPN-100 Interworking*

Hardware requirements for Voice Networking

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

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

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

For a description of the hardware elements of these FPs, see NN10600-170 *Nortel Multiservice Switch 7400 Hardware Description*.

Communication with PBXs

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

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

Voice Networking supports a number of signaling protocols. See Signaling protocols fundamentals (page 78) for more information. For compliance statement information, see Compliance with standards—Voice Networking signaling protocols (page 132).

Audio handling capabilities

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

- define quality of service parameters based on customer needs
- maximize bandwidth savings by using compression and both silence and idle period suppression techniques on both voice and voice-band data traffic

- provide toll-quality voice signals by using echo cancellation, comfort noise generation, and congestion handling techniques
- engineer the flow of audio data to meet network requirements

You define Voice Networking audio handling capabilities by configuring voice routes and voice profiles. To configure the voice route and voice profile, you provision the attributes under the *VoiceRoute Interface* and V*oiceNetworkingCallServer VoiceProfile* components. To configure audio handling capabilities, see Voice Networking configuration (page 8).

Call routing for Voice Networking

There are three types of routing in Voice Networking.

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

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

For a description of these routing features, see Routing fundamentals (page 93).

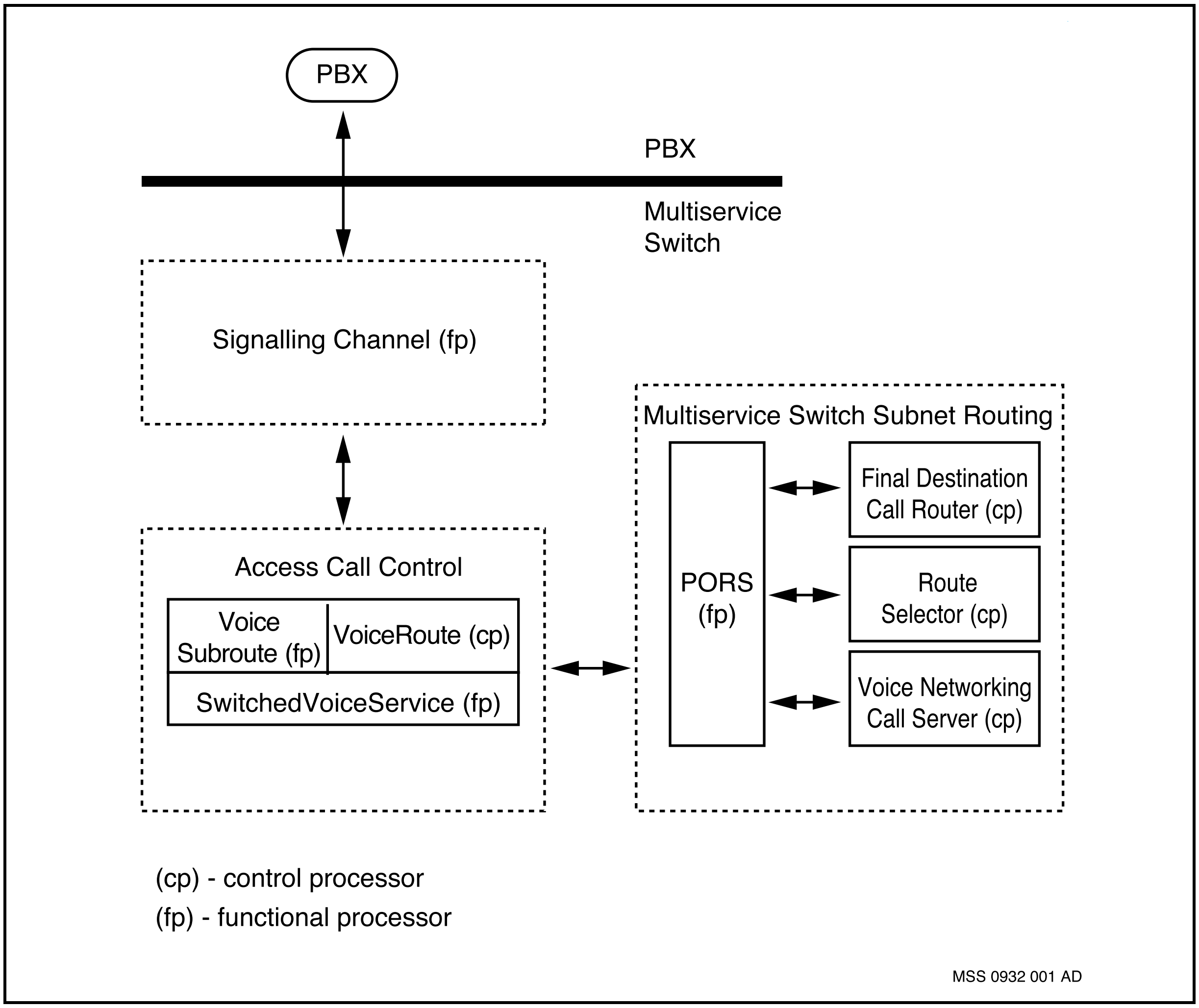
Relationship between Voice Networking software components and systems

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

- Access protocol control (page 72)
- Access call control (page 72)
- Multiservice Switch subnet routing (page 72)

The figure Relationship between Voice Networking components and systems (page 70) describes the relationship between the three groups of components and systems and indicates whether the component or system resides on the control or function processor.

Relationship between Voice Networking components and systems



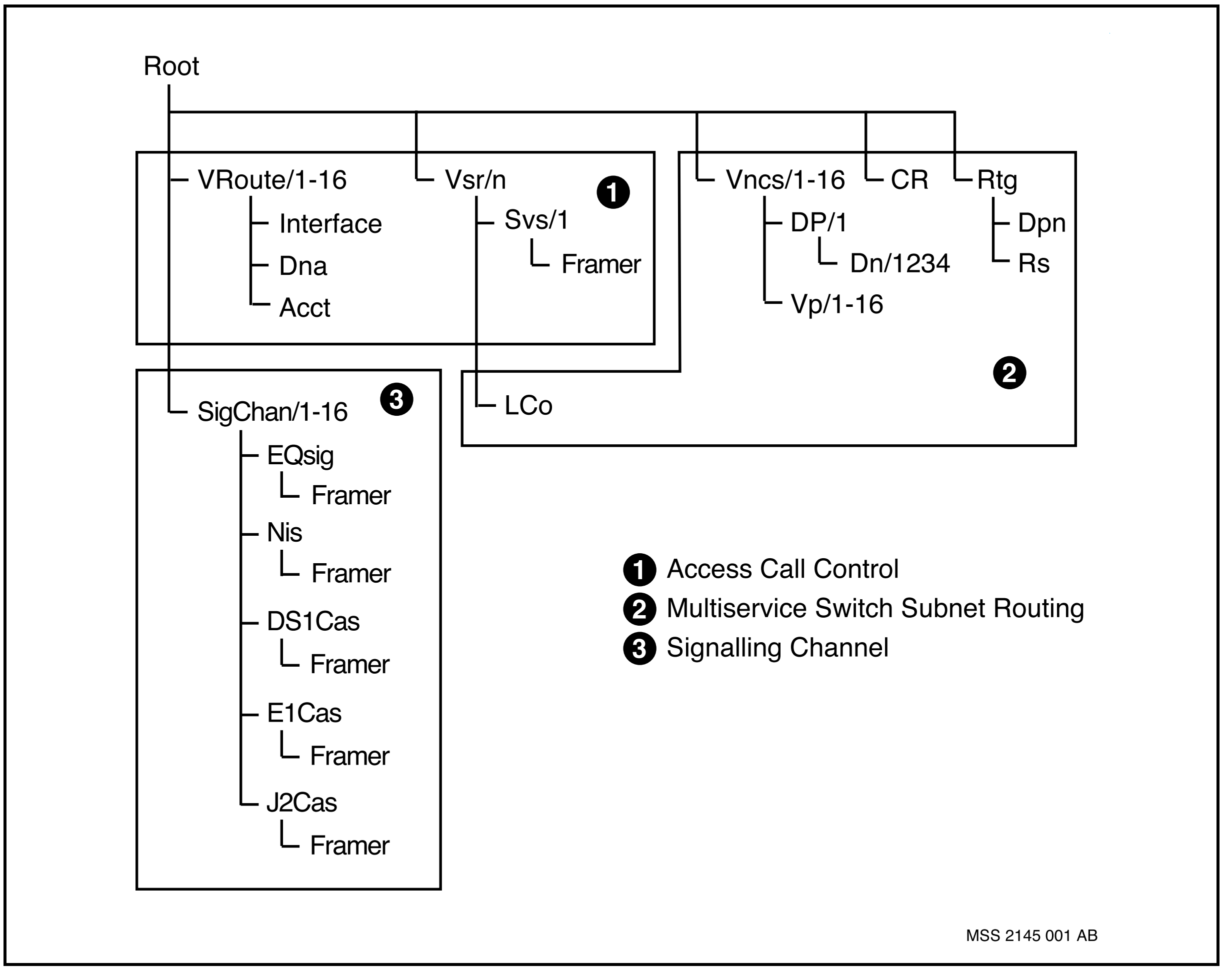
Component relationships

The component hierarchy in the figure Hierarchy of Voice Networking and related provisionable components (page 71) provides a high-level view of the relationship between Voice Networking and related components. Observe the following rules when configuring Voice Networking components:

- There can be up to 16 instances of the *VoiceNetworkingCallServer* component. Each *VoiceNetworkingCallServe*r component instance represents one numbering plan.
- There can be multiple *VoiceRoute* components per Nortel Multiservice Switch node.
- A *VoiceRoute* component can have up to 16 *VoiceSubroute* components, but a *VoiceSubroute* component belongs to only one *VoiceRoute* component.

- A *DataNetworkAddress* component under the *VoiceRoute* component can be defined as a hunt group member through the *HuntGroupMember* and *HuntGroupAddress* components.
- A *VoiceSubroute* component belongs to a DS1 or E1 link (that is, it controls a collection of 64 kbit/s channels on a DS1 or E1 link).
- A *SignallingChannel* component can have multiple *VoiceRoute* and *VoiceSubroute* components.
- A *SignallingChannel* component cannot span multiple DS1 or E1 links.
- A *VoiceRoute* component can have multiple *SignallingChannel* components.
- A DS1 or E1 link can have multiple *VoiceSubroute* components, hence multiple *VoiceRoute* components.
- A *SignallingChannel* component can control up to four *VoiceSubroute* components.

Hierarchy of Voice Networking and related provisionable components



Access protocol control

The *SignallingChannel* component comprises access protocol control. The *SignallingChannel* component defines the signaling characteristics of the connection to the PBX (the external equipment). Voice Networking supports standards-based signaling protocols for both CCS and CAS signaling formats. The *SignallingChannel* component responds to signals received from a PBX and communicates with the *VoiceSubroute* component to initiate call setup and tear-down processes. For details on the signaling protocols supported by Voice Networking, see Signaling protocols fundamentals (page 78).

Access call control

The *VoiceRoute*, *VoiceSubroute*, and *SwitchedVoiceService* components comprise access call control. These three components control the progress and state of a call. For information about the *VoiceRoute*, *VoiceSubroute*, and *SwitchedVoiceService* components and their functions, refer to the NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference*.

Multiservice Switch subnet routing

The PORS and *VoiceNetworkingCallServer*, *RouteSelector*, and *FinalDestinationCallRouter* components comprise Nortel Multiservice Switch subnet routing. Subnet routing elements interact with the *VoiceRoute*, *VoiceSubroute*, and *SwitchedVoiceService* components to find and establish a path across the subnet.

Parameters that impact Voice Networking configuration

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

- Installation (page 72)
- Operational (page 73)
- Logical and function processors (page 73)
- Interworking and gateways (page 75)

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

Installation

When installing Voice Networking, consider the following:

- Voice Networking cannot co-exist on a 1-port MVP-E FP with Voice Transport or the bit transparent data service (BTDS). These services can co-exist on a 4-port MVP-E FP with the restriction of one service type per port.

- The TTC2M MVP-E FP only supports channel associated signaling (CAS) and cannot be used for common channel signaling (CCS) as it does not support the high-level data link control (HDLC) protocol.
- The prior constraint of all ports on a single 4pMVPe FP restricted to running the same protocol no longer applies. Any two protocols (any mix of CAS, CCS, VNET, VTDS) configured across any of the four ports is supported.
- Voice Networking does not support interworking between 4-port MVP-E FPs and Voice FPs
- The 4-port MVP-E FP can interwork with the Passport 4400 series.
- Interworking between the 4-port MVP-E FPs running PCR 4.2 or higher and the 1-port MVP-E FPs running Release 7.0 is supported.

Operational

The Voice Networking service operates under the following conditions:

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

Logical and function processors

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

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

The table FP-to-capability relationships (page 74) describes Voice Networking capability-to-card relationships and restrictions.

FP-to-capability relationships

Capability	card type				
	DS1 MVP-E	**E1 MVP-E**	**DS1 MVP-E (4-port) (see 4)**	**E1 MVP-E (4-port) (see 4)**	**TTC2M MVP-E**
G.728 (LD-CELP) 16 kbit/s voice compression (see 1)	X	X	X	X	X
G.729/G.729A (CS-ACELP) 8 kbit/s voice compression (see 1)	X	X	X	X	X
Fax relay (see 1)	X	X	X (see 2)	X (see 2)	X
G.726 (ADPCM) voice and modem/fax encoding	X	X	X	X	X
Dynamic up- and down-speeding (G.711/ G.726) of audio traffic (see 4)	X	X	X	X	X
Silence suppression, with provisionable support for capping comfort noise	X	X	X	X	X
Fax idle suppression	X	X	X	X	X
Provisionable speech and fax hangover time	X	X	X	X	X
Tandem pass through (see 1)	X	X	X	X	X
Configurable egress buffer	X	X	X	X	X
Facsimile/speech discrimination	X	X	X	X	X
DTMF tone detection and regeneration	X	X	X	X	X
ETSI QSIG segmentation		X		X	
Echo cancellation	X	X	X	X	X
CCS-to-CAS protocol gateway support (see 3)	X	X	X	X	
(1 of 2)					

FP-to-capability relationships (continued)

Capability	**card type**				
	DS1 MVP-E	**E1 MVP-E**	**DS1 MVP-E (4-port) (see 4)**	**E1 MVP-E (4-port) (see 4)**	**TTC2M MVP-E**
Hunt Group Server-based routing	X	X	X	X	X
1) To use G.728 and G.729/729A voice encoding, fax relay, and tandem pass through on an MVP-E FP, you must add g728, g729, faxRelay, and tandemPassThrough, respectively, to the *featureList* attribute under the *Software LogicalProcessorType* component, along with the *voiceNetworking* attribute (G.729A voice encoding is only supported on MVP-E FPs). Refer to the configuration procedures in Voice Networking configuration (page 8) for more details. 2) Since 4-port MVP-E FPs support fax relay V.17 fax calls, you must provision the *v17EncodedAsG711G726* attribute with the value no. 3) CCS-to-CAS protocol gateway support is automatically included when you provision the values MCDN, etsiQsig, nisSig, or casSig for the *featureList* attribute under the *Software LogicalProcessorType* component. For a description of the CCS-to-CAS signaling gateways, see Interworking (page 88). 4) Requires PCR 3.0 software.					
(2 of 2)					

Interworking and gateways

Interworking refers to the interworking of signaling protocols and to connections between MVP-E FPs. For more information on interworking and gateways, see Protocol parameters (page 75) for more information.

Protocol parameters

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

- The table Card types and supported Voice Networking protocols (page 76) maps Voice Networking function processors (FPs) to the protocols that they support
- The table Voice Networking protocol interworking (page 76) describes the protocol interworking supported by Voice Networking. This table only captures interworking scenarios and does not reflect that, by default, native or like protocols communicate with each other (see gray cells in the table).
- The table Voice Networking CCS-to-CAS protocol gateways (page 76) describes the CCS-to-CAS protocol gateways supported by Voice Networking. See Protocol interworking and gateways (page 87) for details about the functionality included with protocol interworking and gateways.

Card types and supported Voice Networking protocols

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

Voice Networking protocol interworking

	Protocols					
Protocols	E1 CAS	DS1 CAS	J2 CAS	NIS	MCDN	Euro-ISDN
E1 CAS		X	X			
DS1 CAS	X		X			
J2 CAS	X	X				
NIS					X	
MCDN				X		

Voice Networking CCS-to-CAS protocol gateways

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

Voice Networking accounting

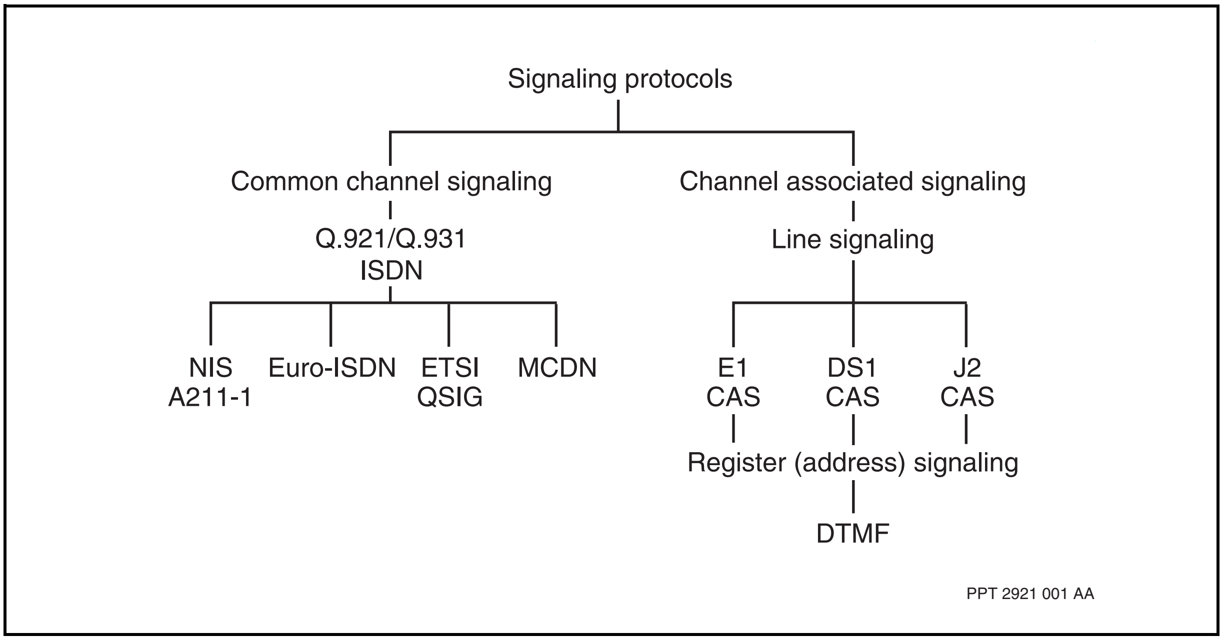
The *VoiceAccounting* component controls accounting for calls processed by its parent component, the *VoiceRoute*. Accounting is turned off by default. Any configuration changes made to the default values of the *VoiceAccounting* component's attributes will take effect on calls that are set up after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

Signaling protocols fundamentals

This chapter provides information about Voice Networking and signaling protocols.

Navigation

Signaling protocols supported by Voice Networking



Common channel signaling (CCS)

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

- Network interface specification (NIS) A211-1 (page 79)
- European Telecommunications Standards Institute Q interface signaling (ETSI QSIG) (page 80)
- Meridian customer defined networking (MCDN) (page 82)
- European Digital Subscriber Signaling System Number One (EDSS1) or Euro-ISDN (page 84)

CCS protocols use one timeslot or channel—called the delta or D-channel—of a given digital link to carry the signaling information for the calls on the remaining channels—called the bearer or B-channels. For DS1 links using CCS protocols, timeslot 24 carries signaling information. For E1 links using CCS protocols, timeslot 16 carries signaling information. Primary rate interface (PRI) is the name given to the DS1 or E1 digital link between the Nortel Multiservice Switch node and PBX (see the figure Typical Multiservice Switch to PBX protocol connection (page 88)).

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

For information on the cause codes Voice Networking supports for each CCS protocol, see Call release cause codes (page 160).

Network interface specification (NIS) A211-1

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

NIS A211-1 supports basic call and supplementary services as shown in the table Supported NIS A211-1 basic call and supplementary services (page 80). For additional information on NIS A211-1 compliance, see Compliance statement for NIS A211-1 (page 133).

Supported NIS A211-1 basic call and supplementary services

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

European Telecommunications Standards Institute Q interface signaling (ETSI QSIG)

ETSI QSIG is a private networking protocol. Voice Networking's application of the ETSI QSIG signaling protocol provides user-to-user interface signaling (that is, between PBXs) over a Nortel Multiservice Switch network. Voice Networking also supports the handling of large ISDN messages produced by particular PBXs by using a segmentation and reassembly technique. As shown in the table Supported ETSI QSIG basic call and supplementary services (page 81), ETSI QSIG supports basic call and supplementary services such as call transfer and name display. For additional ETSI QSIG compliance information, see Supported ETSI QSIG basic call and supplementary services (page 81).

Supported ETSI QSIG basic call and supplementary services

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

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

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

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

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

Meridian customer defined networking (MCDN)

MCDN is a proprietary, private networking protocol developed by Nortel Networks for the Meridian 1 PBX. Voice Networking supports the Peer-to-Peer MCDN variant. Voice Networking support of the MCDN Peer-to-Peer signaling protocol provides user-to-user (Meridian 1-to-Meridian 1 PBX) interface signaling over a Nortel Multiservice Switch network. Multiservice Switch nodes provide tandem node call processing for MCDN basic call and supplementary services. the table Supported MCDN basic call and supplementary services (page 83) contains information about the MCDN basic call and supplementary services that Voice Networking support including network attendant services (NAS), network messaging services (voice mail), automatic call distribution (ACD), and anti-tromboning. For additional MCDN compliance information, see Compliance statement for MCDN (page 134).

Some of the capabilities and services listed in the table Supported MCDN basic call and supplementary services (page 83) can be enabled and disabled through configuration. For more information about configuring MCDN capabilities and services, see Configuring the MCDN signaling protocol (page 30).

Supported MCDN basic call and supplementary services

Category	Details
Basic call services	• Call control and link and channel maintenance messaging • Call setup and tear-down for audio traffic (voice, modem, and facsimile); 64 kbit/s (voice and unrestricted) and 56 kbit/s data bearer services • Channel negotiation • Overlap dialing • Called and calling party number transport • Federal Communications Commission (FCC) equal access for transporting the carrier access code (CAC) with the called party number • Flexible numbering plan
Connection-oriented supplementary services	• Attendant and network attendant services (NAS), including break-in, schedule, night service, timed reminder, and call extension • Anti-tromboning (both trunk and NAS); electronic switched network (ESN) and related services, including basic and network alternate route selection (BARS and NARS), coordinated dialing plan, time of day and automatic least cost routing; network and travelling class of service (NCOS and TCOS); off-hook and remote virtual queuing • Call forward, page, park, hunt (internal and external), pickup (network wide), redirection, transfer, and trace (including malicious); drop back busy • Number and name display with privacy options • Network authorization codes (transmitted as DTMF digits) • Trunk route optimization • Access to intercom and radio paging systems
Connectionless supplementary services	• Network messaging services (voice mail) and related message waiting signaling, call pickup queries (world wide), and electronic lock • Automatic call distribution (ACD) between pools of agents • Remote call forward • Ring again (busy and no reply) • Network time synchronization • Multisite relocation of wireless phones

MCDN signaling protocol configuration

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

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

The following information pertains to MCDN anti-tromboning:

- The value antiTromboning includes both NAS and trunk anti-tromboning.
- For anti-tromboning requests involving a dual egress call scenario (that is, a tandem PBX was used to dial out to the destination and out to the user before dropping out), Voice Networking bills the tandem node for accounting purposes.
- To prevent call failures when anti-tromboning is enabled, (1) each configured directory number (DN) must point to the same destination, and (2) each node in the network must have its own DNs configured in the voice networking call server's database. See Configuring the Voice Networking call server (page 38) for more information.

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

Euro-ISDN is a common implementation of ISDN signaling standards that includes the different ISDN applications used in separate European countries. Euro-ISDN is defined in ETSI specifications and based on ITU-T recommendations. Voice Networking supports three Euro-ISDN variants: generic (non-country specific or ETSI ISDN), Austrian, and German. Voice Networking transports a number of Euro-ISDN basic call and supplementary services (see the table Supported Euro-ISDN basic call and supplementary services (page 85)).

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

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

For Euro-ISDN compliance with standards information, see Compliance statement for Euro-ISDN (page 134).

Supported Euro-ISDN basic call and supplementary services

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

Euro-ISDN signaling protocol configuration

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

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

Channel associated signaling (CAS)

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

- DS1 CAS (page 87)
- E1 CAS (page 87)
- J2 CAS (page 87)

CAS is a method of carrying signaling information for all of the voice timeslots of a digital connection. CAS methods vary according to the signaling formats supported by the interface: DS1, E1, or TTC. The interface defines if signaling information is carried in each traffic-carrying timeslot or in a dedicated signaling channel. CAS includes both line and register signaling. Line signaling includes the state change of distinct signaling bits—A, AB, or ABCD. An example of a state change is going from idle to seized. The actual number of signaling bits used (1, 2, or 4) depends on the interface. Register or address signaling always follows line signaling. Voice Networking only supports dual-tone multifrequency (DTMF) register signaling for digit transmission, and does not support pulsed signaling. Both ends of a call must support DTMF signaling. For more information, see CAS interconnection guidelines (page 86).

CAS interconnection guidelines

Voice Networking can support proprietary or standards-based CAS variants. The CAS variant must use DTMF for digit transmission and steady state signaling. The state changes supported by Voice Networking include idle, seize, connect (answer), wink, seize acknowledgment, clear back, and clear forward. The state changes match the appropriate signaling bit combination (either A, AB, or ABCD). You match the state change indications generated by the connected equipment by configuring the attributes under the *AbcdProv* and *TimerProv* groups. You find the AbcdProv and TimerProv groups of attributes under the *Ds1Cas*, *E1Cas*, and *J2Cas* components. The attributes in these two groups allow Voice Networking to support different combinations of signaling bits according to the equipment or signaling type used.

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

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

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

DS1 CAS

Nortel Multiservice Switch device's DS1 CAS line signaling supports the following two types of DS1 interface framing formats:

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

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

E1 CAS

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

J2 CAS

Nortel Multiservice Switch device's J2 CAS line signaling applies to TTC lines. J2 CAS uses timeslot 16 to carry signaling information and uses one signaling bit—A—to represent all required line states.

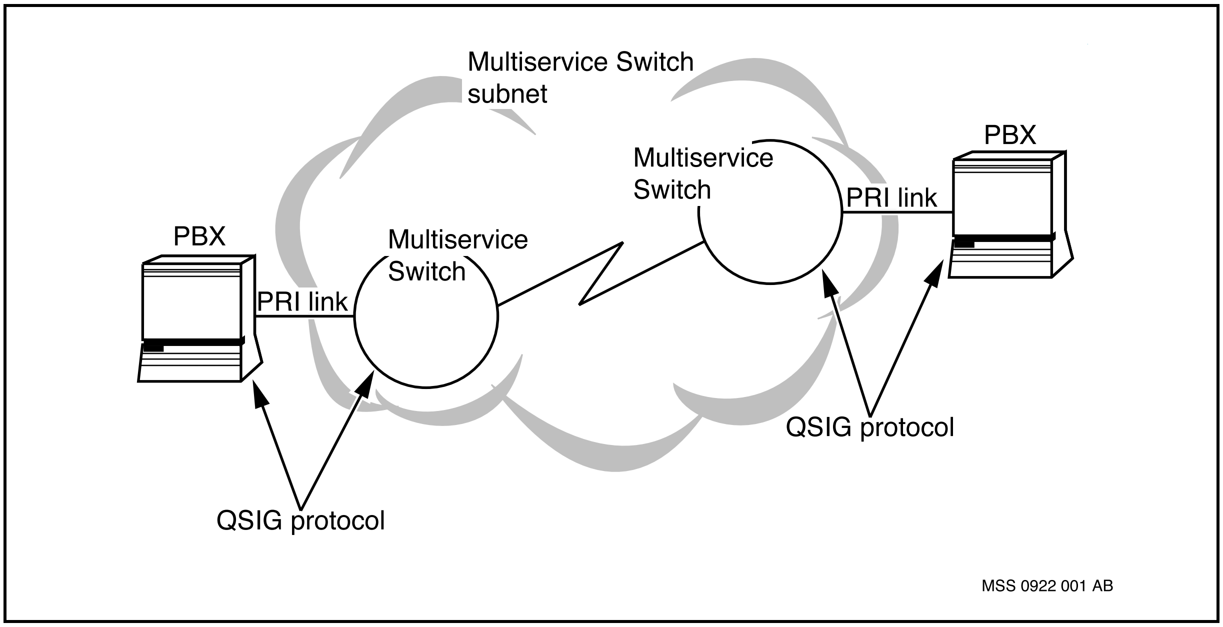
Protocol interworking and gateways

Typically, the signaling protocol is identical on all Nortel Multiservice Switch nodes that operate the Voice Networking service (see the figure Typical Multiservice Switch to PBX protocol connection (page 88)). For example, there is no gateway function or interworking between the CCS protocols NIS A211-1 and ETSI QSIG. If a network of PBXs contains different CCS protocols that require protocol gateway processing, a tandem PBX and not the node must provide a protocol gateway function.

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

- Interworking (page 88)
- Gateways (page 89)

Typical Multiservice Switch to PBX protocol connection



Interworking

With protocol interworking, no conversion takes place between the signaling protocols involved in a call. Voice Networking supports limited CCS-to-CCS interworking and full CAS-to-CAS interworking. The table Protocol interworking scenarios (page 89) contains details about the functionality included in each interworking scenario.

Protocol interworking scenarios

Interworking scenario	Details
NIS A211-1-to-MCDN	Functionality includes call setup and tear-down, progress indicators, and supplementary services such as calling and connected line identification presentation and restriction (CLIP and CLIR, respectively).
Euro-ISDN private, network-side variant (generic ETSI ISDN)-to-Euro-ISDN PSTN, user-side variant (austria or germany)	As stated in European Digital Subscriber Signaling System Number One (EDSS1) or Euro-ISDN (page 84), Voice Networking • supports all Euro-ISDN basic call services • guarantees the transport of Euro-ISDN supplementary services between a PBX (private network) and a CO (public network)
CAS variant-to-CAS variant	Interworking is supported between CAS protocols regardless of the signaling format and interface combination involved in a call (for example, a call between DS1 CAS and E1 CAS or J2 CAS).

Gateways

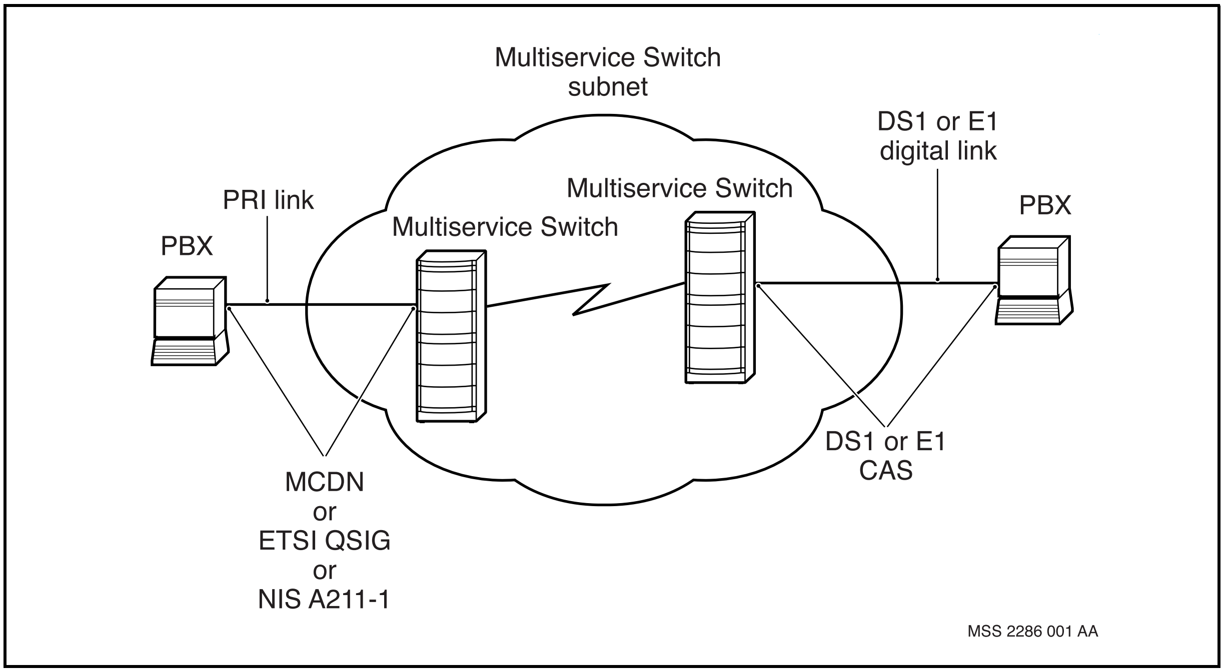
With a protocol gateway, some degree of conversion occurs between two different signaling types. During end-to-end negotiation, calls requiring gateway processing are either accepted or rejected, depending on whether the requested gateway is supported. The egress or destination node makes the determination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

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

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

The CCS-to-CAS protocol gateways support both overlap and enbloc dialing, but only deliver basic call features (such as call setup, tear-down, and messaging for unsuccessful call handling). That is, the CCS-to-CAS protocol gateways do not deliver the supplementary call features supported by MCDN, ETSI QSIG and NIS A211-1. The figure Voice Networking protocol gateways (page 90) describes the protocol gateways supported by Voice Networking.

Voice Networking protocol gateways



Individual channel busy-back

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

Signaling protocols components

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

Consider the following information when configuring the *SignallingChannel* component in Configuring the SignalingChannel component (page 20).

- Voice Networking supports protocol interworking and protocol gateways.
- It is not necessary at this point to provision the subroutes attribute. When you add *VoiceSubroute* components, you can link them to a particular *SignallingChannel* component instance (see Voice subroute (page 131)).

- Interworking between different CAS interface signaling types occurs automatically when you configure the casSig protocol software feature on Nortel Multiservice Switch nodes using different interfaces (for example, DS1 and E1)
- The SigChan instance numbers range from 1 to 153.
- EuroISDN to CAS Gateway is not supported for any of the Voice FPs.

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

The table Relationship between defaultNpiTon attribute and PBX NPI/TON values (page 91) describes how the values of *defaultNpiTon* map to NPI and TON information supplied by the calling PBX.

The values under the *defaultNpiTon* attribute correspond to the default values available under the *VoiceRoute* component's *diallingPlan0* attribute. For more information on how Voice Networking interprets NPI and TON information supplied by the calling PBX, see Interaction between configurable components and attributes during call routing (page 97).

Relationship between defaultNpiTon attribute and PBX NPI/TON values

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

Internal cause values

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

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

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

If you do not specify a value for these attributes, or if you do not add the *InternalCauseMap* component, Voice Networking does not modify or manipulate internally generated cause codes. See Call release cause codes (page 160) for more information.

Routing fundamentals

Nortel Multiservice Switch Voice Networking supports routing based on:

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

Navigation

Routing parameters

Routing depends on configuration of the following parameters:

- numbering plans
- directory numbers
- dialing mode

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

- Numbering plans (page 94)
- Directory numbers (page 95)

Numbering plans

A numbering plan consists of the following:

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

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

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

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

The table Partial numbering plan of a VoiceNetworkingCallServer component (page 96) illustrates a partial numbering plan.

Directory numbers

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

When defining directory number prefixes for dialed numbers, you can use one or multiple wildcard characters, represented by the question mark character (?). The wildcard character in a directory number prefix instructs Voice Networking to ignore the digit in that position when making routing decisions. By using wildcards, you can simplify the configuration process by reducing the number of directory numbers needed to properly route calls. For example, by using the directory number prefix 3?1, you can combine dialed numbers beginning with the following digits under one DirectoryNumber component: 301, 311, 321, 331, 341, 351, 361, 371, 381 and 391. As well, the use of wildcards is important in voice numbering plans that place access codes in arbitrary locations in the dialed number. See the table Partial numbering plan of a VoiceNetworkingCallServer component (page 96) for an example of how Voice Networking uses directory numbers to route dialed numbers.

Overlap and enbloc dialing

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

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

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

For both the PBX and the node, the settings for overlap sending and overlap receiving should be the same. Set both the PBX and the node to either overlap dialing or enbloc dialing.

Examples of resolving dialed numbers through VNCS

Examples of the best match algorithm, as applied to the partial numbering plan shown in the table Partial numbering plan of a VoiceNetworkingCallServer component (page 96), are as follows:

- 6137527 maps to nodeId 26
- 61375127 maps to nodeId 25
- 6137755325 maps to nodeId 31
- 61377553251 maps to nodeId 31
- 0 maps to nodeId 20

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

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

Partial numbering plan of a VoiceNetworkingCallServer component

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

Partial numbering plan of a VoiceNetworkingCallServer component (continued)

Directory number prefix	destination-NodeId attribute	destination-ComponentId attribute	(NPI) and DNA	voiceProfile-Number attribute
613768	20	VoiceRoute/1	(X.121) 55554000200001	3
6137512	25	VoiceRoute/2	(X.121) 55554000250002	3
613752	26	VoiceRoute/1	(X.121) 55554000260001	1
1800	45	VoiceRoute/1	(X.121) 55554000450001	1
0	20	VoiceRoute/3	(X.121) 55554000200003	3
The ? character is a wildcard.				
(2 of 2)				

Interaction between configurable components and attributes during call routing

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

The mapping process occurs in the following order:

1. The PBX numbering plan indicator (NPI) and type of number (TON) that are associated with the dialed number map to a Multiservice Switch node term (for example, international). See Mapping numbers—PBX to Multiservice Switch (page 100).
2. The node term maps to a particular *VoiceRoute* component *diallingPlan* attribute (for example, dp1). See Mapping numbers—Multiservice Switch term to VoiceRoute diallingPlan attribute (page 101).

3 The *VoiceRoute* component *diallingPlan* attribute maps to a *VoiceNetworkingCallServer DiallingPlan* component instance (for example, DP/1). See Mapping numbers—VoiceRoute to VoiceNetworkingCallServer DiallingPlan component (page 102).

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

Call routing example

The following steps show how Voice Networking routes a call using PORS SVCs without DNA support. The figure Conceptual example of Voice Networking call routing (without DNA support) (page 100) illustrates the progression of these steps.

The following steps describe the concepts illustrated in Conceptual example of Voice Networking call routing (without DNA support) (page 100):

1 A subscriber dials 4464.

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

---

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

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

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

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

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

7 Vsr/1 maps to VoiceRoute/1.

8 The combination of NPI 1001 and TON 001 (from the PBX) maps to node term P1 by way of a lookup table. See the table Mapping PBX terms to Multiservice Switch node terms (page 100).

9 Node term P1 maps to the *diallingPlan1* attribute under *VoiceRoute/1*. See the table Mapping example: Multiservice Switch node terms to VoiceRoute component diallingPlan attribute values (page 101).

10 VoiceRoute/1 maps to VoiceNetworkingCallServer/1.

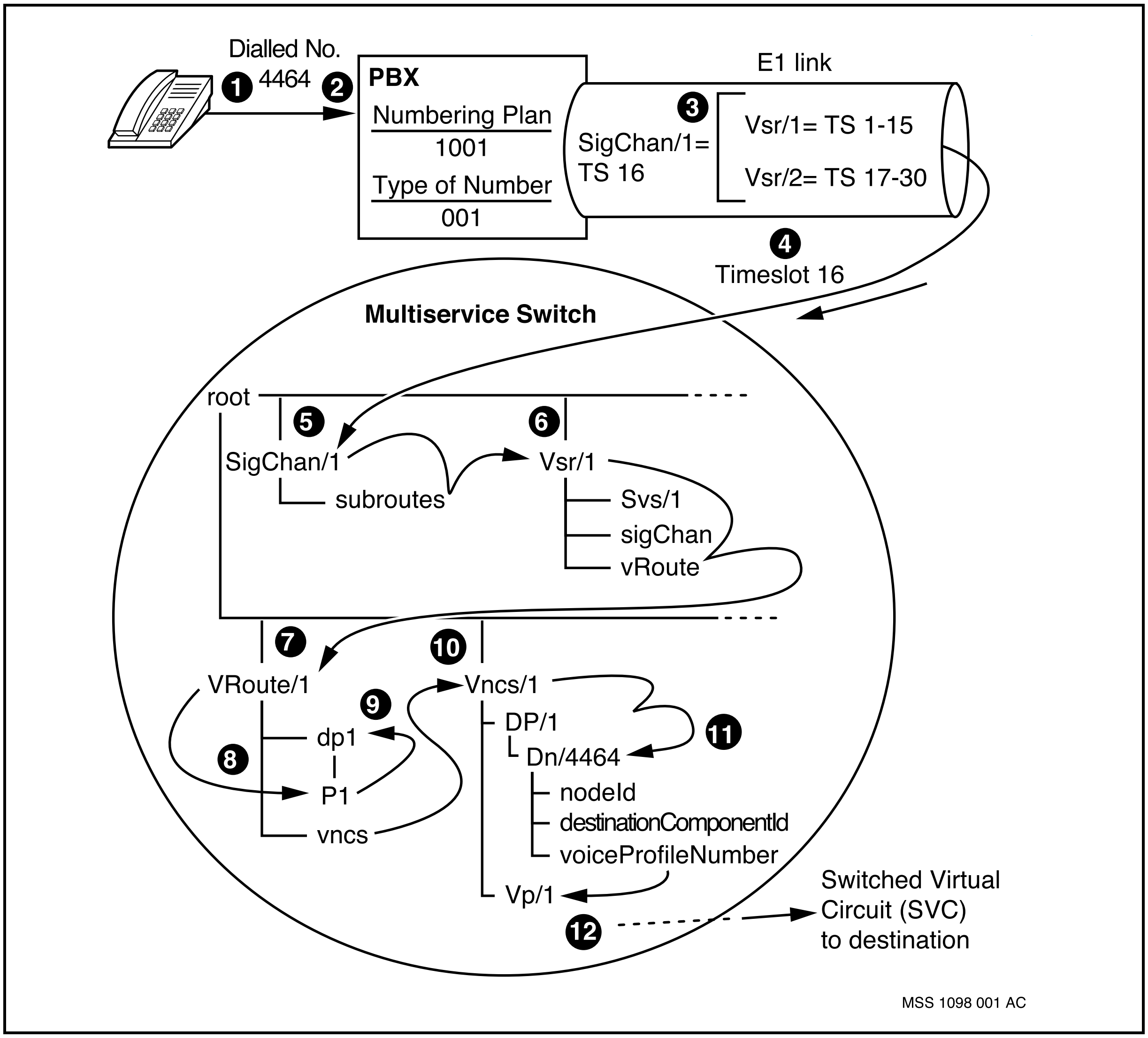
11 VoiceNetworkingCallServer/1 maps the dialed number to DP/1 (from step 9) and Dn/4464.

Dn/4464 maps the dialed number to a destination node and component ID (under the destinationNodeId and *destinationComponentId* attributes, respectively) and a voice profile (under the *voiceProfileNumber* attribute). See the table Partial numbering plan of a VoiceNetworkingCallServer component (page 96).

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

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

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



Mapping numbers—PBX to Multiservice Switch

The table Mapping PBX terms to Multiservice Switch node terms (page 100) illustrates how a PBX NPI and TON combination map to a Nortel Multiservice Switch node term for use within the subnet.

Mapping PBX terms to Multiservice Switch node terms

PBX NPI	PBX TON	Node term
0000 (Unknown)	XXX	Unknown
0001 (E.164)	000	Unknown (see 1)
(1 of 2)		

Mapping PBX terms to Multiservice Switch node terms (continued)

PBX NPI	PBX TON	Node term
	001	International
	010	National
	100	Subscriber
1001 (Private)	000	P0
	001	P1
	010	P2
	011	P3
	100	P4
	101	P5
	110	P6
	111	P7
XXXX	XXX	casUnknown (see 2)
1 PBX NPI 0001 (E.164) and TON 000 combination is identical to the PBX NPI 0000 (Unknown) and TON XXX combination. Voice Networking maps these combinations in the same way. 2 The value casUnknown applies to any number received from channel associated signaling (CAS) trunks. The value casUnknown does not correspond to a particular PBX NPI and TON combination.		
(2 of 2)		

Mapping numbers—Multiservice Switch term to VoiceRoute diallingPlan attribute

The table Mapping example: Multiservice Switch node terms to VoiceRoute component diallingPlan attribute values (page 101) provides an example of how a Nortel Multiservice Switch node term can be mapped to a *VoiceRoute* component *diallingPlan* attribute. In each *VoiceRoute* component, the TON can be mapped to a specific *diallingPlan* attribute (for example, international can be mapped to one of dp0, dp1, or dp2) through configuration. Alternately, a TON (for example, national) can be excluded from a *diallingPlan* attribute.

Mapping example: Multiservice Switch node terms to VoiceRoute component diallingPlan attribute values

Node term									
Unknown	**Inter-national**	**National**	**Sub-scriber**	**P0**	**P1**	**...**	**P7**	**casUnknown**	**dialling-Plan attribute**
X									dp0
(1 of 2)									

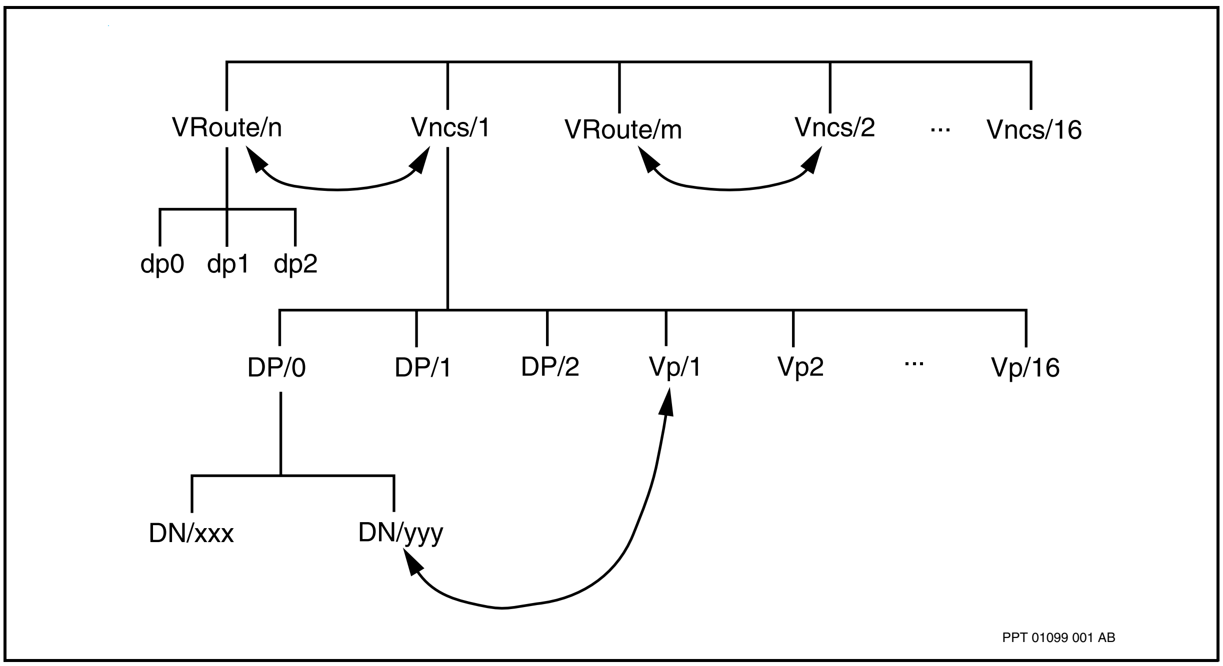
Mapping example: Multiservice Switch node terms to VoiceRoute component diallingPlan attribute values (continued)

Node term									
Unknown	**Inter-national**	**National**	**Sub-scriber**	**P0**	**P1**	**...**	**P7**	**casUnknown**	**dialling-Plan attribute**
				X	X				dp1
	X								dp2
(2 of 2)									

Mapping numbers—VoiceRoute to VoiceNetworkingCallServer DiallingPlan component

A *VoiceRoute* component instance maps to a *VoiceNetworkingCallServer* component instance. The figure *VoiceRoute to VoiceNetworkingCallServer component association (page 102)* provides an example. There can be up to sixteen VoiceNetworkingCallServer instances on each Multiservice Switch node, with each VoiceNetworkingCallServer component containing three *DiallingPlan* component instances (DP/0, DP/1, DP/2). The *DirectoryNumber* component, under a particular *VoiceNetworkingCallServer DiallingPlan* component instance, contains attributes which specify the destination and calling parameters for a dialed number.

VoiceRoute to VoiceNetworkingCallServer component association



Voice Networking using PORS

PORS establishes and maintains a transmission path to transmit voice and data traffic across a Nortel Multiservice Switch network. PORS offers two services to maintain transmission paths: failure recovery and, optionally, periodic optimization of voice paths. For more information on PORS, refer to NN10600-435 *Nortel Multiservice Switch 7400/15000/20000 Operations: Path-Oriented Routing System*.

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

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

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

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

- PORS and switched voice calls (page 103)
- Numbering plans (page 94)
- Directory numbers (page 95)
- Overlap and enbloc dialing (page 95)
- Examples of resolving dialed numbers through VNCS (page 96)

PORS and switched voice calls

The PORS application service interface supports the following:

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

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

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

Description of PORS call setup and tear-down

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

- Call setup begins at the ingress Multiservice Switch node (page 104)
- Call setup continues at the egress Multiservice Switch node (page 106)
- Call tear-down (page 108)

Call setup begins at the ingress Multiservice Switch node

Call setup begins at the ingress Nortel Multiservice Switch node as described in the following steps. The figure Message flow during call establishment begins at the ingress Multiservice Switch node (page 106) illustrates these concepts.

1. The *SignalingChannel* component on the ingress Multiservice Switch node receives the incoming dialed number (for example, 55543897) from PBX A.
2. The *SignalingChannel* component sends the formatted message (55543897) to the *VoiceSubroute* component, which in this case acts as a signal multiplexor/de-multiplexor.
3. The *VoiceSubroute* component passes the message (55543897) to the appropriate *SwitchedVoiceService* component.
4. The *SwitchedVoiceService* component initiates an address resolution request to the *VoiceNetworkingCallServer* component. In this example, the *VoiceNetworkingCallServer* component contains the following configuration information:

   VoiceNetworkingCallServer/4 DiallingPlan/1 DirectoryNumber/5554

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

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

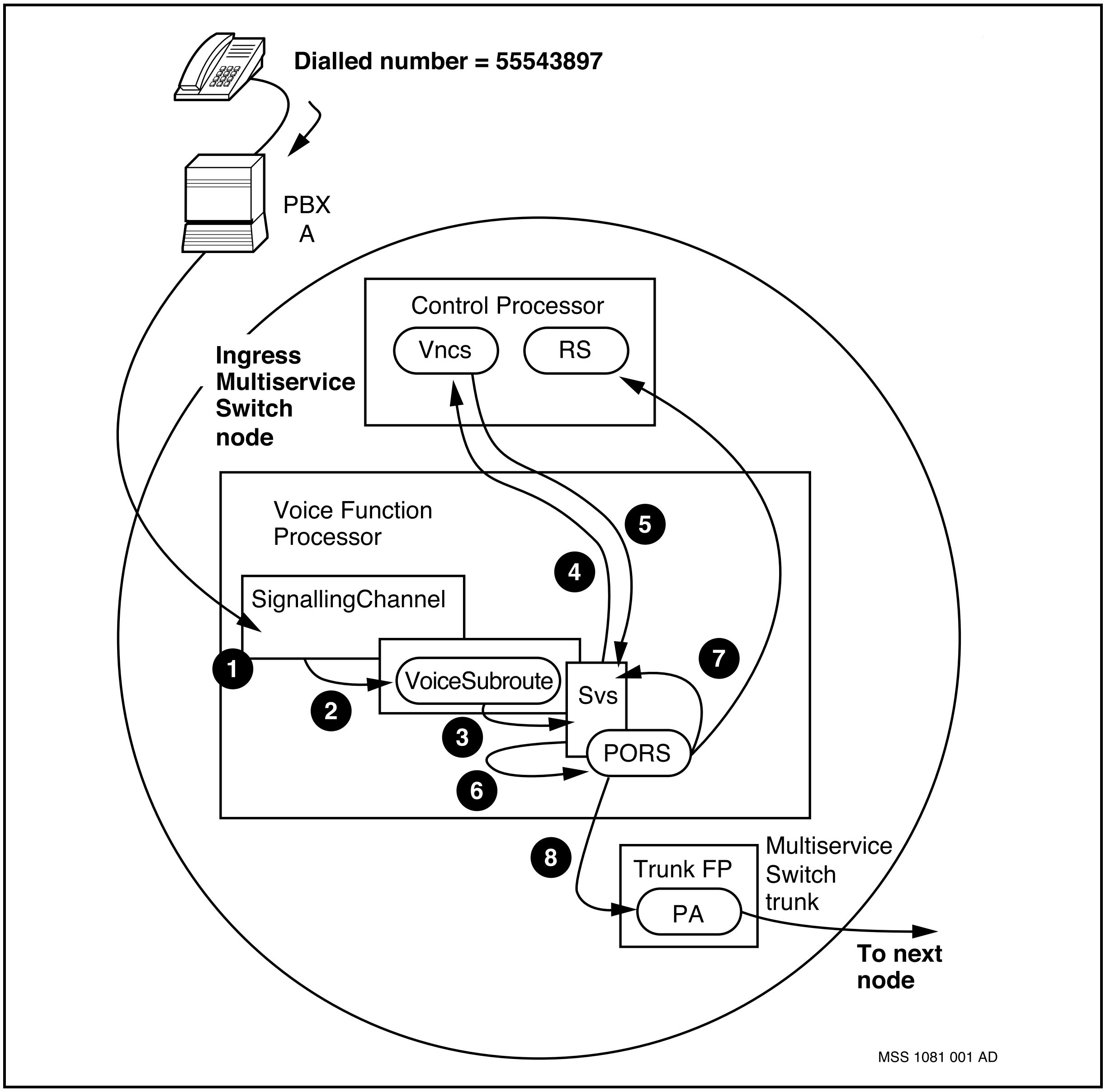
6 The SwitchedVoiceService component passes the information to the PORS.

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

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

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

Message flow during call establishment begins at the ingress Multiservice Switch node



Call setup continues at the egress Multiservice Switch node

Call setup continues at the egress Multiservice Switch node as described in the following steps. The figure Message flow during call establishment begins at the egress Multiservice Switch node (page 108) illustrates these concepts.

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

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

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

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

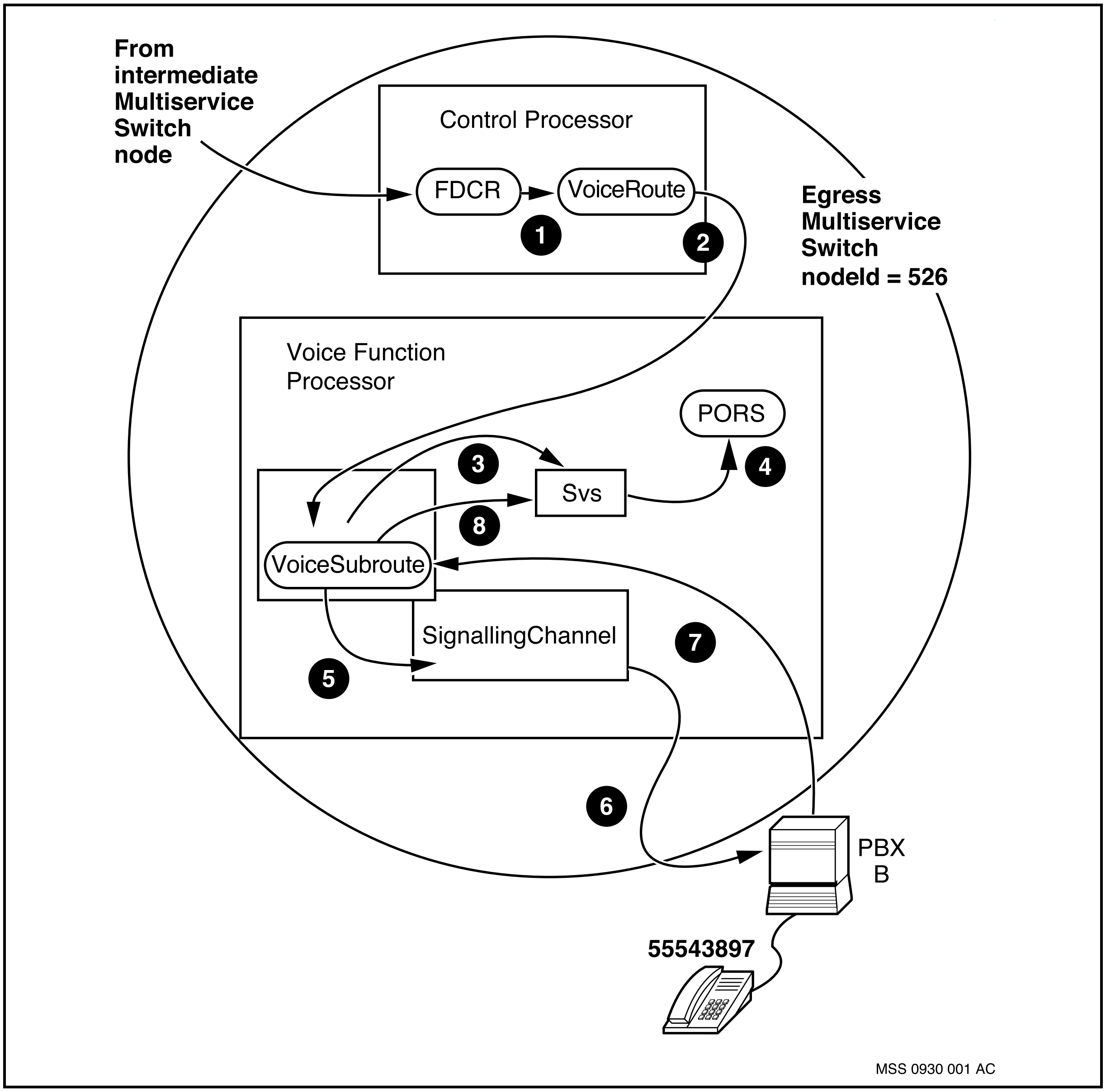
5 The *VoiceSubroute* component passes the call setup message to the *SignalingChannel* component.

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

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

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

Message flow during call establishment begins at the egress Multiservice Switch node



Call tear-down

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

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

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

3 The *SwitchedVoiceService* component forwards the disconnect message through the Multiservice Switch subnet.

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

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

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

Voice Networking based on DNAs and DPRS

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

For more information on DPRS, see NN10600-425 *Nortel Multiservice Switch 7400/15000/20000 Operations: Dynamic Packet Routing System*.

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

- DNAs and PORS SVCs (page 109)
- Call setup steps with DNA support (page 110)
- Prerequisites to implementing DNA support (page 111)
- Implement DNA support in a new installation (page 112)
- How to migrate to DNA support in an existing network (page 113)

DNAs and PORS SVCs

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

Address resolution for PORS SVCs with DNA support involves mapping a received dialed number to an internal DNA under a *DirectoryNumber* component. DNAs are associated with a numbering plan indicator (NPI) which is either X.121 or E.164 (see NN10600-405 *Nortel Multiservice Switch 7400/15000/20000 Operations: Call Server* for more information on DNAs and numbering plans).

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

Call setup steps with DNA support

The following steps describe a typical Voice Networking call setup using PORS SVCs with DNA support. The figure Conceptual example of call establishment using PORS SVCs with DNA support (page 111) illustrates this example.

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

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

---

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

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

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

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

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

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

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

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

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

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

The PORS call is now established.

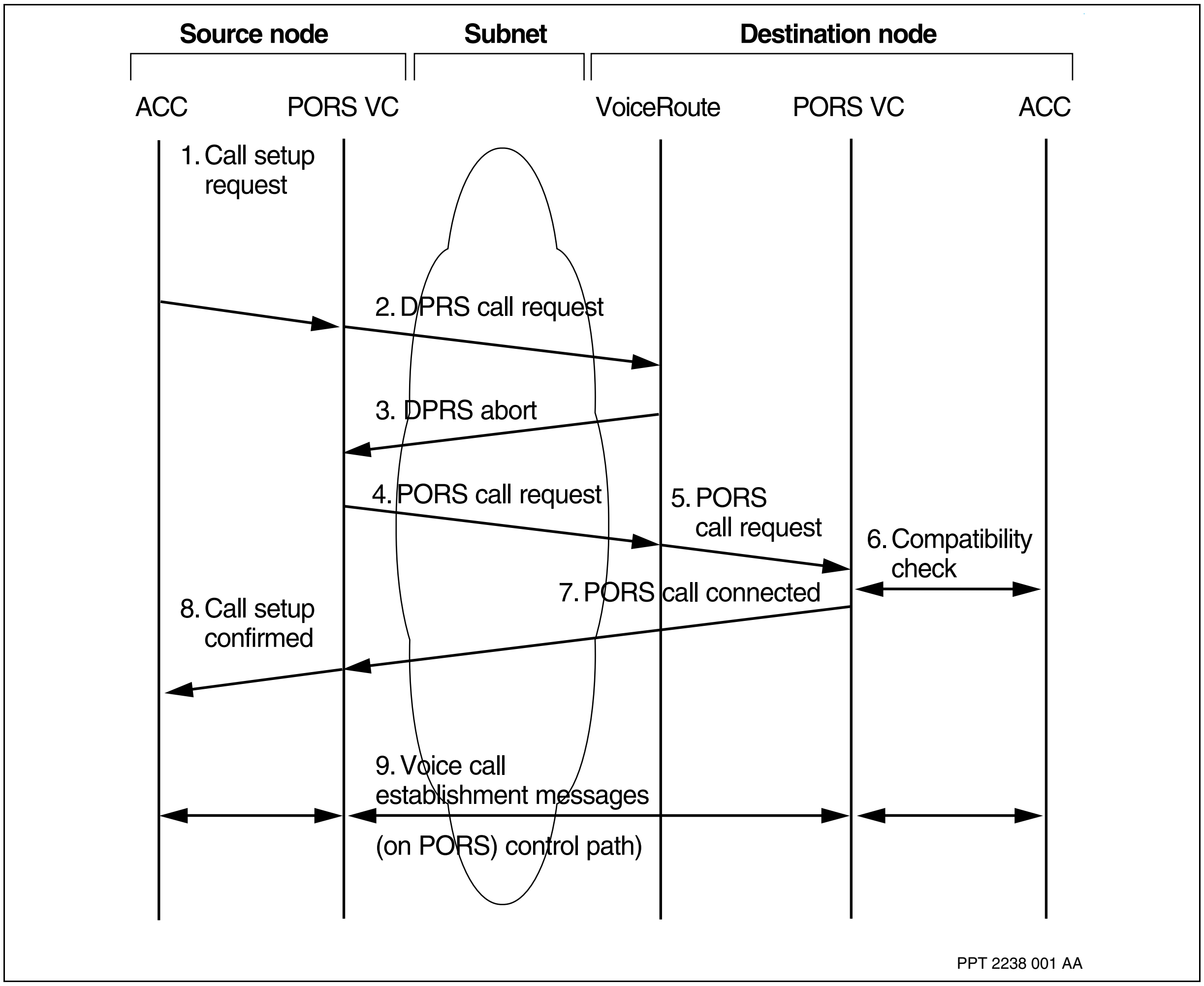
8 Call setup confirmed message from PORS VC to ACC of the source node.

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

9 Voice Networking call establishment.

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

Conceptual example of call establishment using PORS SVCs with DNA support



Prerequisites to implementing DNA support

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

- CallRouter component on Nortel Multiservice Switch
- Call Server Resource Manager (CSRM) through a DPN gateway

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

Implement DNA support in a new installation

To implement DNA support for Voice Networking, assign each *VoiceRoute* component a *DataNetworkAddress* subcomponent. See Considerations for attributes under the DataNetworkAddress component (page 112) for more information. This makes each *VoiceRoute* component visible to DPRS. The *DataNetworkAddress* subcomponent has the following two configurable attributes:

- *dataNetworkAddress*
- *numberingPlanIndicator*

Considerations for attributes under the DataNetworkAddress component

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

- The NPI/DNA pair has to be unique in the entire network. Each *VoiceRoute* component registers its NPI/DNA pair at its node's FDCR.
- The *VoiceRoute* component's NPI/DNA pair is used to configure *VoiceNetworkingCallServer DiallingPlan DirectoryNumber* component entries which terminate on the VoiceRoute component.
- Configure DPRS to support NPI/DNA pairs for all *VoiceRoute* components in the network by configuring DNA prefixes on the *CallRouter* component. This implies the following:
  - To keep the number of entries in the *CallRouter* component's table low, configure all *VoiceRoute* components on a single node with the same NPI and DNA prefix. This creates only one entry for each node in the *CallRouter* component's table.
  - To keep DNA prefixes in the *CallRouter* component's table short, the *VoiceRoute* components on different nodes require different DNA prefixes or NPIs, or both.
  - If a *VoiceRoute* component is added with a *DataNetworkAddress* component that does not follow the NPI or DNA prefix, or both, of the existing *VoiceRoute* components on the node, ensure that a DNA prefix exists in the *CallRouter* component's table to properly route calls to the added *VoiceRoute* component's node.

For more information on the call router, refer to NN10600-405 *Nortel Multiservice Switch 7400/15000/20000 Operations: Call Server*.

How to migrate to DNA support in an existing network

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

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

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

  Make sure that the *Routing DpnAddressPlan* component exists and is configured properly. If the *Routing DpnAddressPlan* component does not exist, add the dpnRouting feature to the *featureList* attribute of the *Software LogicalProcessorType* component instance for the control processor (CP). For more information on configuring the Routing DpnAddressPlan component, refer to NN10600-450 *Nortel Multiservice Switch 7400: Operations: DPN-100 Interworking*.

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

3 Set the *VoiceRoute DataNetworkAddress* component's attributes.

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

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

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

5 Add DNAs to the *VoiceNetworkingCallServer* component's database. However, do not remove the destination node IDs and destination component IDs at this time. See Considerations for attributes under the DataNetworkAddress component (page 112).

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

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

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

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

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

Call redirection: when primary destinations are unavailable

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

Call redirection is formally known as Voice Networking call hunting. For information on this capability, see Voice Networking call hunting (page 119).

Digit manipulation

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

Information on digit manipulation is organized into the following sections:

Summary of features

Nortel Multiservice Switch node's digit manipulation supports the following capabilities:

- deletion at the ingress node of up to 16 leading digits from the dialed number digit string received from the originating PBX
- insertion at the ingress node of a digit sequence of up to 16 digits at the beginning of the dialed number digit string received from the originating PBX
- deletion at the egress node of up to 16 leading digits from the ingress called number digit string to be sent to the destination PBX
- insertion at the egress node of a digit sequence of length up to 16 digits at the beginning of the ingress called number digit string to be sent to the destination PBX
- configuration at the voice route level, which allows the service provider to
  - — apply the same digit manipulation rules to all subroutes
  - — turn digit manipulation off for any voice route without affecting other voice routes
- configuration of call routing to use either dialed digits or manipulated digits

- logging for ingress accounting records, to record the dialed number (as received from the PBX) as the called number
- logging for egress accounting records, to record the called number as the called number received from the ingress side prior to any egress digit manipulation
- support for both en bloc and overlap dialing types

Supported digit types

Nortel Multiservice Switch nodes digit manipulation supports the following digits types:

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

Through digit manipulation, the Multiservice Switch node can delete any of these digit types from the called number. However, the switch can insert only digit types 0 through 9, octothorpe, and asterisk into the called number. If digit manipulation is configured to insert an octothorpe or asterisk, then routing is based on the dialed digits (not the modified digits) since only the digit type 0 through 9 is supported by VNCS. For example, if the string that digit manipulation inserts is

613*

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

Empty digit strings at ingress

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

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

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

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

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

Empty digit strings at egress

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

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

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

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

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

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

Attempt call completion

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

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

Release the call

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

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

Accounting impacts of digit manipulation

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

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

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

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

Voice Networking call hunting

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

To configure call hunting, a primary hunt group server with voice routes as hunt group members is configured on a node. For redundancy, a backup hunt group server can also be configured on another node in the network. The backup hunt group server supports call hunting when the primary hunt group server is out of service. See NN10600-415 *Nortel Multiservice Switch 7400/15000/20000 Operations: Hunt Group Server*.

Overview of call hunting routing process

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

Calls destined for a Nortel Multiservice Switch hunt group server DNA are routed to the hunt group server, which in turn chooses an eligible hunt group member DNA from the list of members. In this configuration, the members are voice route DNAs. If the voice route is available, the network forwards the call. Otherwise, the hunt group server selects the next eligible member in the list. The choice of hunt group member depends on the search mode that you configure for the server and the availability of the voice route. Availability of each voice route is signaled to the hunt group server through the availability message packet (AMP).

This process continues until a voice route can carry the call or the member list is exhausted. If the member list is exhausted and the call is not completed, the network rejects the call if a backup hunt group server is not available. See NN10600-415 *Nortel Multiservice Switch 7400/15000/20000 Operations: Hunt Group Server* for further explanation of search modes and using the backup hunt group server.

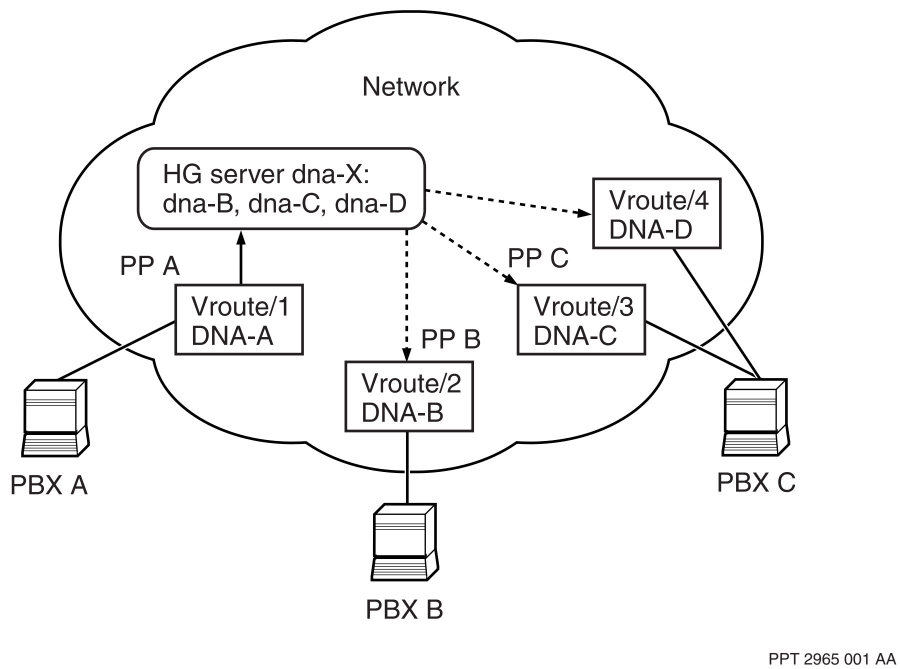
Call hunting supports call routing under the CCS and CAS protocols.

Example of call setup

The figure Example of a Voice Networking call hunting (page 120) provides an example of call hunting through a hunt group server.

A call request comes from PBX A to a Nortel Multiservice Switch node called PP A. The VNCS for PPA resolves the destination address which, in this example, is the hunt group server DNA X. A call request is sent to the hunt group server based on the DNA. The hunt group server, upon receiving the call request, will select one of it's members (dna-B, dna-C, dna-D), based on the server's search mode (see NN10600-415 *Nortel Multiservice Switch 7400/15000/20000 Operations: Hunt Group Server*), and forward the call request to that member. If the selected member's availability is sufficient to handle the call, the call will proceed as a normal Voice Networking call. If the selected member's availability is not sufficient to handle the call, the call will be redirected back to the hunt group server. On a redirected call request, the hunt group server selects the next hunt group member.

Example of a Voice Networking call hunting



Implementation and configuration considerations

At least one Nortel Multiservice Switch hunt group server must be present in the network to support call hunting. See NN10600-415 *Nortel Multiservice Switch 7400/15000/20000 Operations: Hunt Group Server* for descriptions and configuration information for hunt group servers.

If the primary hunt group server cannot be reached or is out of service, the call can route through backup hunt group server. The backup hunt group server must be configured in the call redirection server (CRS). See NN10600-410 *Nortel Multiservice Switch 7400/15000/20000 Operations: Call Redirection Server* for descriptions and configuration information for CRS.

The following points provide an overview to implementing call hunting:

1. Set up a primary hunt group server, with the DNAs of voice routes as hunt group members. This configuration enables call hunting among a set of destination voice routes to complete a given call. Also, specify the type of hunting algorithm to use: start from zero, rotary, or most available.
2. Optionally, set up a backup hunt group server with a member set that is identical to that on the primary server.
3. Specify the DNA address of the primary hunt group server in the *dataNetworkAddress* attribute under the *Vncs DiallingPlan DirectoryNumber* component.
4. Add a *HuntGroupMember* component under each *VoiceRoute Dna* component. Optionally, change the value of the *usageDeltaUpdateThreshold* attribute, which defines the threshold for availability of the member to both the primary and backup hunt group servers. Specify the DNS of the primary hunt group server; optionally, specify the DNS of the backup hunt group server. This DNS entry defines the location of the servers to which the voice route sends the AMP.

For configuration details and procedures, see Configuring call hunting through a hunt group server (page 61).

Voice Networking call server configuration

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

To prevent the clipping of voice traffic on MVP-E FPs when the negotiated value of the *silenceSuppression* attribute is on, ensure that the *VoiceRoute* Interface component's *speechHangoverTime* attribute is set appropriately. You can also specify a cap on the amount of background or comfort noise to generate when silence suppression is operational. See Voice routes (page 129) for more information.

End-to-end negotiation determines whether to support FIS. FIS operates on MVP-E FPs when the negotiated modem/fax encoding choice is g711G726 or g726. To prevent the clipping of portions of facsimile transmissions on MVP-E FPs when the negotiated value of the *faxIdleSuppressionG711G726* attribute is on, ensure that the *faxHangoverTimeG711G726* attribute under the *VoiceRoute Interface* component is set appropriately. See Voice routes (page 129) for more information.

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

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

Voice networking call server components

You configure the voice networking call server by configuring the *VoiceNetworkingCallServer*, *VoiceProfile*, *DiallingPlan*, and *DirectoryNumber* components. The *VoiceNetworkingCallServer* component provides address resolution information to the PORS during switched voice call establishment by means of the *DialingPlan* and *DirectoryNumber* components. The *DirectoryNumber* component contains attributes to define PORS-based routing using destination node and component identification or PORS-based routing with DNA support. The *VoiceNetworkingCallServer* component gathers audio handling parameters from the *VoiceProfile* component associated with particular *DialingPlan* and *DirectoryNumber* component instance combinations. Before proceeding with configuration the attributes under the *VoiceProfile* component, see Voice, modem, and facsimile encoding choices and rates (page 123).

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

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

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

Voice, modem, and facsimile encoding choices and rates

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

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

- Considerations for voice, modem, and facsimile traffic configuration (page 123)
- VoiceEncodingChoice and voiceRates attributes (page 124)
- ModemFaxEncodingChoice and modemFaxRates attributes (page 125)
- Defining traffic handling options (page 128)

Considerations for voice, modem, and facsimile traffic configuration

To successfully establish a call,

- each FP must support a configured encoding choice and certain rates between the minimum and maximum inclusive
- a corresponding encoding feature, for certain encoding types, must be configured under the *featureList* attribute

In some cases, an MVP-E FP supports an encoding choice but modifies certain configured rate values. For example, MVP-E FPs support dynamic up- and down-speeding of audio traffic for dealing with congestion in the network. However, MVP-E FPs do not support all of the available ITU-T G.726 ADPCM voice and modem/fax encoding rates. The table Relationship between FPs, encoding choices, and supported rates (page 124) lists the available voice and modem/fax encoding choice and FP combinations, and the rates each combination supports.

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

Relationship between FPs, encoding choices, and supported rates

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

VoiceEncodingChoice and voiceRates attributes

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

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

  It is possible to provision the *voiceEncodingChoice* attribute with an encoding choice which requires a corresponding feature and not provision the *Software* component's *featureList* attribute. Further, the system does not notify you of this omission when you check your configuration. Only during end-to-end negotiation are encoding feature omissions discovered. Therefore, to avoid potential call setup problems, you must provision the encoding choice's corresponding feature. See the table Default and provisionable voice encoding rates for each voice encoding choice (page 125).

The table Default and provisionable voice encoding rates for each voice encoding choice (page 125) contains the default and provisionable values for the minimum and maximum rates of the *voiceRates* attribute corresponding to each of the *voiceEncodingChoice* attribute's values. This table also lists whether an encoding choice requires you to provision a corresponding feature. As the table shows, only g711G726 encoding allows you to set the minimum and maximum rates to a value other than the default value. Thus, setting encoding choices for voice traffic requires very little configuration.

Default and provisionable voice encoding rates for each voice encoding choice

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

ModemFaxEncodingChoice and modemFaxRates attributes

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

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

Attention: It is possible to provision the *modemFaxEncodingChoice* attribute with an encoding choice which requires a corresponding feature and not provision the *Software* component's *featureList* attribute. Further, the system does not notify you of this omission when you check your configuration. Only during end-to-end negotiation are encoding feature omissions discovered. Therefore, to avoid potential call setup problems, you must provision the encoding choice's corresponding feature. See the table Default and provisionable modem/fax encoding rates for each modem/fax encoding choice (page 126).

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

The table Default and provisionable modem/fax encoding rates for each modem/fax encoding choice (page 126) contains the default and provisionable values for the minimum and maximum rates of the *modemFaxRates* attribute corresponding to each of the *modemFaxEncodingChoice* attribute's values. The table also lists whether an encoding choice requires you to provision a corresponding feature. As the table shows, only the value g711G726 allows you to set both the minimum and maximum rates to a value other than the default value. The value relay only allows you to set the maximum rate to a value other than the default value.

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

- Encoding modem/fax and fax traffic (page 127)
- Considerations for Fax relay configuration (page 128)
- Using the value none (page 128)

Default and provisionable modem/fax encoding rates for each modem/fax encoding choice

modemFax-EncodingChoice attribute values	Feature required	Default modemFaxRates attribute rates (in kbit/s)	Provisionable modemFaxRates attribute rates (in kbit/s)
g711Only	N/A	min = 64 max = 64	min = 64 max = 64
g711G726 (see 1)	N/A	min = 32 max = 64	min = 16, 24, 32, 64 max = 16, 24, 32, 64
(1 of 2)			

Default and provisionable modem/fax encoding rates for each modem/fax encoding choice

modemFax-EncodingChoice attribute values	Feature required	Default modemFaxRates attribute rates (in kbit/s)	Provisionable modemFaxRates attribute rates (in kbit/s)
relay (see 1)	faxRelay	min = 0.0 max = 14.4	min = 0.0 max = 2.4, 4.8, 7.2, 9.6, 12.0, 14.4
none (see 2)	N/A	N/A	N/A
1 MVP-E FPs support g711G726 as a modem/fax encoding choice, but only at the following rates: 32 and 64 kbit/s. 2 The minimum fax relay rate of 0.0 kbit/s is fixed. The encoding choice none signifies the end of the encoding list. See Using the value none (page 128) for more information on setting the three elements of the modemFaxEncodingChoice attribute to none, none, and none.			
(2 of 2)			

Encoding modem/fax and fax traffic

Once end-to-end negotiation determines the preferred voice choice to use on voice traffic, the two highest remaining preferred modem/fax choices combine to determine the encoding of modem/fax and fax traffic. The table Modem/Fax encoding choice combinations and traffic type matrix (page 127) describes all possible modem/fax choice combinations and the negotiated encoding choice for each traffic type.

Modem/Fax encoding choice combinations and traffic type matrix

Modem/Fax choices remaining (after end-to-end negotiation)		Traffic type and possible negotiated encoding values	
First choice	**Second choice**	**Modem/Fax traffic**	**Fax traffic**
relay	none	v29V27Relay or v17V29V27Relay (see 1)	v29V27Relay or v17V29V27Relay
relay	g711Only	g711	v29V27Relay or v17V29V27Relay
relay	g711G726	g711G726, g726	v29V27Relay or v17V29V27Relay
g711Only	--- (see 2)	g711	g711
g711G726	--- (see 2)	g711G726, g726	g711G726, g726
(1 of 2)			

Modem/Fax encoding choice combinations and traffic type matrix (continued)

Modem/Fax choices remaining (after end-to-end negotiation)		Traffic type and possible negotiated encoding values	
First choice	**Second choice**	**Modem/Fax traffic**	**Fax traffic**
none	none (see 3)	See Using the value none (page 128)	
1 The use of relay for modem/fax traffic means that modem calls are not supported (see Considerations for Fax relay configuration (page 128)). 2 If either g711Only or g711G726 is the first modem/fax encoding choice, then it is used for both modem/Fax and Fax traffic. Voice Networking ignores the second modem/Fax choice. 3 End-to-end negotiation can result in the modification of the remaining modem/Fax choices to none and none (different from setting the *modemFaxEncodingChoice* attribute to none and none). In this situation—having no remaining supported modem/Fax encoding choices—the call is rejected.			
(2 of 2)			

Considerations for Fax relay configuration

Fax relay encoding applies only to fax traffic. If you need to support modem calls and want to use Fax relay encoding, you must provision the *modemFaxEncodingChoice* attribute with the value relay and a modem/Fax encoding choice for modem/Fax traffic. Use the information in the table Modem/Fax encoding choice combinations and traffic type matrix (page 127) as a guideline.

Using the value none

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

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

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

Defining traffic handling options

When configuring voice and modem/fax encoding choices, you can simplify the configuration process by categorizing and defining how to handle voice, modem/Fax, and Fax traffic. The table Available Voice Networking traffic handling options (page 129) describes the categorization options available and the FPs that support each option. If necessary, see also the table Modem/ Fax encoding choice combinations and traffic type matrix (page 127) for more information.

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

Available Voice Networking traffic handling options

Option	FP type	Configuration details
Support all call types and encode fax traffic using Fax relay	MVP-E only	set Vncs/# Vp/# modemFaxEncodingChoice first relay set Vncs/# Vp/# modemFaxEncodingChoice second <g711Only or g711G726>
Support all call types and encode modem/Fax and Fax traffic using g711G726 or g711Only	MVP-E	set Vncs/# Vp/# modemFaxEncodingChoice first <g711Only or g711G726>
Support only voice and facsimile calls and encode modem/Fax and Fax traffic using relay	MVP-E only	set Vncs/# Vp/# modemFaxEncodingChoice first relay set Vncs/# Vp/# modemFaxEncodingChoice second none set Vncs/# Vp/# modemFaxEncodingChoice third none
Treat all traffic as voice traffic (see Using the value none (page 128) for more information)	MVP-E	set Vncs/# Vp/# modemFaxEncodingChoice first none set Vncs/# Vp/# modemFaxEncodingChoice second none set Vncs/# Vp/# modemFaxEncodingChoice third none

Voice routes

The *VoiceRoute* component controls from 1 to 16 *VoiceSubroute* components.

Configuring the voice route and DNA

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

Each *diallingPlan* attribute (dp0, dp1, and dp2) corresponds to a like-numbered *DiallingPlan* component (DP/0, DP/1, and DP/2) under the *VoiceNetworkingCallServer* component. You can thus organize numbers according to type or customer group using the *diallingPlan* attributes and define how to handle each group using the *DirectoryNumber* subcomponents under each *DiallingPlan* component.

In some cases, the signaling protocol defines how to configure the *diallingPlan* attributes. For example, the MCDN signaling protocol supports specific types of numbers according to the 3 supported Meridian 1 numbering plans: Unknown, E.164, and Private. See the table Supported MCDN numbering plan and type of number combinations (page 130)). For MCDN, you map a particular NPI and TON combination—corresponding to a particular node value—under a specific *diallingPlan* attribute instance: dp0, dp1, or dp2.

As shown in the table Supported MCDN numbering plan and type of number combinations (page 130), coordinated dialling plan (CDP) numbers map to the *diallingPlan0* attribute. Therefore, you assign and organize dialed numbers that belong to the CDP type to *DirectoryNumber* subcomponents under the *DiallingPlan*/0 component of the appropriate *VoiceNetworkingCallServer* component instance.

Supported MCDN numbering plan and type of number combinations

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

Configuring the voice route interface

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

Tandem Pass Through

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

For TPT to operate,

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

TPT also introduces a certain amount of delay into the end-to-end connection. You can compensate for the delay by adjusting the jitter buffer on the egress node. You adjust the jitter buffer by configuring the *insertedOutputDelay* attribute under the *VoiceProfile* component (see Voice Networking call server configuration (page 121)).

For information on supported TPT configurations, refer to NN10600-750 *Nortel Multiservice Switch 7400 Operations: Voice Transport*.

Voice subroute

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

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

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

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

Compliance with standards—Voice Networking signaling protocols

This appendix contains compliance information for both the CCS- and CAS-based signaling protocols supported by Voice Networking.

Navigation

Compliance statement for ETSI QSIG

Voice Networking's implementation of the ETSI QSIG signaling protocol complies, in general, with the ETSI QSIG standards listed in the table Voice Networking's ETSI QSIG compliance with standards (page 132).

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

Voice Networking's ETSI QSIG compliance with standards

Standard	Details
ETS 300 172, 3rd Edition	Basic call (call related APDU transport, call related notification transport, connection oriented call independent APDU transport)
ETS 300 239, 3rd Edition	Generic functional procedures
ETS 300 172, Annex ZA (1995)	Segmentation and reassembly
ETS 300 239, 2nd Edition	Transport of manufacturer specific information
(1 of 2)	

Voice Networking's ETSI QSIG compliance with standards (continued)

Standard	Details
ETS 300 238, 2nd Edition	Name identification
EN 301 048, V1.1.1	Transit counter
ETS 300 261, 2nd Edition	Call transfer
ETS 300 257, 2nd Edition	Call diversion
ETS 300 259, 2nd Edition	Path replacement
ETS 300 362, 2nd Edition	Call offer
ETS 300 366, 2nd Edition	Call completion
ETS 300 239, 2nd Edition	Called/Busy number identification
ETS 301 360/255, 2nd Edition	Message waiting indication
ETS 300 257, ETS 300 239, ETS 301 255 (2nd Edition)	Centralized voice mail
(2 of 2)	

Compliance statement for NIS A211-1

Voice Networking's implementation of the NIS A211-1 signaling protocol complies, in general, with the standards listed in the table Voice Networking's NIS A211-1 compliance with standards (page 133). This includes support for:

- basic call using 64 kbit/s bearer channels
- facility associated signaling (23 B + D)
- listed services applying to call associated signaling
- Layer 1, 2 and 3: maintenance messages, signaling and line requirements for the preceding items

Voice Networking's NIS A211-1 compliance with standards

Standard	Details
NIS A211-1, Standard Release 6, (1994-03)	Basic call control, link and channel maintenance messaging, called and calling number transport
NIS A211-1, Standard Release 6, (1994-03)	56 kbit/s clear data call transport
NIS A211-1, Standard Release 6, (1994-03)	Calling, connected party number display (presentation and restriction)
NIS A211-1, Standard Release 6, (1994-03)	Network name display (presentation and restriction)
NIS A211-1, Standard Release 6, (1994-03)	Network redirection and reason
NIS A211-1, Standard Release 6, (1994-03)	Special number services
(1 of 2)	

Voice Networking's NIS A211-1 compliance with standards (continued)

Standard	Details
NIS A211-1, Standard Release 6, (1994-03)	Federal Communications Commission (FCC) Equal Access
NIS A211-1, Standard Release 6, (1994-03)	Integrated services access
(2 of 2)	

Compliance statement for MCDN

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

- basic call support, as defined in MCDN Release 1 (1988), TR-88-5003-R/M/S, TR-87-0041-R/M/S, TR-88-5505-S
- called, calling, and connected number support, as defined in MCDN Release 1 (1988), TR-87-0045-R/S, TR-88-0047-R/M/S
- subaddressing support
- channel negotiation and overlap dialing support
- maintenance messaging support
- connection-oriented feature transport support (part of NAS development)
- connectionless feature transport support, as defined in MCDN Release 1 (1988), PS-87-0010,and including network ring again, as defined in TR-87-0041-R/M/S
- redirection information support, as defined in TR-88-5004-R/M/S

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

See the table Supported MCDN basic call and supplementary services (page 83) for the MCDN basic call and supplementary capabilities supported by Voice Networking.

Compliance statement for Euro-ISDN

Voice NetworkSing's implementation of the Euro-ISDN signaling protocol complies, in general, with the ETSI and ITU-T standards described in the table Voice Networking's Euro-ISDN compliance with standards (page 135).

Voice Networking's Euro-ISDN compliance with standards

Standard	Details
ITU-T Q.931, (1993-03)	Digital subscriber system number one (DSS1); ISDN UNI Layer 3 specification for basic call control
ETSI EN 300 403-1, V.1.2.2 (1998-04) **Attention:** Includes support of ETS 300 102-1 (1990).	ISDN; DSS1; Signaling network layer for circuit-mode basic call control
ETSI EN 300 196-1, V.1.2.2 (1998-04)	ISDN; DSS1; Generic functional protocol for the support of supplementary services
ETSI ETS 300 138-1, Second edition (1997-05)	ISDN; DSS1; Closed user group (CUG) supplementary service
ETSI ETS 300 092-1, (1992-03)	ISDN; DSS1; Calling line identification presentation (CLIP) supplementary service
ETSI ETS 300 093-1, (1992-03)	ISDN; DSS1; Calling line identification restriction (CLIR) supplementary service
ETSI ETS 300 064-1, Second edition (1996-09)	ISDN; DSS1; Direct dialing in (DDI) supplementary service
ETSI ETS 300 061-1, (1991-10)	ISDN; DSS1; Subaddress (SUB) supplementary service
ETSI ETS 300 097-1, (1992-05)	ISDN; DSS1; Connected line identification presentation (COLP) supplementary service
ETSI ETS 300 098-1, (1992-05)	ISDN; DSS1; Connected line identification restriction (COLR) supplementary service
ETSI ETS 300 286-1, (1996-02)	ISDN; DSS1; User-to-user signaling (UUS) supplementary service
ETSI ETS 300 130-1, (1992-05)	ISDN; DSS1; Malicious call identification (MCID) supplementary service
ETSI ETS 300 207-1, (1994-12)	ISDN; DSS1; Diversion supplementary services, including call forwarding busy (CFB), call forwarding no reply (CFNR), call forwarding unconditional (CFU), and call deflection (CD)
ETSI ETS 300 182-1, (1993-04)	ISDN; DSS1; Advice of charge (includes AOC-D and AOC-E) supplementary service
ETSI ETS 300 185-1, (1993-04)	ISDN; DSS1; Conference call, add-on (CONF) supplementary service
ETSI ETS 300 359-1, (1995-11)	ISDN; DSS1; Completion of calls to busy subscriber (CCBS) supplementary service
(1 of 2)	

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

Standard	Details
ETSI EN 301 065-1, V.1.1.1 (1997-12)	ISDN; DSS1; Completion of calls on no reply (CCNR) supplementary service
ETSI ETS 300 210-1, (1996-02)	ISDN; DSS1; Freephone (FPH) supplementary service
ETSI ETS 300 745-1, (1997-07)	ISDN; DSS1; Message waiting indicator (MWI) supplementary service
ETSI EG 201 189-1, V.1.2.1 (1997-07)	ISDN; DSS1; Master list of codepoints and operation values
(2 of 2)	

Compliance statements for CAS

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

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

Network migration considerations

This appendix describes how to migrate, or upgrade, a Nortel Multiservice Switch Voice Transport permanent logical connection or PLC-based network to a Multiservice Switch Voice Networking switched virtual circuit or SVC-based network.

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

Navigation

Upgrade options

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

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

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

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

Preliminary considerations

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

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

Attention: Throughout this chapter, the term “PLC” describes a group of PLC connections (rather than a single PLC connection) between two PBXs across a Multiservice Switch subnet.

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

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

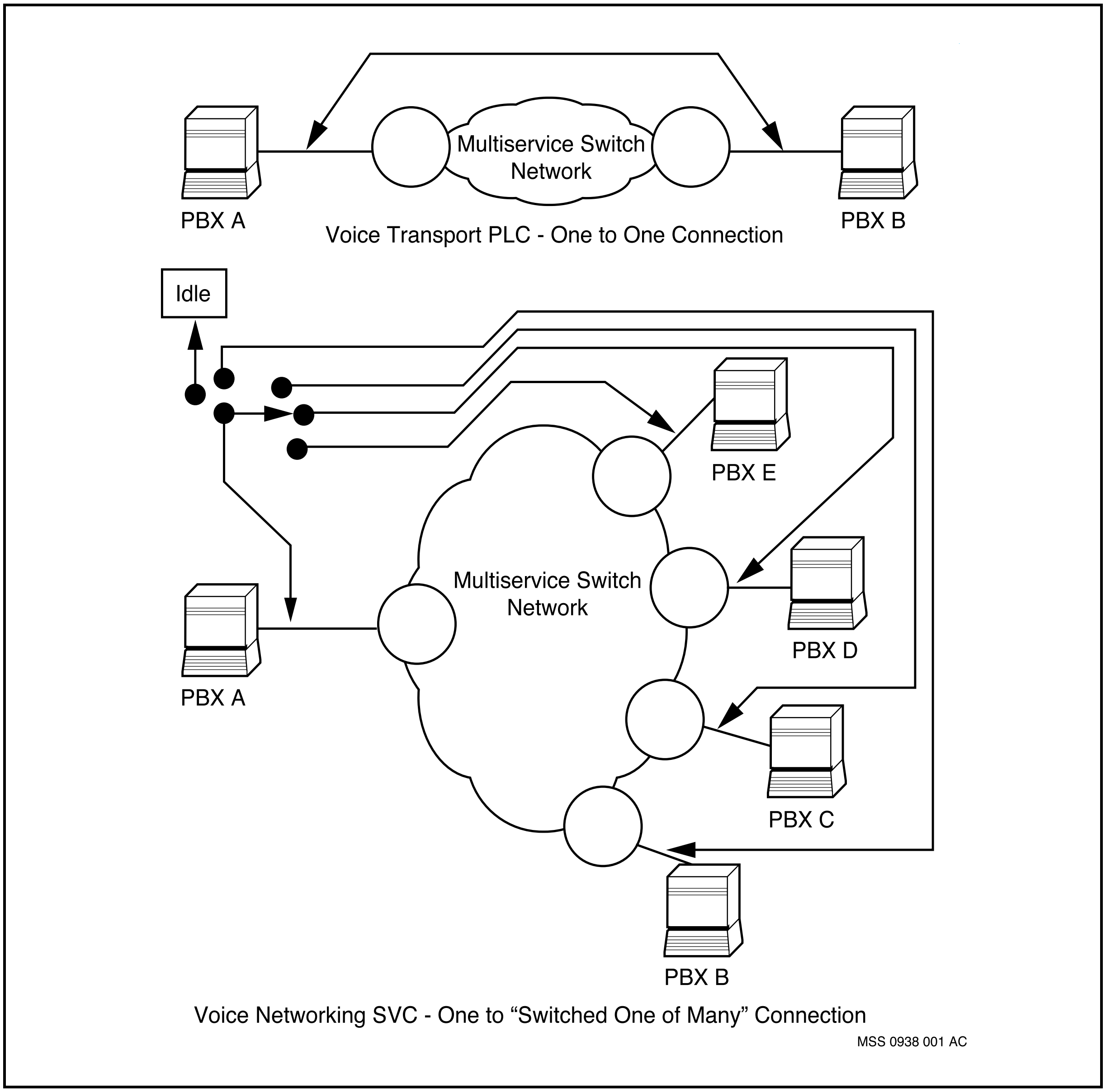
  When PBX A on node A terminates a call to PBX B on node B, clearing the call has no effect on the interconnecting link; the link remains in place and its bandwidth reservation remains in effect.

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

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

The figure Voice Transport PLC versus Voice Networking SVC (page 139) illustrates the difference between a Voice Transport PLC-based call and a Voice Networking switched-call. A Voice Transport PLC is a fixed point to point connection. Whereas a Voice Networking SVC is a point to multipoint connection; the end points are selected as part of the setup, and disassociated after clearing.

Voice Transport PLC versus Voice Networking SVC



Mixing PLCs and VoiceNetworking

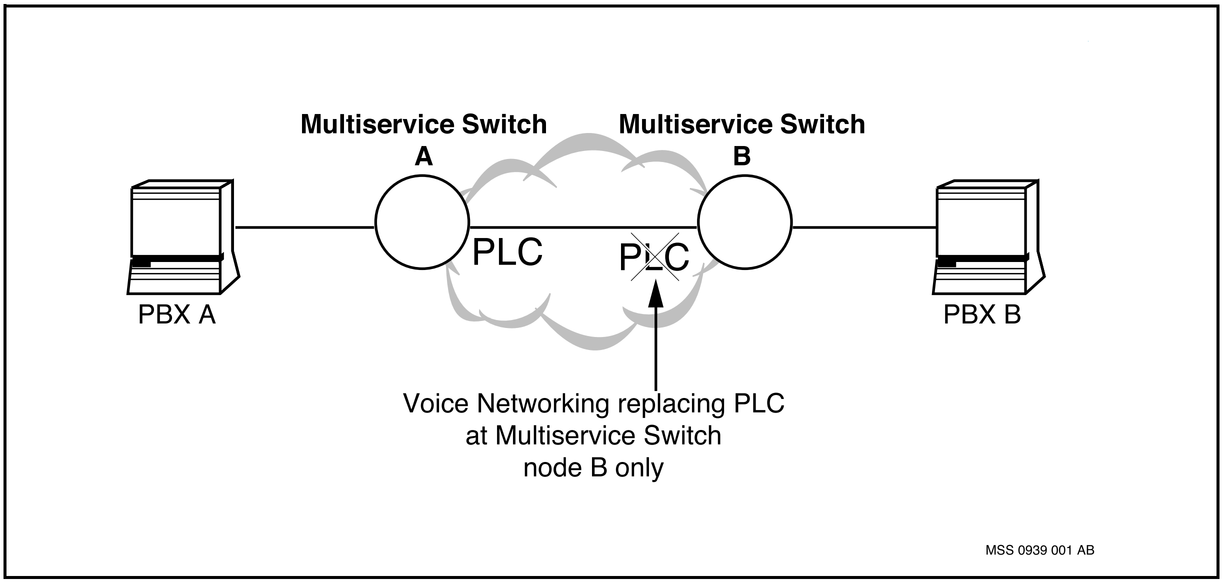
Voice Networking is incompatible with other Nortel Multiservice Switch voice services on a PBX link by PBX link basis.

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

If a PLC-based connection exists between node A and node B, the PLC connection cannot be replaced at node B alone. See the figure Example of incompatible configurations (page 140). The PLC connection at node A must also be replaced or no calls can traverse the subnet, since the PLC at node A neither analyses nor provides digits.

In the figure Example of incompatible configurations (page 140), Voice Networking at node B requires digit information to complete connections for the B-channel.

Example of incompatible configurations

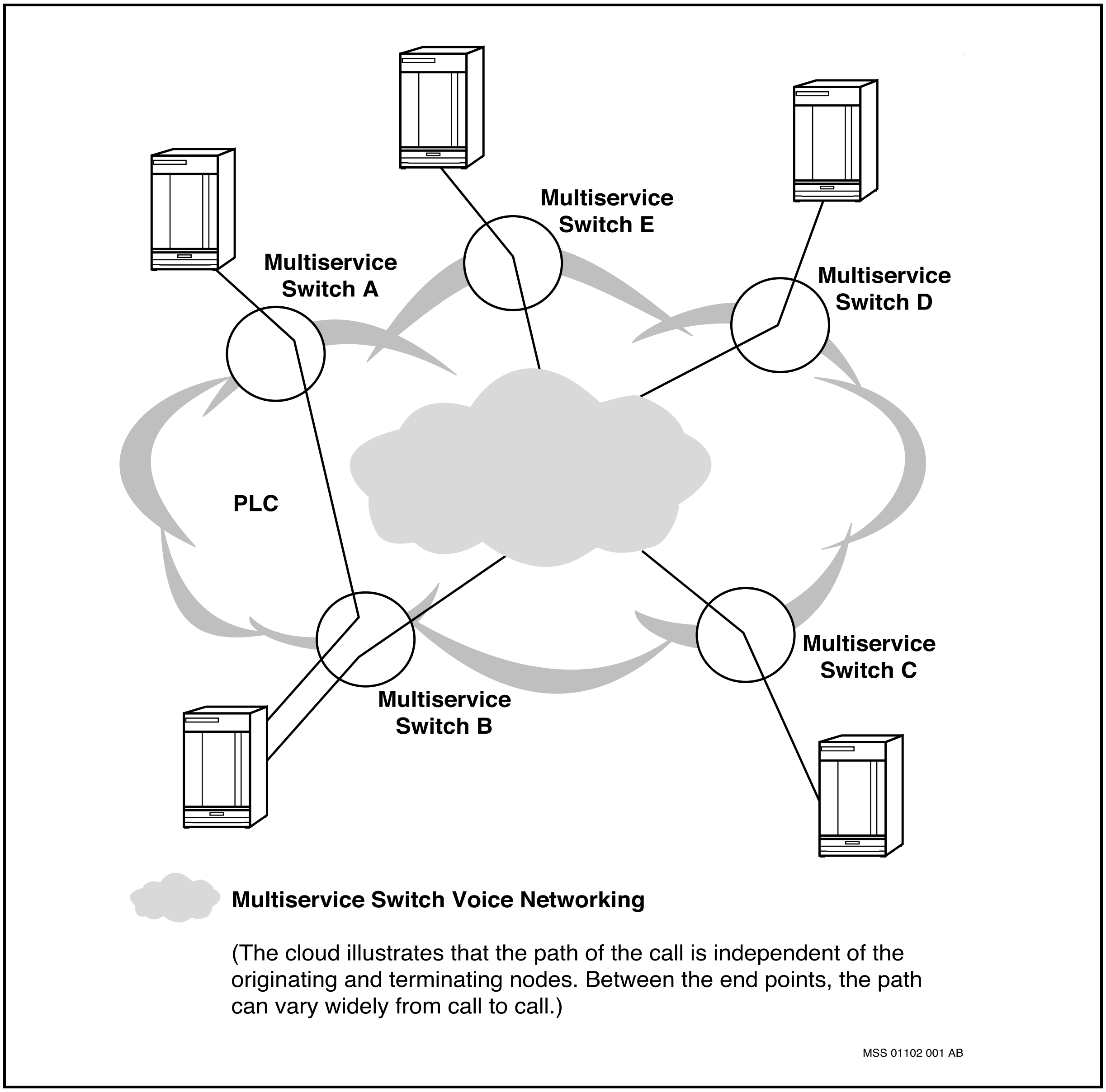


However, if nodes A and B are interconnected by a PLC, as shown in the figure Example of mixed configurations (page 141), and nodes B and C are interconnected by Voice Networking, calls can pass from node C to A by way of B.

In the figure Example of mixed configurations (page 141), nodes B, C, D and E are connected by Voice Networking. Regardless of the route through the subnet, a single call only involves two PBXs–the originating and the terminating ones.

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

Example of mixed configurations



Upgrading to Voice Networking

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

- upgrading a Nortel Multiservice Switch network on a basis of easy changes first
- dropping a Multiservice Switch network into a private network

In each scenario the method is identical but the deciding criteria are different. See Terminology (page 142) for information on the terms used in the following section. The process of inserting Voice Networking is described in Methodology (page 143). See Migration: example one (page 144) and Migration: example two (page 151) for details.

Terminology

The following terms are used throughout this section:

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

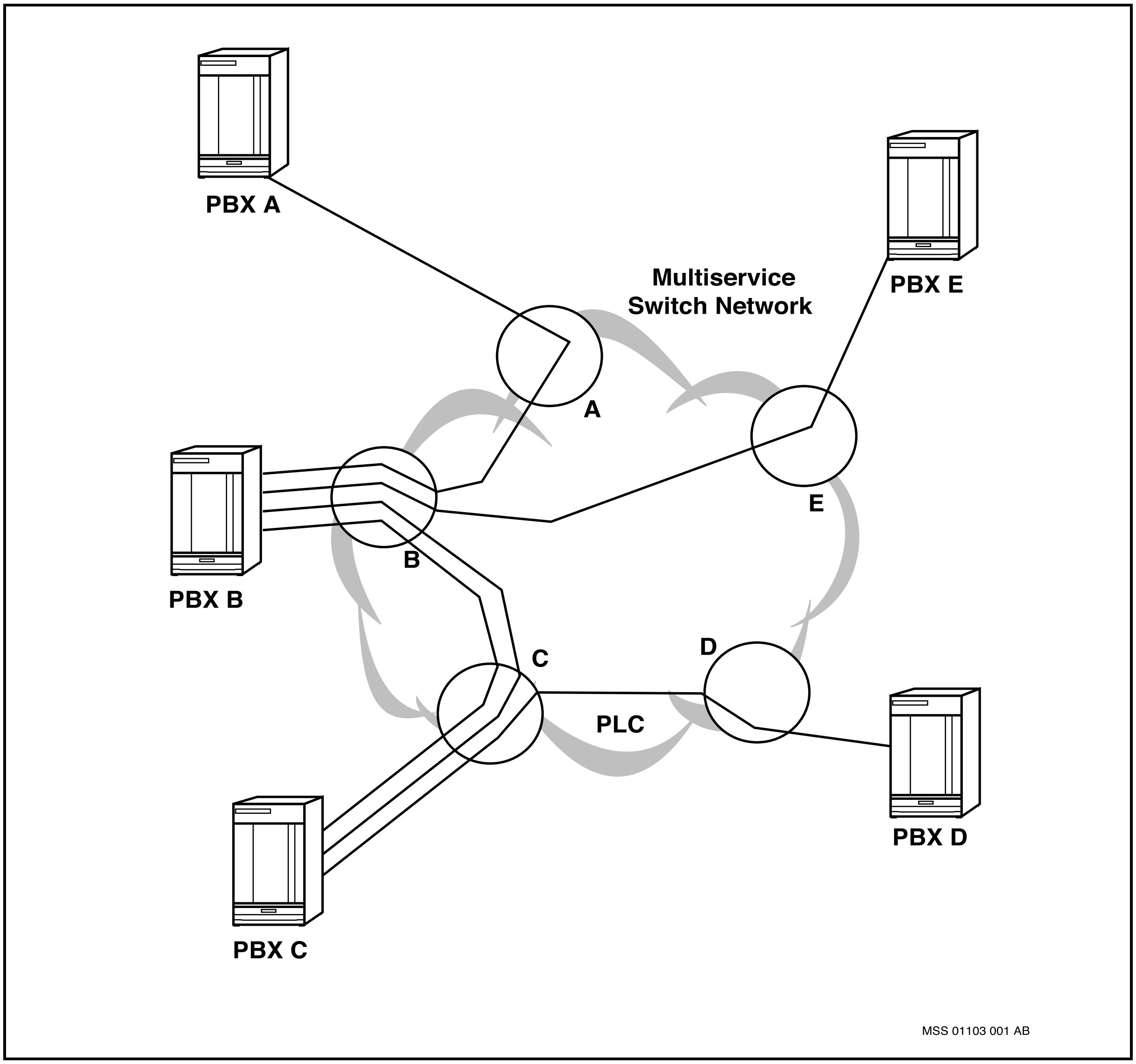
  An access path is shown in the figure Example of access paths and high volume path (page 143), interconnecting PBX/node D and PBX/node C.

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

  A high volume path is shown in the figure Example of access paths and high volume path (page 143), interconnecting PBX/node B and PBX/node C.

In the figure Example of access paths and high volume path (page 143), PBXs A, D and E are single point of entry stubs; they have a single connection to the subnet. PBXs B and C provide a high volume path, as D and C reach all other PBXs through the C to B link. Similarly, they are reached by the other three PBXs through the B to C link.

Example of access paths and high volume path



Methodology

Voice Networking can be inserted into an existing Voice Transport network that is based on point to point PLC connections. The sequence below shows an example of migrating from Voice Transport to Voice Networking

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

— Preferred links are either the highest volume path through the network, or single point of entry stubs connecting an isolated PBX to the network. They can be addressed in series (end to end path installation) or parallel (do all discrete paths first).

Group the discrete links into a VoiceRoute. For example, in the figure Example of access paths and high volume path (page 143), group the two links between PBXs B and C under a single *VoiceRoute* component.

— At the selected node to PBX connections add an FP link as a Voice Networking SVC.

— Provision the node for Voice Networking on the new logical link. Set up the SVC using the *overrideDirectoryNumber* attribute. All calls to this VoiceRoute for the local PBX are then routed to a specific endpoint regardless of the dialed number. The endpoint is defined in the voice networking call server (VNCS) database.

— Optionally, configure the VNCS numbering plan to destination PBX data. This step can be left until the network is configured more completely. For an example of a completed list of paths, see the table Multiservice Switch node VNCS database programming for migration example one (page 150).

— Provision the PBX to access the Voice Networking link (single E1 or DS1 on the PBX shelves).

2 Prove-in the SVC.

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

4 Repeat steps 1 to 3 with other PLC links until the network has been re-engineered.

5 If not yet created (see bullet in step), create the VNCS data and download it to all nodes.

6 Monitor the results of the VNCS mapping to confirm that calls terminate correctly. See the table Multiservice Switch node VNCS database programming for migration example one (page 150) for an example of a completed list of paths.

7 On a node by node basis, remove values from the *overrideDirectoryNumber* attribute for the selected PBX-to-node links. This allows calls originating on the selected PBX-to-node links to switch throughout the subnet.

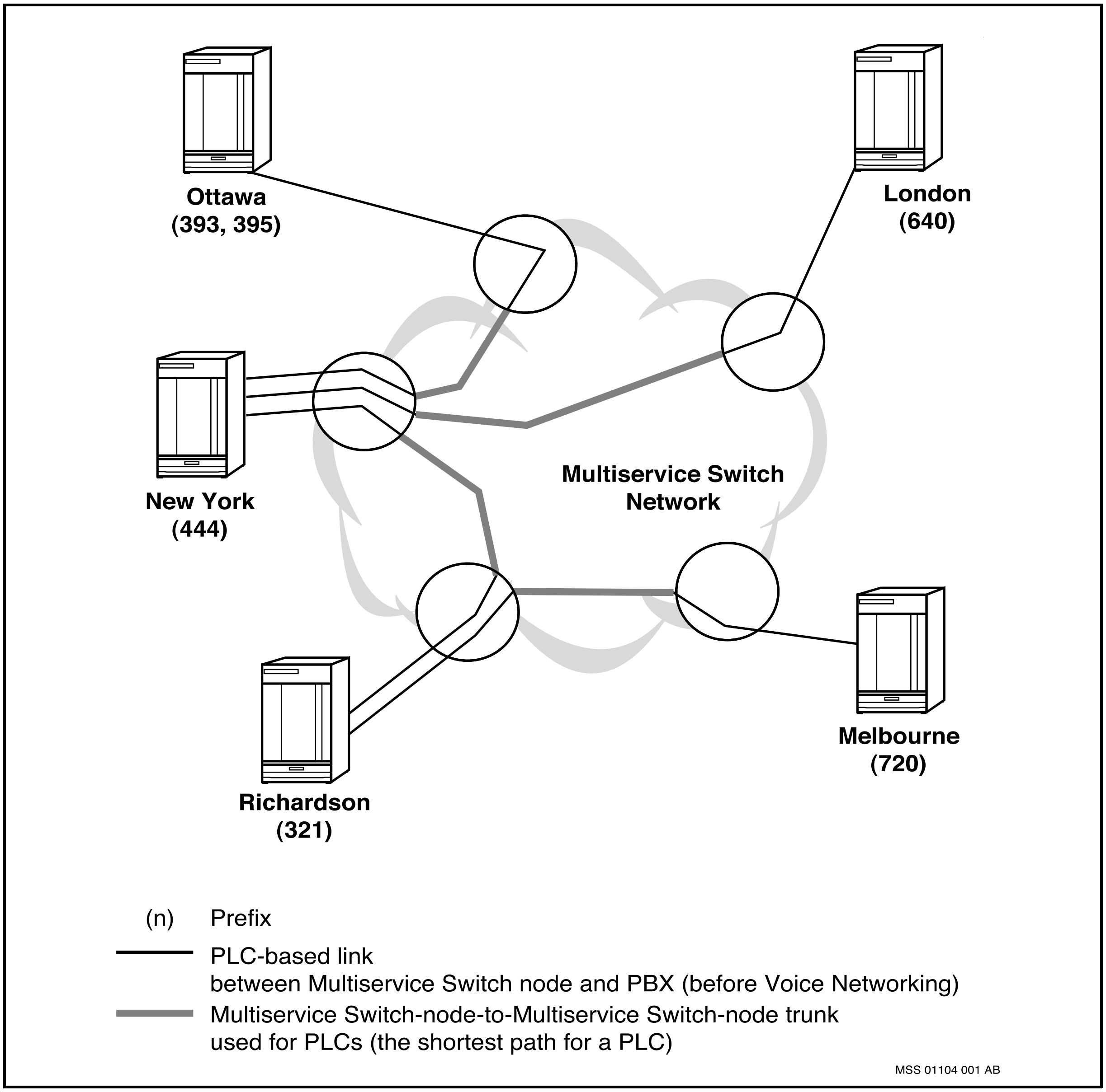
8 Optionally, optimize the node-to-PBX links to match traffic volumes by removing redundant DS1 or E1 links. That is, if a PBX to node link has four E1 cards (providing a total of 120 time slots) but has a peak volume of 75 calls, reduce the E1 card count to three (for a total of 90 time slots).

Migration: example one

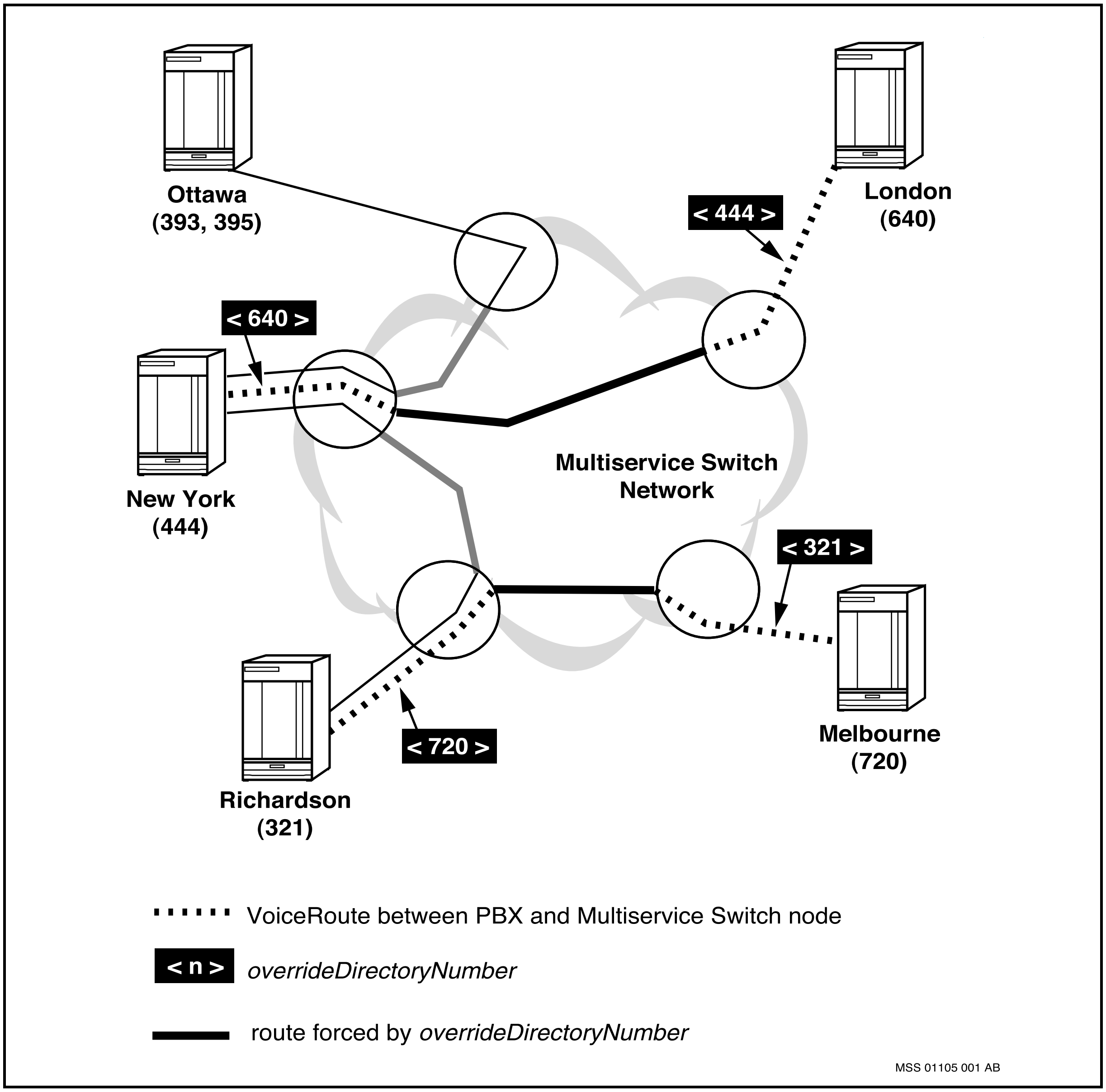
The figure Initial Multiservice Switch ISDN network - using PLCs (page 145) illustrates a network of five PBXs. One Nortel Multiservice Switch node has three PLC links, another has two. The remaining nodes have only one PLC link.

Logically, a Voice Networking connection to any single link PBX (such as London) behaves like a PLC at the stub end. Changing one or more of the Ottawa / London / Melbourne links is simple, with the stub end programming identical at all phases of the change over.

Initial Multiservice Switch ISDN network - using PLCs



Conversion of Multiservice Switch ISDN network



The sequence below shows an example of upgrading a network on a basis of "easy changes first".

1 Initially, London and Melbourne will be migrated to Voice Networking (see the figure Stage 2 of conversion of Multiservice Switch ISDN network (page 148)). Add Voice Networking switched links–parallel to the PLCs that already connect each node with their associated PBX.

2 In New York, London, Richardson and Melbourne create the VNCS database shown in the table VNCS database (page 147).

VNCS database

Prefix	Destination	NodeId	Route	VoiceProfile
444	New York	1021	VRoute/x	2
321	Richardson	1019	VRoute/x	2
720	Melbourne	32	VRoute/x	1
640	London	97	VRoute/x	1

3 Using the *overrideDirectoryNumber* attribute in the *VoiceRoute* component, force all calls originating in Melbourne to be routed to Richardson and all calls originating in London to be routed to New York.

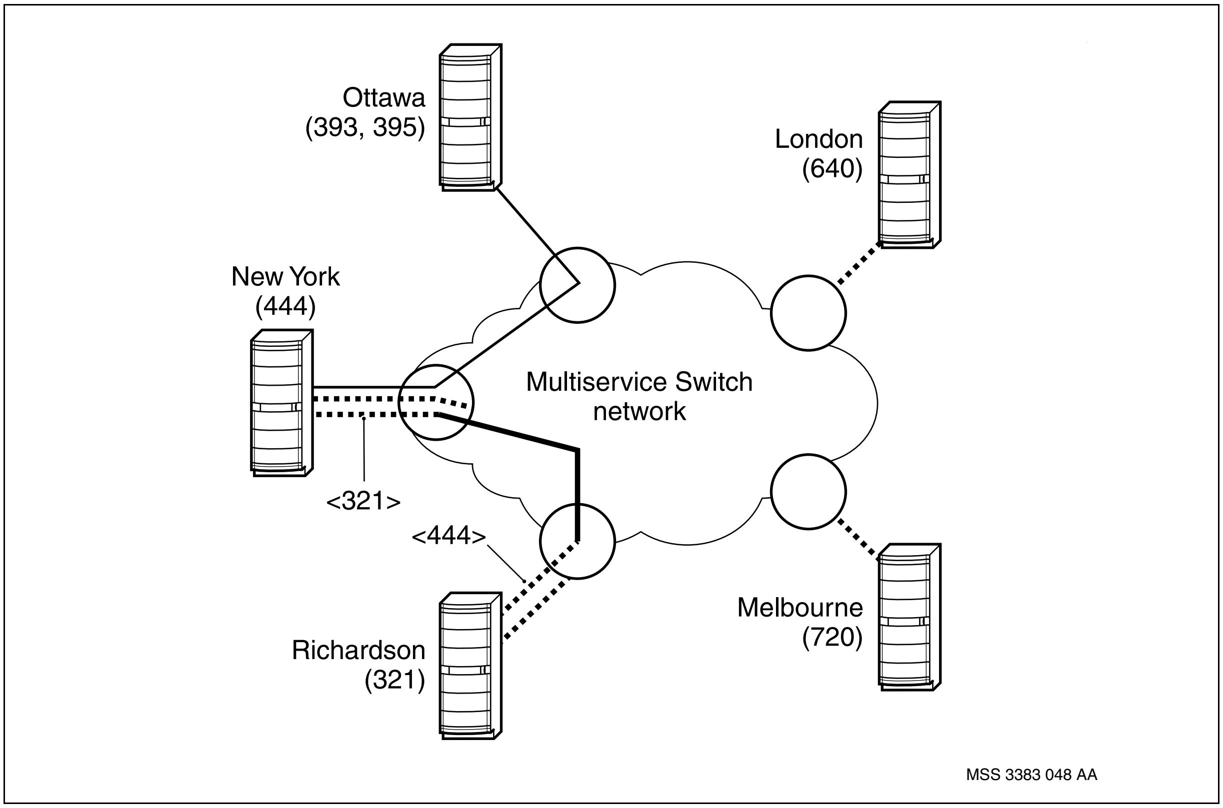
4 Prove-in the links and de-commission the PLCs (New York-London and Richardson-Melbourne) at the Voice Networking sites.

There is no physical change to the network topology. However, the “Richmond-Melbourne” and “New York-London” voice connections are now dynamic–lasting only as long as a call is active.

The initial links are now in. Since all other links involve New York, it is probable that due to shelf space constraints only one link can be converted at a time. In addition, should the provisioner be updating VNCS data during the installation of links, the risks of error are higher when configuring two links at the New York site at once.

5 Assume that the New York to Richardson link is desired next, to allow maximum intercontinental savings. See the figure Stage 2 of conversion of Multiservice Switch ISDN network (page 148). Duplicate the New York to Richardson PLC link and program the nodes to use the new switched links as though they were PLC connections (using overrideDirectoryNumber). Note that the VNCS database in the table VNCS database (page 147) does not need to be updated as it already contains the prefixes for New York and Richardson.

Stage 2 of conversion of Multiservice Switch ISDN network



6 Prove-in the switched voice links.

7 Decommission and remove the PLC links.

8 Duplicate the New York-Ottawa link (the only remaining link). Update the VNCS database and set the overrideDirectoryNumber accordingly. See the table VNCS database (page 147).

9 Prove-in the switched voice links.

10 Decommission and remove the New York-Ottawa PLC link.

11 Reprogram the subnet.

At this time, Voice Networking has been configured on all nodes using the overrideDirectoryNumber. This means that Voice Networking is acting like an “intelligent PLC”. In other words, connections between nodes are only present when a call is active, but the end-points of the connection are independent of the dialed number because overrideDirectoryNumber is configured.

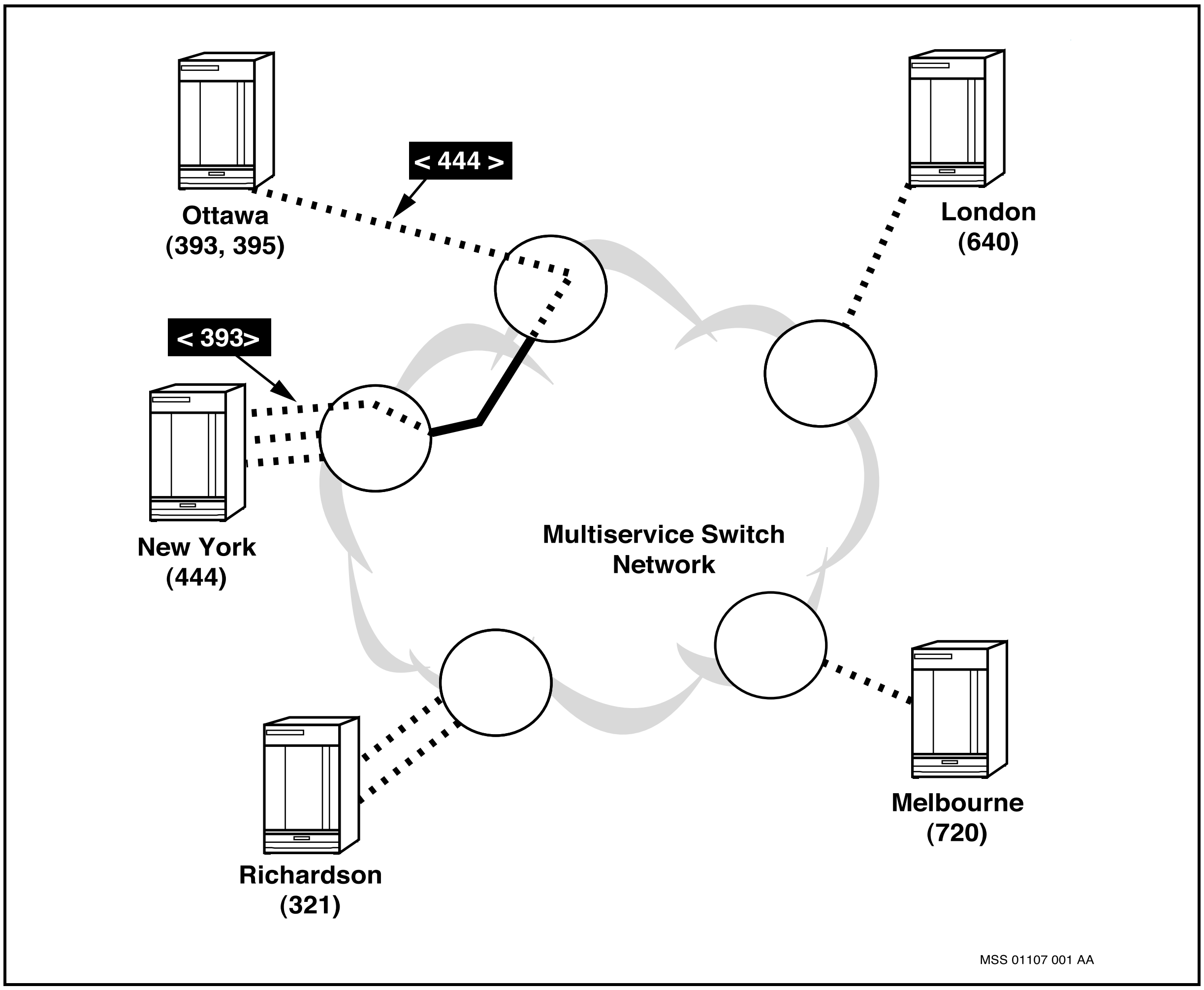
12 If not already done, ensure that the VNCS tables are identical on every node.

13 Start setting the overrideDirectoryNumber to “nothing”. For example, if overrideDirectoryNumber is set to “nothing” on Melbourne, then a Melbourne to Ottawa call would go directly instead of tandeming through Richardson and New York.

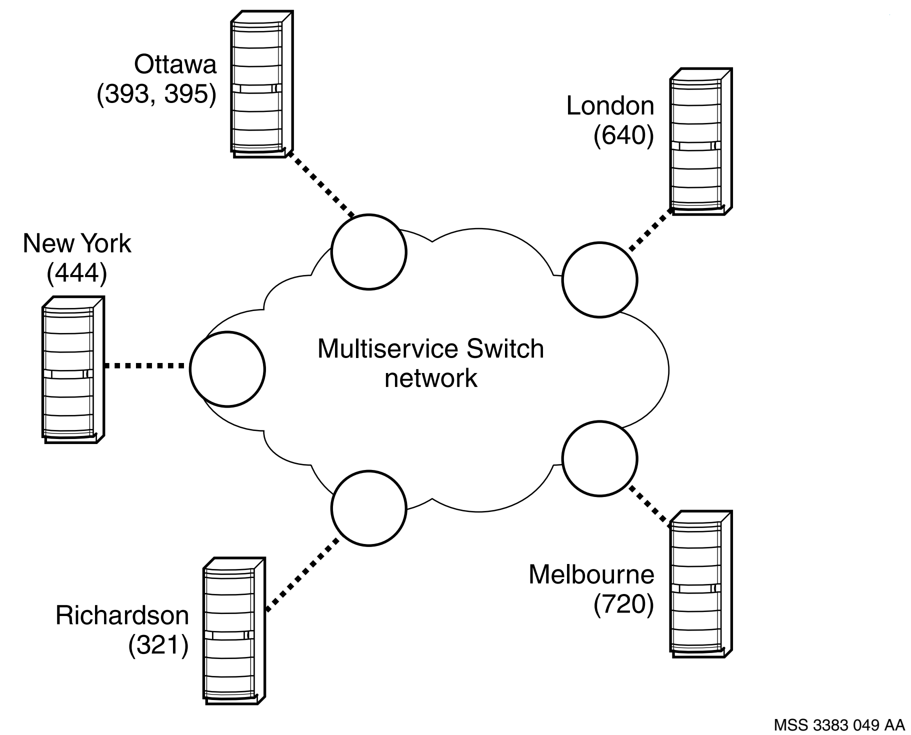
14 Provision the nodes identically. If not already done, provision the VNCS tables, and confirm that calls route correctly.

The table Multiservice Switch node VNCS database programming for migration example one (page 150) shows all of the configuration data for VNCS tables, although sites do not use their own node names. That is, a call from Melbourne to Melbourne does not enter the subnet so it never accesses that entry.

Final stage of conversion of Multiservice Switch ISDN network



Multiservice Switch ISDN network after completion of VNCS table entry



Multiservice Switch node VNCS database programming for migration example one

Prefix	Destination	NodeId	Route
444	New York	1021	VRoute/x
393, 395	Ottawa	632	VRoute/x
321	Richardson	1019	VRoute/x
720	Melbourne	32	VRoute/x
640	London	97	VRoute/x

Having replaced all PLC services, it is advisable but not necessary to optimize. If all routes to a single PBX created during the switch-over could be grouped in one single route on the PBX, the hardware and software can be optimized. If, for example, only two links worth of traffic now uses the three links at New York (since no tandem calls now use the PBX), one link can be removed.

Inserting Voice Networking into a PBX network

Voice Networking can be inserted into an existing network of PBXs based on point to point TIE (leased line PRI or CAS) connections. However, doing so requires hardware and data operations on both the PBXs and Nortel Multiservice Switch nodes involved. See Migration: example two (page 151).

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

   — Preferred links are either the highest volume path through the network, or single point of entry stubs connecting an isolated PBX to the network. They can be addressed in series (end to end path installation) or parallel (do all discrete paths first).

   — At the selected node to PBX connections add a Voice Networking SVC.

   — Provision the node for Voice Networking on the new logical link. Set up the SVC using an overrideDirectoryNumber to mimic PLC connections.

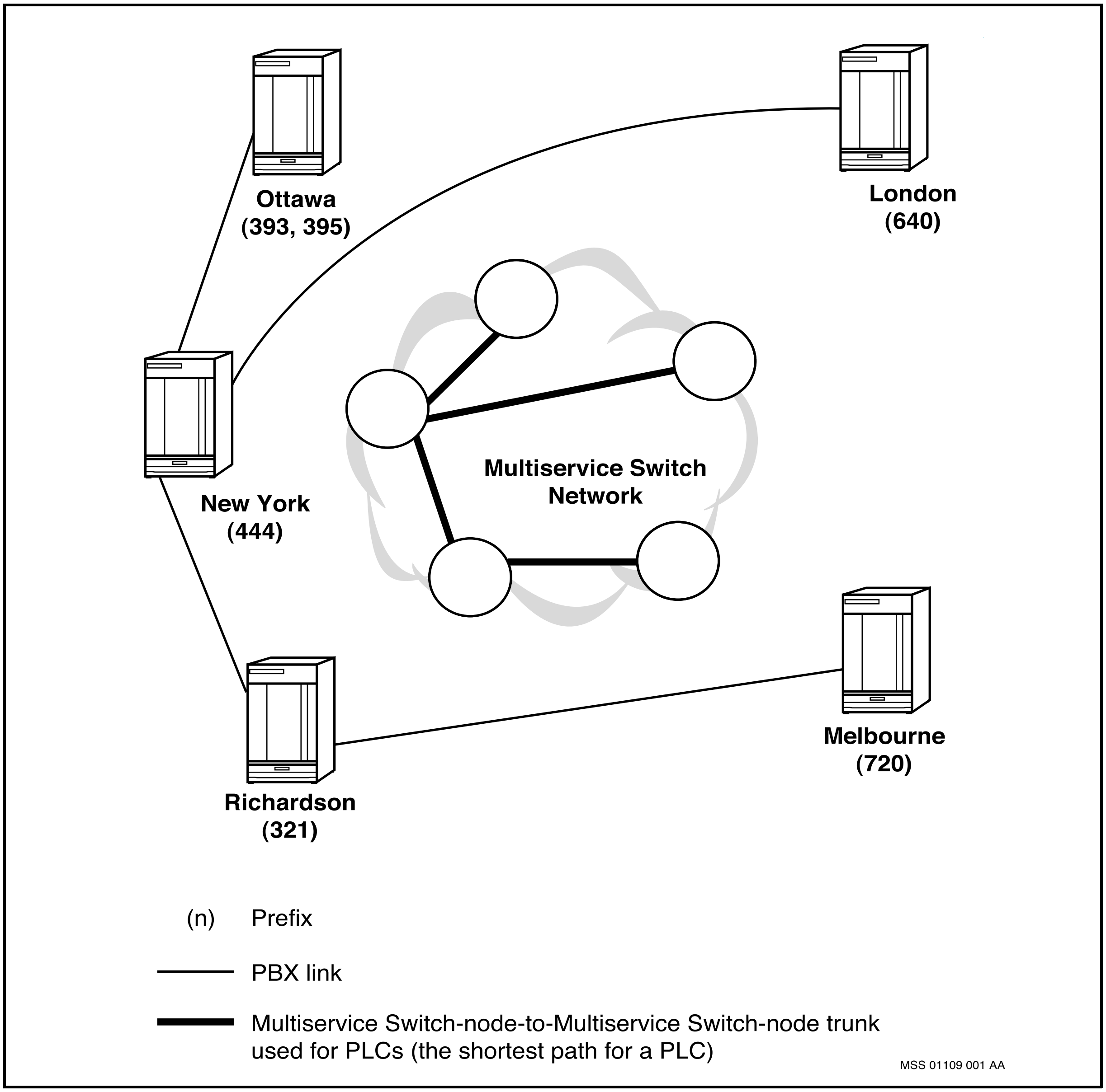
   — Provision the PBX to access the Voice Networking link (single E1 or DS1 on the PBX shelves).
2. Prove-in the PBX to node link.
3. Disable and remove the PBX to PBX TIE trunk.
4. Repeat steps 1 to 3 with other PLC links until the network has been re-engineered.
5. If not yet created (see bullet in step 1), create the VNCS data and download it to all nodes.
6. Monitor the results of the VNCS mapping to confirm that calls terminate correctly. See the table Multiservice Switch node VNCS database programming for migration example one (page 150) for an example of a completed list of paths.
7. On a node by node basis, turn off the overrideDirectoryNumber for the selected PBX-to-node links. This allows calls originating on the selected PBX-to-node links to switch throughout the subnet.
8. Optionally, optimize the node-to-PBX links to match traffic volumes by removing redundant DS1 or E1 links. That is, if a PBX to node link has four E1 cards (providing a total of 120 time slots) but has a peak volume of 75 calls, reduce the E1 card count to three (for a total of 90 time slots).

Migration: example two

The figure Initial Multiservice Switch ISDN network—using PLCs (page 152) illustrates a network of five PBXs. One Nortel Multiservice Switch node has three PLC links, another has two. The remaining nodes have only one PLC link.

Logically, a Voice Networking connection to any single link PBX (such as London) behaves like a PLC at the stub end. Changing one or more of the Ottawa / London / Melbourne links is simple, with the stub end programming identical at all phases of the change over.

Initial Multiservice Switch ISDN network—using PLCs

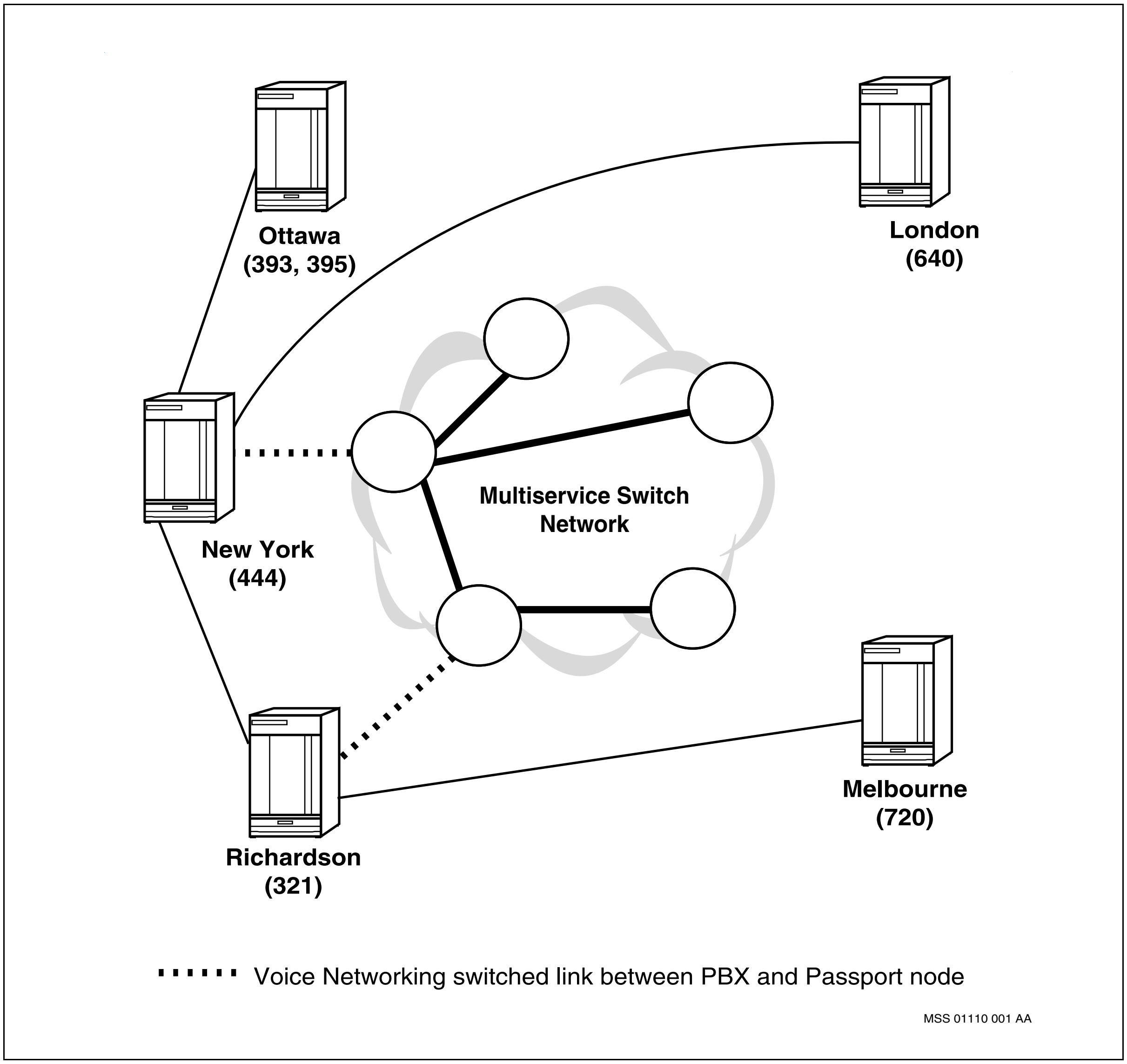


This sequence shows an example of adding a Multiservice Switch subnet to a private network.

1 Initially, the New York to Richardson link will be migrated to Voice Networking, See the figure Stage 1—conversion from PABX ISDN network to Multiservice Switch ISDN network (page 153). In this example, this is the highest traffic link. Add parallel SVCs connecting the node with the PBX. These are commissioned and replace the PBX TIE trunk links.

2 Duplicate the selected link. Program the nodes at the two ends (New York, Richardson) for "fixed end point" operation. Program the PBX to send all calls to the Voice Networking link instead of the TIE trunk. Prove in the new link.

Stage 1—conversion from PABX ISDN network to Multiservice Switch ISDN network

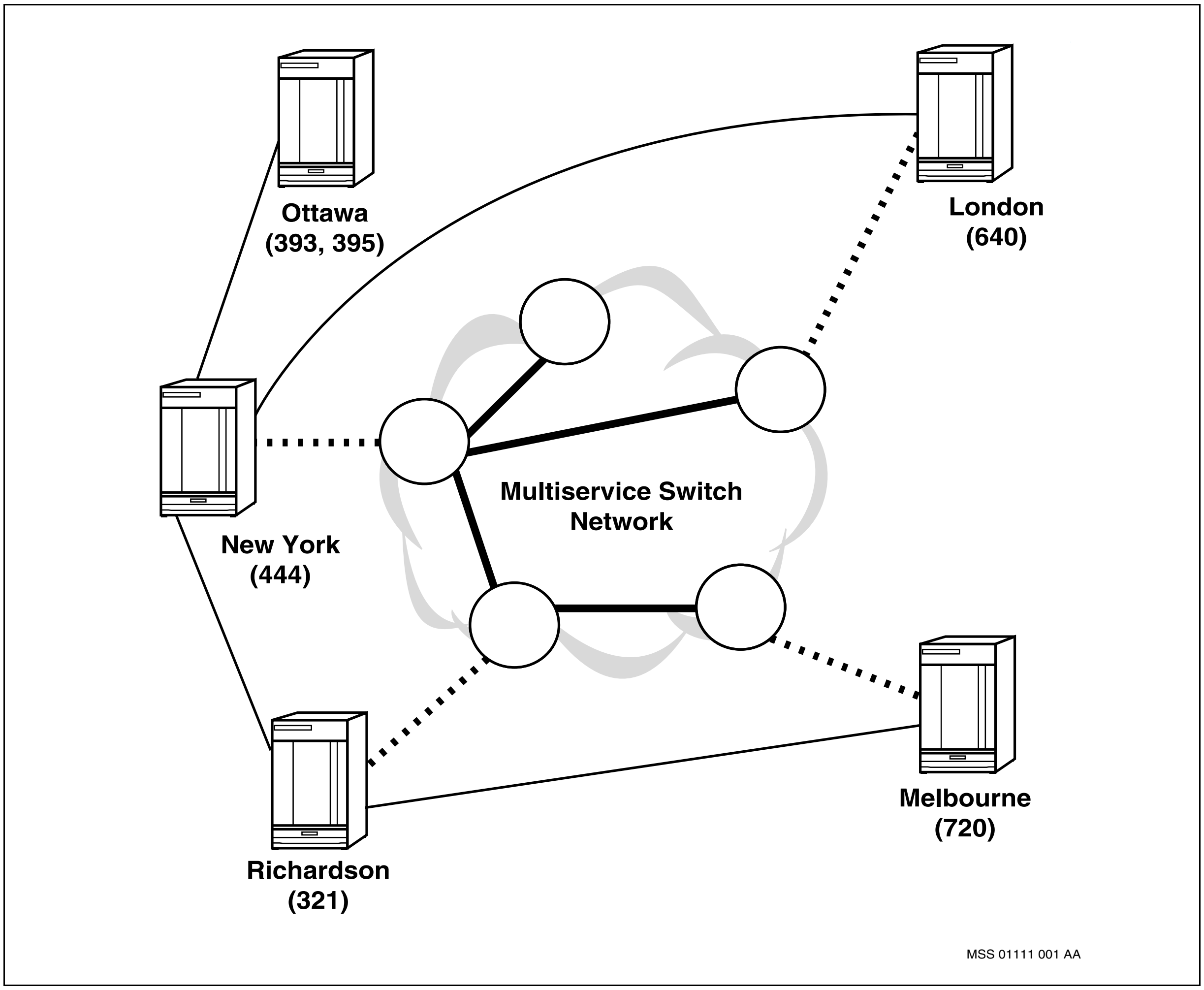


3 Prove-in the links, de-commission the PBX links (New York - Richardson).

The initial link is now in. See the figure Stage 1—conversion from PABX ISDN network to Multiservice Switch ISDN network (page 153). The New York to Ottawa, New York to London, and Richardson to Melbourne stubs remain. Since traffic to Canada represents the lowest cost (in this example), leave Ottawa for last.

4 Duplicate the selected links. Program the nodes at the two ends of each link (Richardson and Melbourne, New York and London) for “fixed end point” operation. Program the PBX to send all calls to the Voice Networking link instead of the TIE trunk. Prove in the new links.

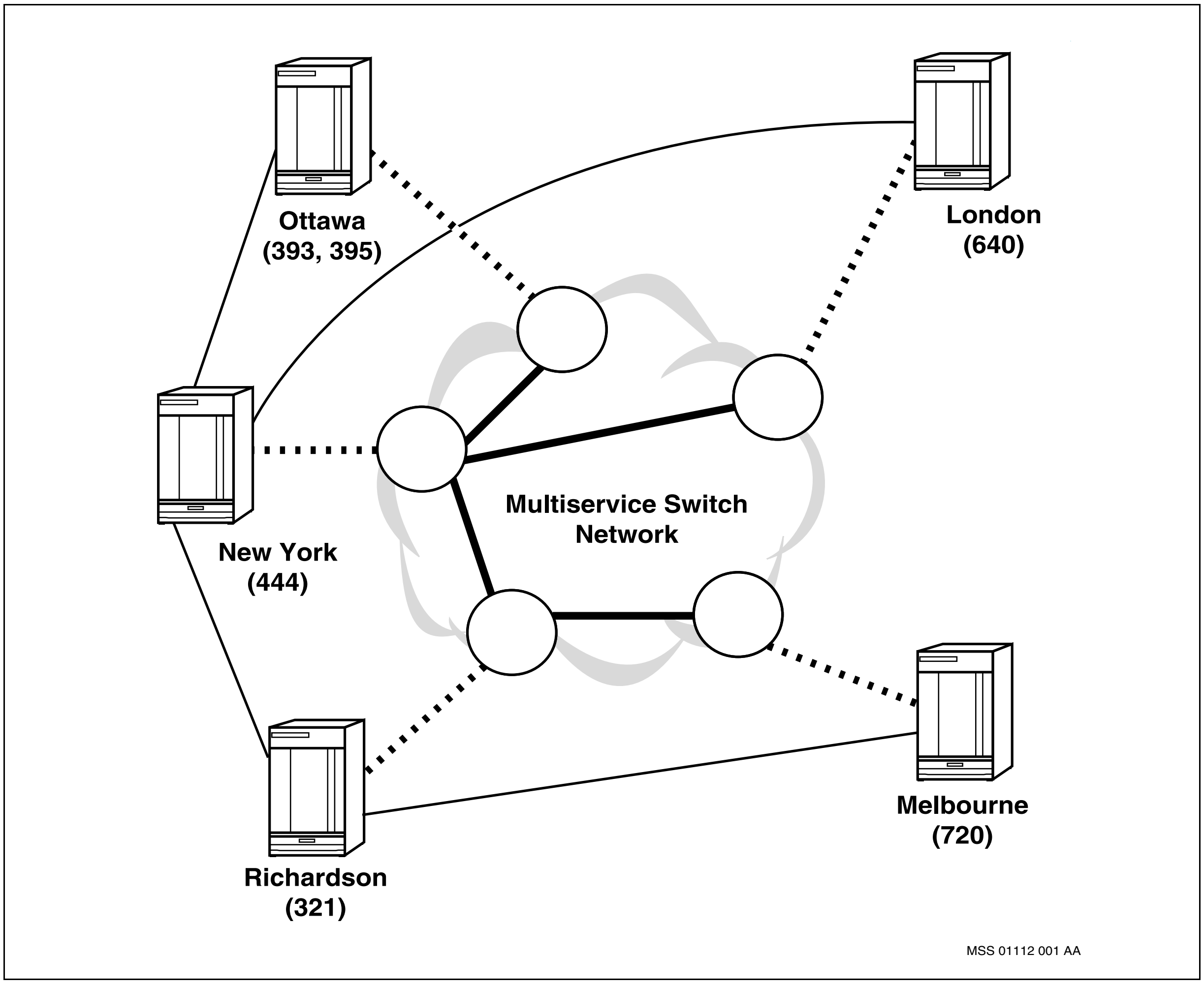
Stage 2—conversion from PABX ISDN network to Multiservice Switch ISDN network



5 Prove-in the switched voice links, decommission and remove the PBX links. See the figure Stage 2—conversion from PABX ISDN network to Multiservice Switch ISDN network (page 154). If updating the VNCS data as each node is configured, reprogram the New York, Richardson, Melbourne and London nodes to access the applicable routes. (This may be phased; it may be desirable to confirm the new Voice Networking links and then reprogram the nodes).

6 Duplicate the remaining link. Program the nodes at the two ends of the link (New York and Ottawa) for “fixed end point” operation. Program the PBX to send all calls to the Voice Networking link instead of the TIE trunk. Prove in the new link.

Final stage—conversion from PABX ISDN network to Multiservice Switch ISDN network



7 Prove-in the link from New York to Ottawa, decommission the PBX link and reprogram the subnet. All nodes should now require identical configuration. If it was not already done, provision the VNCS tables, and confirm that calls route correctly.

8 Having completed the hardware conversion, the links are as shown in the figure Final stage—conversion from PABX ISDN network to Multiservice Switch ISDN network (page 155). The configuration required to access these is required.

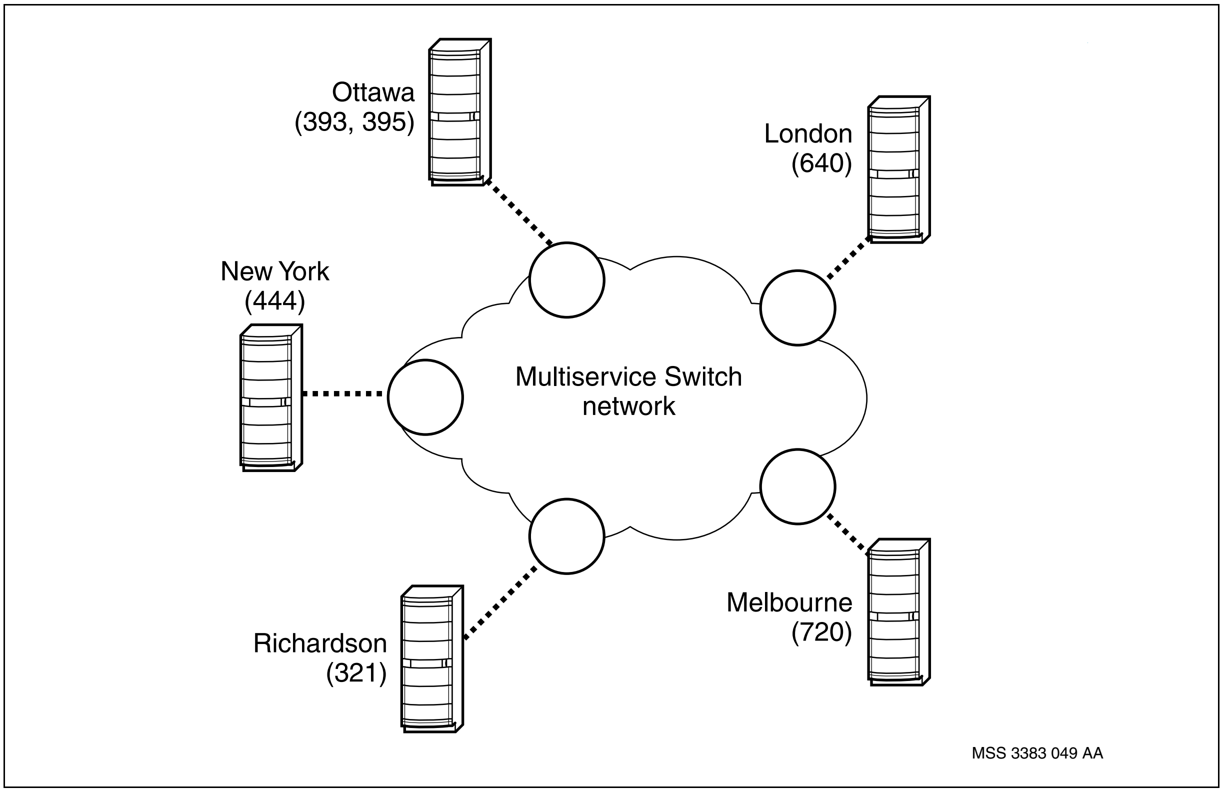
9 The table Multiservice Switch node VNCS database programming for migration example two (page 156) shows all of the configuration data for VNCS tables, although sites do not use their own node names. That is, a call from Melbourne to Melbourne does not enter the subnet so it never accesses that entry.

Multiservice Switch node VNCS database programming for migration example two

Prefix	Destination	NodeId	Route	VoiceProfile
393, 395	Ottawa	632	VRoute/x	2
444	New York	1021	VRoute/x	2
321	Richardson	1019	VRoute/x	2
720	Melbourne	32	VRoute/x	1
640	London	97	VRoute/x	1

Having replaced all PBX TIE trunks, inefficiencies may exist, and it may be advisable (but not necessary) to optimize. If all routes to a single PBX created during the switchover could be grouped in one single route on the PBX, the hardware and software can be optimized. If, for example, only two links worth of traffic now uses the three links at Richardson (since no tandem calls now use the PBX), one link can be removed.

Completed conversion from PABX ISDN network to Multiservice Switch ISDN network

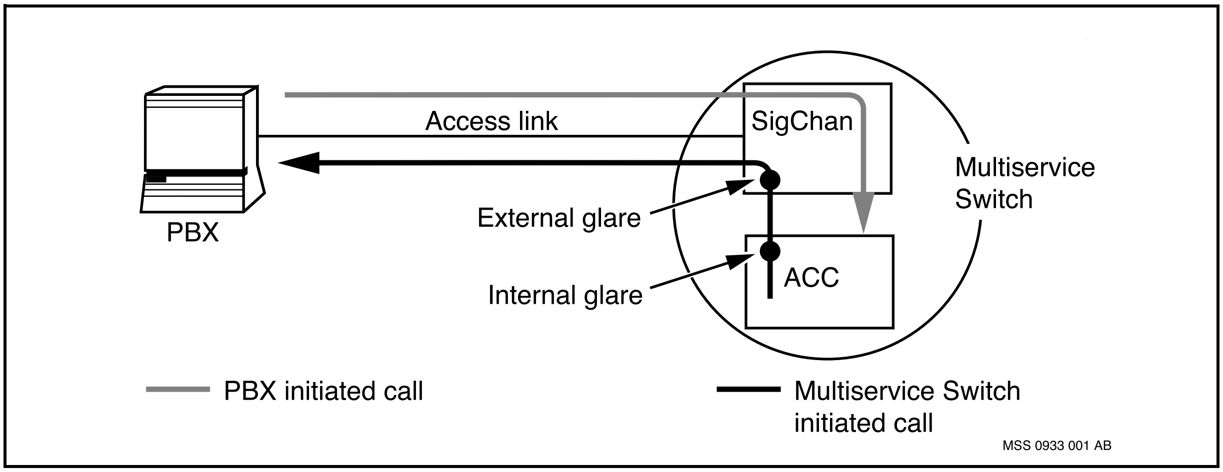


Glare processing

Glare, also referred to as call collision or dual seizure, occurs if both a PBX and its access Nortel Multiservice Switch node try to seize the same channel at the same time (see the figure Glare (call-collision) on access signaling (page 158)). From the node's perspective, glare can be

- Internal glare (page 158)
- External glare (page 159)

Glare (call-collision) on access signaling



Navigation

- Internal glare (page 158)
- External glare (page 159)

Internal glare

The Nortel Multiservice Switch node initiated call has not yet been passed to the *SignallingChannel* component when glare is detected. This glare is not detectable by the PBX. The call handling is as follows:

1 The PBX gets the requested channel for its call.

2 The node selects another channel for its call. That is, the node backs-off; no channel negotiation is necessary.

External glare

The call initiated by the Nortel Multiservice Switch node has already been passed to the *SignallingChannel* component when glare is detected. Call handling proceeds differently depending on which access signaling—CAS or CCS—is being used.

For CAS signaling, the node backs off and tries to reestablish the call.

For CCS signaling, call handling depends on who is the network and who is the user, and proceeds as follows:

1. The network side gets the requested channel for its call (call setup is done using ‘exclusive’).
2. The user side may be offered another channel for its call by the network side. That is, the user backs-off; a channel negotiation is initiated by the network (call setup is done using ‘preferred’).

Call release cause codes

Call release cause codes only apply to Voice Networking calls that use common channel signaling (CCS) protocols. Voice Networking handles internally and externally generated call release cause codes for failed (for example, during call establishment) and clearing calls. Internal cause codes are those generated by the Nortel Multiservice Switch node or from within the subnet. External cause codes are those received by a node from a PBX. The table Call release cause code values (page 161) contains all cause values in the range 0 to 127. In most cases, Voice Networking transports without modification the cause values that it receives. However, some cause values do not apply at all points in a connection and not all cause values are supported by all protocols and PBXs.

Standard cause code values are commonly defined for all protocols in the ITU-T Q.850 specification. However, each individual protocol may introduce a different handling of a given cause value or additional cause code values. The word No in the table Call release cause code values (page 161) means the cause value is not supported by the particular CCS protocol. The word Yes means that the CCS protocol supports the cause value and Voice Networking does not map the value to another, unless otherwise indicated. The table Call release cause code values (page 161) also notes the cause values that Voice Networking maps to more common values. See also Internal cause code definition (page 165) for more information about cause code handling.

Attention: Unless otherwise specified, the supported Euro-ISDN cause values include the ETSI Austrian and German user variants and the generic ETSI ISDN network variant.

Call release cause code values

Cause value	Description	Supported by...			
		NIS A211-1	ETSI QSIG	Euro-ISDN	MCDN
0	Unallocated (unassigned) number	No (maps to 31)	No (maps to 31)	No	No (0 is sent)
1	Unassigned (unallocated) number	Yes	Yes	Yes	Yes
2	No route to specified transit network	Yes	No (maps to 3)	Yes (maps to 3)	Yes
3	No route to destination	Yes	Yes	Yes	Yes
4, 5	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
6	Channel unacceptable	No (maps to 31)	No (maps to 31)	Yes	Yes
7	Call awarded and being established	No (maps to 31)	No (maps to 31)	No	Yes
8 - 15	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
16	Normal call clearing	Yes	Yes	Yes	Yes
17	User busy	Yes	Yes	Yes	Yes
18	No user responding	Yes	Yes	Yes	Yes
19	No answer from user (user alerted)	Yes	Yes	Yes	Yes
20	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
21	Call rejected	Yes	Yes	Yes	Yes
22	Number changed	Yes	Yes	Yes	Yes
23 - 25	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
26	Non-selected user clearing	No (maps to 31)	No (maps to 31)	Yes (maps to 31)	Yes
27	Destination out-of-service	Yes	Yes	Yes	Yes
28	Invalid number format	Yes	Yes	Yes	Yes
29	Facility rejected	Yes	No (maps to 31)	Yes (maps to 31)	Yes
30	Response to status enquiry	No (maps to 31)	No (maps to 31)	No	Yes
31	Normal, unspecified	Yes	Yes	Yes	Yes
(1 of 5)					

Call release cause code values (continued)

Cause value	Description	Supported by...			
		NIS A211-1	ETSI QSIG	Euro-ISDN	MCDN
32 - 33	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
34	No circuit/channel available	Yes	Yes	Yes	Yes
35 - 37	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
38	Network out of order	No (maps to 31)	No (maps to 41)	Yes (maps to 41)	Yes
39 - 40	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
41	Temporary failure	Yes	Yes	Yes	Yes
42	Switching equipment congestion	Yes	No (maps to 41)	Yes (maps to 41)	Yes
43	Access information discarded	Yes	No (maps to 31)	Yes (maps to 31)	Yes
44	Requested circuit/channel not available	Yes	Yes	Yes	Yes
45	Channel preempted	No (maps to 31)	No (maps to 41)	No	Yes
46	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
47	Resources unavailable, unspecified	Yes	No (maps to 44)	Yes	Yes
48	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
49	Quality of service unavailable	No (maps to 31)	No (maps to 58)	Yes (maps to 58)	Yes
50	Requested facility not subscribed	Yes	No (maps to 31)	Yes (maps to 31)	Yes
51	Call barred due to access group restrictions	No (maps to 31)	No (maps to 31)	No	Yes
52	Outgoing call barred	No (maps to 31)	No (maps to 31)	No	Yes
(2 of 5)					

Call release cause code values (continued)

Cause value	Description	Supported by...			
		NIS A211-1	ETSI QSIG	Euro-ISDN	MCDN
53	Outgoing call barred within customer user group (CUG)	No (maps to 31)	No (maps to 31)	Yes (ETSI Germany only; otherwise maps to 31)	No (maps to 31)
54	Incoming call barred	Yes	No (54 is sent)	No	Yes
55	Incoming calls barred within CUG	No (maps to 31)	No (maps to 31)	Yes (ETSI Germany only; otherwise maps to 31)	No (maps to 31)
56	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
57	Bearer capability not authorized	Yes	Yes	Yes	Yes
58	Bearer capability not presently available	Yes	Yes	Yes	Yes
59 - 62	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
63	Service or option unavailable, unspecified	Yes	Yes	Yes	Yes
64	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
65	Bearer capability not implemented	Yes	Yes	Yes	Yes
66	Channel type not implemented	Yes	No (maps to 65)	Yes (maps to 65)	Yes
67 - 68	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
69	Requested facility not implemented	No (maps to 31)	No (maps to 63)	Yes (maps to 63)	Yes
70	Only restricted digital information bearer capability is available	Yes	No (maps to 63)	Yes (maps to 63)	Yes
71 - 78	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
79	Service or option not implemented	Yes	No (maps to 63)	Yes (maps to 63)	Yes
(3 of 5)					

Call release cause code values (continued)

Cause value	Description	Supported by...			
		NIS A211-1	ETSI QSIG	Euro-ISDN	MCDN
80	N/A	No (maps to 31)	No (maps to 31)	No	No (80 is sent)
81	Invalid call reference value	Yes	Yes	Yes	Yes
82	Identified channel does not exist	Yes	No (maps to 44)	Yes (maps to 44)	Yes
83	A suspended call exists, but this call identity does not	Yes (maps to 31)	No (maps to 0)	No	Yes (maps to 0)
84	Call identity in use	Yes (maps to 31)	No (maps to 0)	No	Yes (maps to 0)
85	No call suspended	Yes (maps to 31)	No (maps to 0)	No	Yes (maps to 0)
86	Call having the requested call identity has been cleared	Yes (maps to 31)	No (maps to 0)	No	Yes (maps to 0)
87	User not member of CUG	No (maps to 31)	No (maps to 31)	Yes (ETSI Germany only; otherwise maps to 31)	No (maps to 31)
88	Incompatible destination	Yes	Yes	Yes	Yes
89	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
90	Non-existent CUG	Yes	No (maps to 31)	Yes (ETSI Germany only; otherwise maps to 31)	No (maps to 31)
91	Invalid transit network selection	No (maps to 31)	No (maps to 31)	No	Yes
92 - 94	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
95	Invalid message, unspecified	Yes	No (maps to 31)	Yes (maps to 111)	Yes
96	Mandatory information element missing	No (maps to 31)	No (maps to 31)	Yes	Yes
97	Message type non-existent or not implemented	No (maps to 31)	No (maps to 31)	Yes	Yes

(4 of 5)

Call release cause code values (continued)

Cause value	Description	Supported by...			
		NIS A211-1	ETSI QSIG	Euro-ISDN	MCDN
98	Message type either incompatible with current state or not implemented	No (maps to 97 or 101)	No (maps to 31)	Yes	Yes
99	Information element non-existent or not implemented	No (maps to 31)	No (maps to 31)	Yes	Yes
100	Invalid information element contents	No (maps to 31)	No (maps to 31)	Yes	Yes
101	Message not compatible with call state	No (maps to 31)	No (maps to 31)	Yes	Yes
102	Recovery on timer expiry	No (maps to 31)	No (maps to 31)	Yes	Yes
103 - 110	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
111	Protocol error, unspecified	No (maps to 31)	No (maps to 31)	Yes	Yes
112 - 126	N/A	No (maps to 31)	No (maps to 31)	No	No (maps to 31)
127	Interworking, unspecified	No (maps to 31)	No (maps to 31)	Yes	No (maps to 31)
(5 of 5)					

Internal cause code definition

For certain internal cause codes relating to call establishment you can configure Voice Networking to transmit a particular cause code value. This capability allows you to exactly match the cause value required by an originating PBX to establish a call on an alternate route. The capability to define a cause value applies to calls using the following CCS protocols: ETSI QSIG, NIS A211-1, and Euro-ISDN (the MCDN protocol uses the drop back busy capability to reroute calls).

The table Definable internal cause codes (page 166) contains the cause, typical clearing reason(s), and examples of the cause value that is sent for each protocol. When defining a cause value for any of the causes listed in the table, any value in the range 0 to 127 is acceptable. See Configuring the SignalingChannel component (page 20) for procedural information on how to specify cause code values.

Definable internal cause codes

Cause	Typical clearing reason(s)	Example cause value sent		
		NIS A211-1	ETSI QSIG	Euro-ISDN
Egress link out of service	• The data link to the remote PBX is down. • The channels at the remote end are in maintenance mode.	27	27	27
No circuit/channel available	• No timeslot available for an incoming call.	34	34	34
Temporary failure	• path-oriented routing system (PORS) fails to establish a call due to subnet congestion. • A trunk is disabled. • Calling side fails to decode the payload capabilities sent by the called side.	41	41	41
Switched equipment congestion	• Allocation of internal resources for an incoming call fails (for example, the *SignallingChannel* component is down).	42	41	42
Requested circuit/ channel not available	• An exclusively requested channel is not in service. • A channel request is unsuccessful.	44	44	44
Resource unavailable, unspecified	• Path establishment timer expires. • An H-channel (hybrid channel) is requested. • The *SwitchedVoiceService* component associated with a particular call is locked on the egress Nortel Multiservice Switch node.	47	44	47
Service not allowed	• The requested call type (voice or data) is not compatible with the configured value of the *typeOfRoute* attribute under the *VoiceRoute* component at the calling side.	63	63	63
No such channel	• The requested channel does not exist.	82	44	82
Incompatible destination	• The capabilities of the called and calling side do not match.	88	88	88

Procedure conventions

This document uses the following procedure conventions:

- You can enter commands using full component and attribute names, or you can abbreviate them. The commands used in the procedures contain the full component and attribute names in the first instance. In the second instance, the component and attribute names are abbreviated. For more information on abbreviating component and attribute names, see NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference*. All component and attribute names are formatted in italics.
- The introduction of every procedure states whether you must perform the procedure in operational mode or provisioning mode. For more information on these modes, see Operational mode (page 167) or Provisioning mode (page 168).
- When you complete a procedure, you can verify your changes and then activate them as the new node configuration. For more information on completing configuration changes and exiting provisioning mode, see Activating configuration changes (page 168).

Operational mode

Procedures contained within this document can either be performed in operational mode or provisioning mode. When you initially log into a node, you are in operational mode. Nortel Multiservice Switch systems use the following command prompt when you are in operational mode:

```
#>
```

where:
is the current command number

In operational mode, you work with operational components and attributes. In operational mode, you can

- list operational components and display operational attributes to determine the current operating parameters for the node
- control the state of parts of the node by locking and unlocking components

- set certain operational attributes and enter commands to perform diagnostic tests

Provisioning mode

To change from operational mode to provisioning mode, type the following command at the operator prompt:

```
start Prov
```

Only one user can be in provisioning mode at a time. Nortel Multiservice Switch systems use the following command prompt whenever you are in provisioning mode:

```
PROV #>
```

where:

is the current command number

In provisioning mode, you work with the provisionable components and attributes that contain the current and future configurations of the node. You can add and delete components, and display and set provisionable attributes. For information on completing the configuration changes, exiting provisioning mode, and returning to operational mode see Activating configuration changes (page 168).

For information on operational and provisionable attributes, see NN10600-060 *Nortel Multiservice Switch 7400/15000/20000 Component Reference*.

Activating configuration changes

Several procedures in this document ask that you complete the configuration changes. When you complete the configuration changes, you are activating the configuration changes, confirming that you want to activate them, and saving the changes. You are instructed to complete the configuration changes only at the end of procedures that you perform in provisioning mode.

CAUTION
Activating a provisioning view can affect service
Activating a provisioning view can result in a CP reload or restart, causing all services on the node to fail. See NN10600-050 *Nortel Multiservice Switch 7400/15000/20000 Command Reference*, for more information.

CAUTION
Risk of service failure
When you activate the provisioning changes (see step 3), you have 20 minutes to confirm these changes. If you do not confirm these changes within 20 minutes, the shelf resets and all services on the node fail.

1 Verify that the provisioning changes you have made are acceptable.

```
check Prov
```

Correct any errors and then verify the provisioning changes again.

2 If you want to store the provisioning changes in a file, save the provisioning view.

```
save -f(<filename>) Prov
```

3 If you want these changes as well as other changes made in the edit view to take effect immediately, activate, confirm, and commit the provisioning changes.

```
activate Prov
confirm Prov
commit Prov
```

4 End the provisioning session.

```
end Prov
```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