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VoIP User Manual

Release 4.5

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**This manual documents Release 4.5 Voice over IP (VoIP) hardware and software.
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1 VoIP Overview

Introduction

This chapter describes Alcatel's H.323 Voice over IP (VoIP) gateway and how telephone or fax calls can be programmed to automatically go through either an enterprise's Virtual Private Network (VPN) via the gateway, and/or the Public Switched Telephone Network (PSTN), a circuit-switched public telephone network that consists of all the interconnected calling networks in the world.

Alcatel's H.323 VoIP gateway provides transparent, seamless delivery and connection of local and long distance, inbound and outbound telephone calls initiated through standard PSTN North American (T1), European (E1), and Euro ISDN (Integrated Services Digital Network) digital telephone transmission lines. For specific details on the precise types of calls handled, see Chapter 2, "VoIP Daughtercards" and Chapter 5, "VoIP Commands."

As with standard T1, E1, ISDN (Euro) telephone service, VoIP calls can be transmitted full-duplex (simultaneously in both directions). Likewise, Alcatel's H.323 VoIP gateway digitizes phone or fax call signals and uses one of these call transmission services, depending on the type of call, to channel the calls, i.e., carry information to a destination point in the VoIP network. Depending on the configuration of the VoIP network, the calls may also go through the PSTN. For more details on the digitizing process, see *Payload Packetization and Digital Signal Processing* on page 1-12 for more details.

This chapter provides general background information on VoIP networks, clients, gateways and gatekeepers, and includes a list of key features provided specifically by Alcatel's H.323 VoIP network. A VoIP call scenario is illustrated and described along with details on the technologies used in VoIP to explain how calls can be placed over IP. Elements of converged VoIP network are also shown and described, and significant telephone and data communications technologies are explained in relation to the VoIP gateway. Technical standards for the most prominent technologies used in Voice over IP are briefly summarized at the end of this chapter, since as a whole, H.323 ITU standards define the major components using VoIP technologies in network-based VoIP communications systems.

VoIP Networks

Alcatel's H.323 VoIP gateway for packet-switched IP networks combines the speed, versatility and low cost of IP telephony with standard telephone features for enterprises in North American and Europe (this necessarily entails other continents or countries, such as Mexico, that may have the same requirements). These networks are referred to as VoIP networks. Because data networks as such usually operate with extra carrying capacity (bandwidth), most IP networks are easily able to accommodate voice/fax traffic once the VoIP network is set up.

The Internet Protocol (IP) is used mainly because it is supported over many layer 1 and 2 network technologies including Ethernet (10, 100, 1000 Mbps), Token Ring, FDDI and Frame Relay to name a few, including leased lines and satellites. Nearly every router, frame relay device, and network switch used today supports the Internet Protocol. IP delivers any transport media used between local and wide area networks.

Enterprise IP networks consist of local area networks (LANs) installed at corporate offices often joined together by corporate wide area networks (WANs). Usually the local area networks support IP on various types of standard data communications technologies such as Ethernet, Token Ring, ATM (Asynchronous Transfer Mode) and FDDI (Fiber Distributed Data Interface).

Wide area networks are used to support IP connections over leased lines, public frame relay, ATM, satellite and ISDN. At each branch office location, enterprises use routers to connect the remote LANs to the IP WAN. When used with Alcatel's H.323 VoIP gateway, these Virtual Private Networks, or VPNs, allow a portion of the Public Switched Telephone Network to be managed and used by the enterprise.

Alcatel's H.323 VoIP gateway provides the Voice over IP network capabilities by means of either digital or analog VoIP daughtercards installed in the switch. The VoIP daughtercards specifically enable enterprises to control the call routing capabilities of their own organizations by using a portion of the PSTN.

Getting Started with VoIP

Once an enterprise network is ready to provide VoIP using Alcatel's H.323 VoIP gateways, Network Administrators can begin setting up VoIP networks by installing and configuring the appropriate voice switching daughtercard(s). As a whole, Alcatel's VoIP H.323 gateways can be scaled from a minimum of two voice channels per switch to a maximum of 120 voice channels per switch. Switches with the greatest VoIP scalability will use voice switching modules (VSXs) in Omni Switch/Routers. See Chapter 2, "VoIP Daughtercards," for further details on supported configurations and scalability.

Initially, an Alcatel VoIP network dialing scheme (AVNDS) must also be selected and deployed via a text-based configuration boot file, wherein each daughtercard must be assigned a unique IP address among other unique gateway identifiers. From that point, operational parameters such as channel and port types can be set using the command line interface (CLI) configuration tool. Comparable text-based (ASCII) configuration boot files may also be quickly generated to configure multiple VoIP-enabled switches with similar requirements.

Also, stored in the **vsmbboot.asc** files are voice coding parameters which are pre-configured and kept in profiles. Coding Profiles are configured directly to the components, and define which operational VoIP characteristics will be used, and then implemented according to the instructions contained in the profiles. Coding Profiles consist of general caller information, voice and fax transmission, coding/decoding settings. Preferred Coding Profiles can be automatically selected based upon payload requirements. Coding Profiles are configured at the channel level. VoIP configurations for VoIP callers are established by setting up profiles and then assigning the profiles to each individual H.323 VoIP gateway or daughtercard. Profiles can be created, modified, copied and deleted using one of the available configuration tools. It should be known that in most circumstances, the default settings for the Coding Profiles are sufficient.

Additional parameters that require configuration include calling Destinations and Network Numbering Schemes, the latter being comprised primarily of Numbering Plans, Phone Groups and Hunt Methods. Altogether, use of these parameters enable VoIP networks to translate IP addresses from telephone numbers, and allow communications between the VoIP branch offices to be configurable. For more details, see *Alcatel VoIP Network Dialing Schemes (AVNDS)* on page 1-15.

See Chapter 5, "VoIP Commands," for details on using these CLI commands once the H.323 VoIP gateway is configured; refer to this chapter as well if using an optional third-party gatekeeper (server) i.e., NT100 RADVision, on a PC for example, as some additional network parameters will need to be set.

For details on configuring the AVNDS, see "Chapter 3, "Network Dialing Schemes" and Chapter 5, "VoIP Commands." For details on installing the cards and setting up VoIP H.323 Gateways, see Chapter 4, "Setup and Installation."

Alcatel's H.323 VoIP Gateway Key Features

Alcatel's H.323 VoIP gateways, which connect voice and data networks, minimize call complexity and dependency on leased telephone lines by allowing enterprises more control over their own call processing. Alcatel's H.323 VoIP gateway is used to transport digitized voice conversations over IP local area networks, which are then sent over wide area networks using such protocols as Frame Relay or ATM. All VoIP daughtercards are compatible with the Alcatel OmniAccess 512 and Omni Switch/Router. As shown below, the following features of Alcatel's H.323 VoIP gateway are supported in this release.

- Digital T1/E1 voice and fax transport over IP networks
- T1 and E1 telephony interface links to digital Private Branch Exchanges (PBXs) via digital or analog VoIP daughtercards.
- T1 robbed-bit Channel Associated Signaling (CAS)
- E1 Primary Rate Interface (Euro PRI) and E1 QSIG ISDN Common Channel Signaling (CCS)
- E1 Basic Rate Interface (Euro BRI) ISDN; (E1 ETSI)
- Foreign Exchange Station (FXS) telephony Loop interface via analog VoIP daughtercard with (FXS) grand-daughtercard (variations includes FX Office— FXO).
- H.323 Network Call Control Gateway (establishes Local Area Network (LAN) terminal links; performs call setup and voice translation functions; provides communications procedures between LANs)
- Voice Codecs: Pulse Code Modulation (G.711), Internet Speech (G.723.1), Standard Telephone Quality (G.729A), Realtime Fax over IP (Fax T.38).
- Non-Voice Signal Monitoring, Detection and Transmission Protocols:
 - Dual Tone Multi Frequency/Modem Fax Relay
 - Fax Transparency and Fax Relay
 - Modem Transparency and Modem Relay
- PSTN Fallback via Deadman Relay Switch
 - The H.323 VoIP gateway is capable of providing PSTN fallback for VoIP calls in the event of a power failure in the VoIP network by means of a Deadman relay switch on the digital VoIP daughtercards. For more information on the Deadman switch, see Chapter 2, VoIP Daughtercards.”
- Echo and jitter controls on digital VoIP daughtercards.
- Pre-configured, modifiable AVNDS (Alcatel VoIP Network Dialing Schemes) with corresponding text-based (ASCII) configuration boot files (**vsmboot.asc** files).
- VoIP Text-based Command Line Interface (CLI) configuration tool.

◆ **Note** ◆

When used separately, the terms E1 and ETSI both entail European PRI and BRI interfaces. E1 ETSI used together as one term refers specifically to Euro PRI.

VoIP Telephone Calls

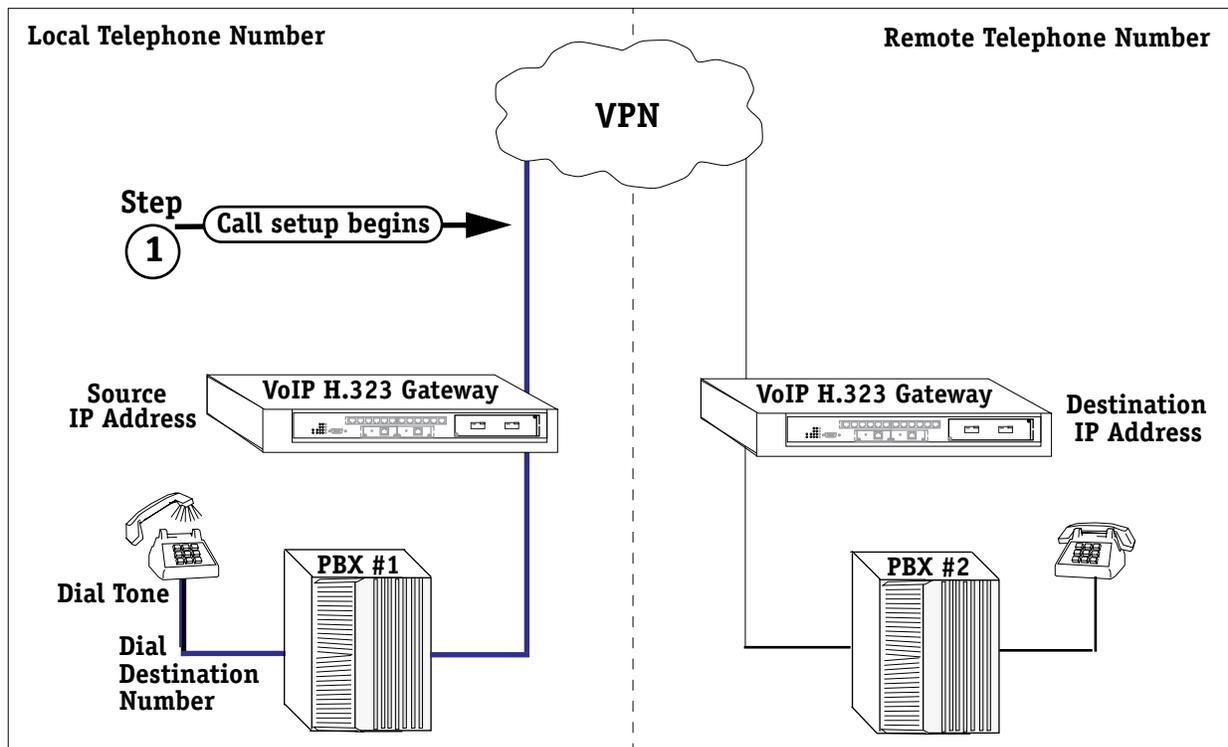
H.323 VoIP telephone calls, which can carry either voice, facsimile, or modem transmissions over IP networks, are switched to the packet-based network and connected to the calling destination (an IP device) via a unique IP address and local/remote dialing plan (actually two Alcatel VoIP Network Dialing Schemes rolled into one). The numerical IP address, also serving to identify calls intended for VoIP networks, is determined and translated from a destination telephone number in a phone directory database while it is being entered, and the call is in progress. (It should be noted here that callers do not need to remember the IP addresses, only the called party or destination phone number). See Chapter 3, "Network Dialing Schemes," for more information on the AVNDS.

H.323 VoIP telephone calls are transparent so callers don't have to worry about any special procedures, except being aware of a dialing plan that may require them to dial a prefix, such as 7, before a call can be placed across the VoIP network. This would be similar to current dialing plans requiring callers to dial 9 before an office call can be placed (9 is the prefix most often used by PBXs to access the PSTN).

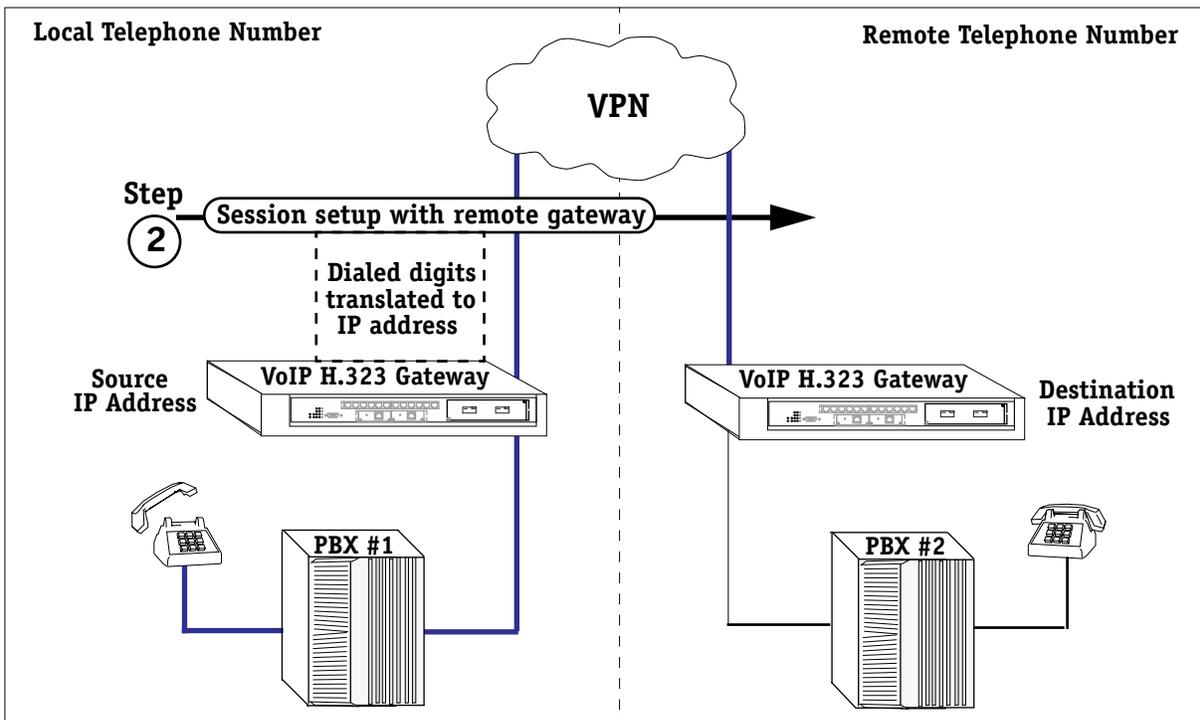
VoIP calls initiated from standard telephone handsets after a preset number of digits are dialed, for example, can be immediately transmitted using IP data networks whereby digital or analog signals, meant to set up connections for carrying information, are intercepted by Alcatel's H.323 VoIP gateways in the network. These gateways translate the phone numbers into IP addresses, convert the information to digital packet form, and then deliver the calls over the network and the PSTN as shown below.

A VoIP Call Scenario

Once a VoIP network is set up a typical VoIP call scenario might go something like this.



VoIP Call Scenario — Step 1: Call Setup

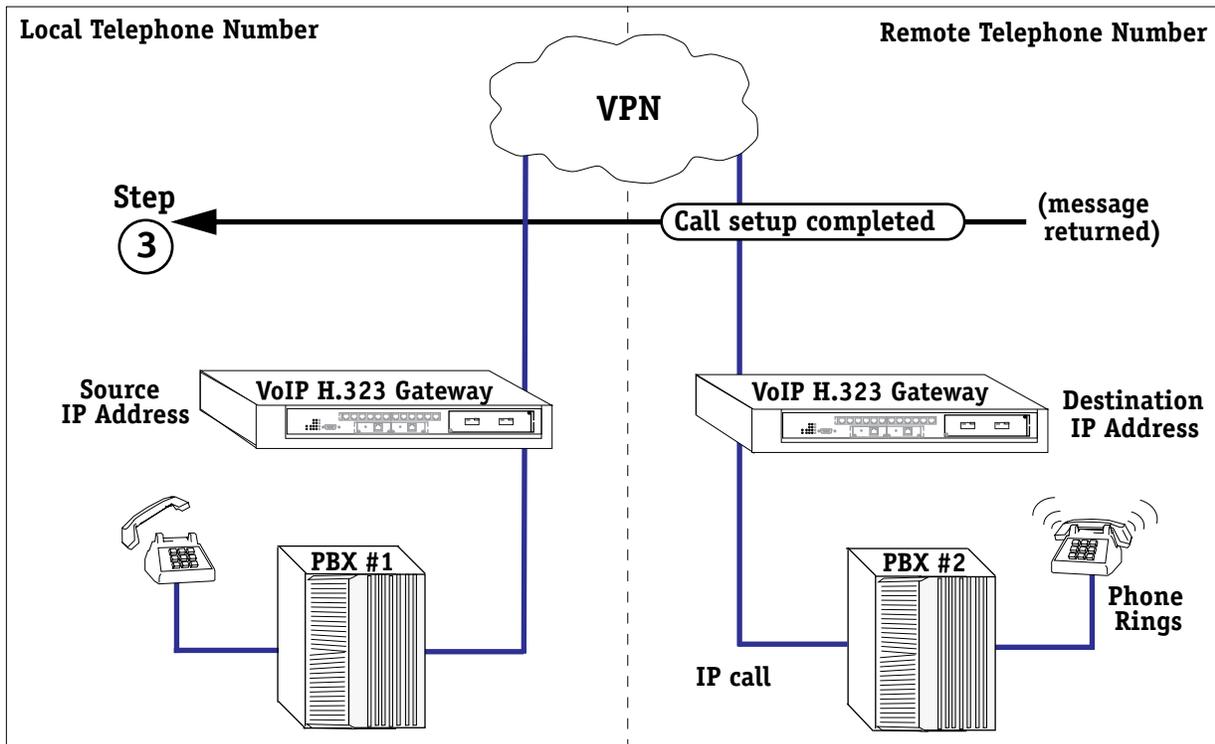


VoIP Call Scenario — Step 2: Call Progress

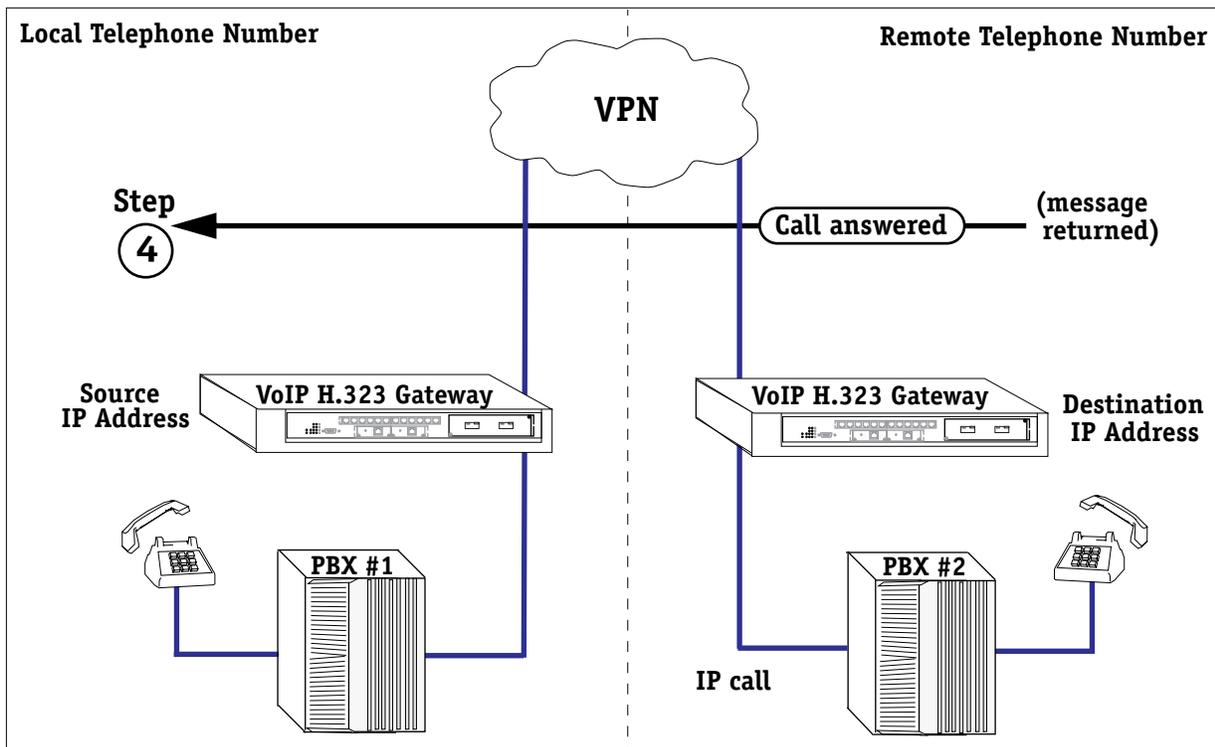
As the caller dials, the H.323 VoIP gateway collects the dialed digits and then ultimately translates the digits using a pre-configured Numbering Plan and Phone Group into the IP address. A VoIP session is then initiated with the remote gateway (when gatekeeper not used).

After the gateways determine that the VoIP call can be placed across the IP network, the gateways negotiate call capabilities using preconfigured coding profiles, and then optionally strip before sending the extension digits from the local to the remote gateway where they are delivered either to the phone, PBX, or keyset. The call can be processed as either a local or long distance call depending on how the remote gateway is configured. A ringing or busy signal is transmitted to the caller once the call is connected. If the call is answered, the gateway sends the voice or fax transmissions. If the wide area network is unavailable, calls may not go through, in which case callers receive a busy signal.

When a caller hangs up the receiver, the VoIP call session is terminated. Multiple gateway trunks may be used for all calls except those initiated from keysets which must go directly to the gateway.



VoIP Call Scenario — Step 3: Call Setup Completed



VoIP Call Scenario — Step 4: Remote Call Answered as VoIP Call

Elements of a Converged Network

Alcatel's H.323 VoIP gateway is based on a complex, dual-technology infrastructure taken from what have been in the past two fairly distinct industries — namely, Telecommunications (a.k.a. Telephony) and Data Communications. It converges voice and data into enterprise, Internet Service Providers (ISPs), and carrier networks to provide various levels of VoIP services using intelligent switches in order to generate long-term cost reductions for telephone services between sites.

The standard or key elements of a converged H.323 VoIP network are described below and shown in the illustration *Elements of Converged Voice/Data Packet-Switched VoIP Network and Circuit-Switched PSTN* on page 1-9. This illustration is intended to provide a sample, visual representation of all the various devices that may be used in a VoIP network and how they may interconnected. VoIP network interoperability is based on ITU H.323 network call control standards and multiple vocoder support. See also abbreviated *International Telecommunication Union (ITU) Standards* on page 1-17.

By means of either digital or analog VoIP daughtercards installed in Alcatel switches, the basic elements required for providing enterprise H.323 VoIP gateways in packet-switched IP networks are readily accommodated, including the client, the gateway, and the gatekeeper as described.

VoIP H.323 Client

The *Client* is the device initiating and/or receiving the call. This can be a standard telephone handset or some other H.323 VoIP-capable device in an IP network.

VoIP H.323 Gateway

Alcatel's H.323 VoIP Gateway is the device used to make the transition from the packetized voice network to a circuit-switched network, e.g., PSTN, and back. Functionally, the enterprise VoIP gateway is comprised of voice to IP network converter components, .e.g, DSPs, on the voice switching daughtercards. In VoIP, the process for call placement is the same as in a service provider system except that the gateway is accessed from Customer Premise Equipment (CPE) instead of from a local service provider, e.g. CLEC (Certified Local Exchange Carrier).

◆ Note ◆

PBX and Key Systems setup, installation and configuration procedures are beyond the scope of this manual.

Gateway devices intercept then direct electric signals between networked devices. With VoIP, gateways translate transmission formats between voice CPE and H.323 IP network call control endpoints and terminals, including communications procedures between gateways. They also translate between codecs, perform call setup/teardown on LANs and on circuit-switched telephone networks. Gateways are entrance and exit points into VoIP networks that without hardwiring perform code and protocol conversions, as well as signal filtering.

VoIP gateways contain a user-definable phone directory database of phone number to IP address mappings; this is called an Alcatel VoIP Network Dialing Scheme (AVNDS). See Chapter 3, "Network Dialing Schemes," for details. Modifications to the local phone directory database are downloaded through the IP network to the switch, and may be accessed using the VoIP configuration interface. The phone directory database is built as the VoIP network is configured, and is contained in the VoIP configuration boot file (**vsmboot.asc**). Numbering plans, phone groups and destinations as part of the AVNDS comprise a portion of the phone directory database used by Alcatel's H.323 VoIP gateway.

Gateways are considered H.323 terminals or H.323 endpoints in H.323 IP networks. Terminals are also the endpoints where telephone lines connect to network circuits. Terminals provide real time, two-way communications for local area network (LAN) endpoint destinations. All terminals as such must support voice communications and H.245 in-band call controls to use and negotiate channels. See also abbreviated *International Telecommunication Union (ITU) Standards* on page 1-17.

VoIP H.323 Gatekeeper (Optional)

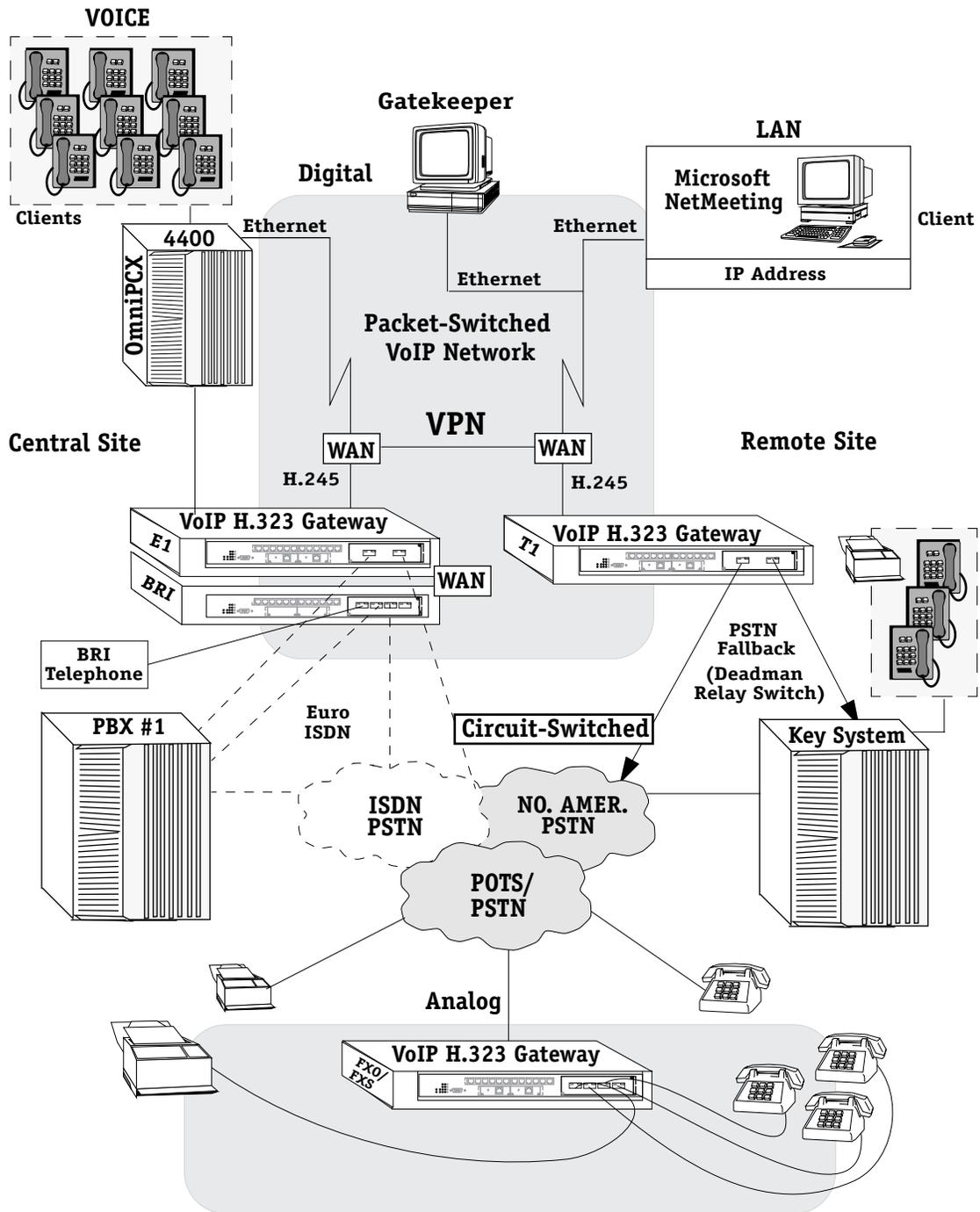
The H.323 Gatekeeper (server or workstation) is the device that verifies client VoIP privileges and translates telephone numbers into IP addresses. It should be noted that H.323 gatekeepers are not required to use Alcatel's H.323 VoIP Gateway. In lieu of an H.323 VoIP gatekeeper, Alcatel's H.323 VoIP gateway uses its patent-pending Alcatel VoIP Network Dialing Scheme (AVNDS) to perform IP address translations.

◆ Note ◆

Gatekeeper setup, installation and configuration procedures are beyond the scope of this manual.

Alcatel recommends and has tested extensively use of Alcatel's H.323 VoIP gateway with the NT100 RADVision Gatekeeper.

Gatekeeper devices identify, track and control traffic flowing through them, and perform other functions such as gateway registration, admission and bandwidth controls.



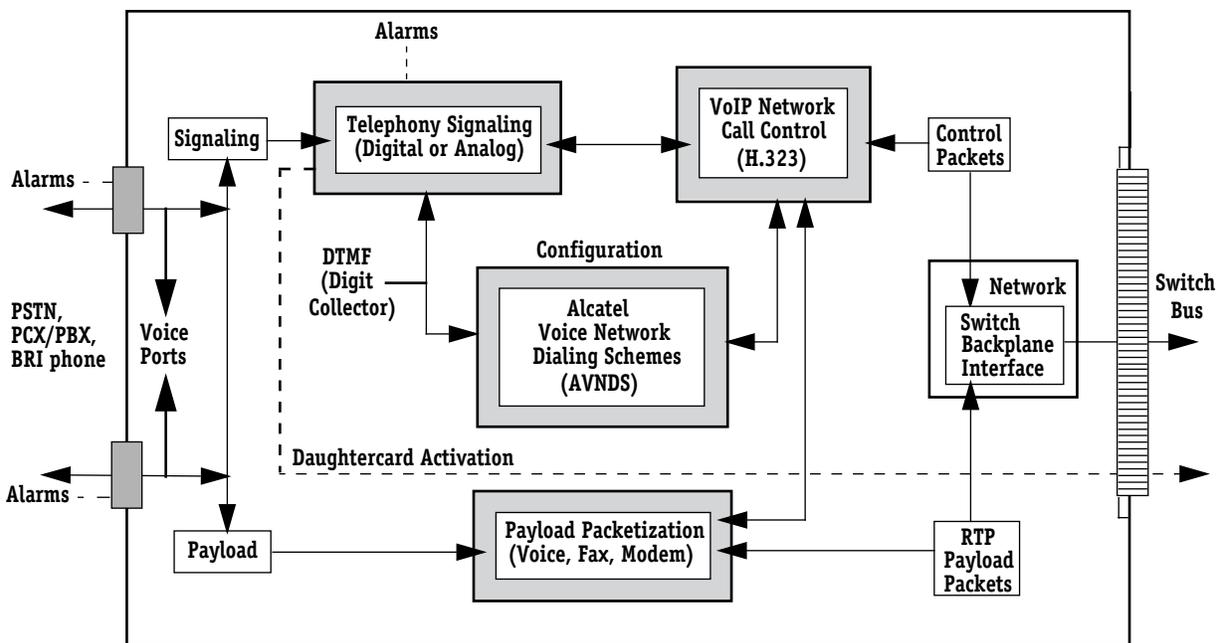
Elements of Converged Voice/Data Packet-Switched VoIP Network and Circuit-Switched PSTN

H.323 VoIP Gateway Voice and Convergence Features

As shown below, the main functions handled by Alcatel's H.323 VoIP Gateway include the following:

- Telephony Signaling — used to communicate with the PSTN or Customer Premises Equipment (CPE).
- Payload Packetization and Digital Signal Processing (via DSP) — converts PCM voice packets from circuit-switched network to H.323 packets on IP network and the reverse.
- H.323 Network Call Control — handles H.245 and H.225 packet processing, e.g., connect, disconnect.
- Alcatel VoIP Network Dialing Schemes (AVNDS) — handles conversions between phone numbers and IP address of H.323 devices.
- Network Switch Backplane Interface — connects H.323 VoIP gateways to switch and ultimately to IP network.

These functions can generally be divided into either voice or convergence features, based on the controls they provide over VoIP in the switch. For the most part, the voice features include separate controls for signaling and for voice interoperability, whereas, the convergence features encompass H.323 call control and voice/data interoperability via the use of AVNDS in IP networks.



VoIP Daughtercards and Enterprise VoIP Features

Signaling Control and Voice Interoperability (Voice Features)

The ability to accommodate voice traffic using VoIP switches installed in data networks is achieved by means of signaling controls and voice interoperability features. The VoIP signaling control and voice interoperability functions includes telephony signaling and payload packetization as described.

Telephony Signaling

Telephony signaling is used for signaling with telephone equipment, e.g., PBX, via the telephony interface, as well as to control the communication signaling between the H.323 VoIP gateway and the Customer Premises Equipment (CPE). It detects the presence of new calls, collects dialed digit information (telephone number in some form or another) entered by the caller to route a call via an AVNDS to its destination point, and is also used to detect the end of calls (off hook).

Telephony signaling provides call progress supervision by generating supervisory and call progress tones, as well as DTMF (Dual Tone Multi Frequency) tones for outbound calls. It also provides DSP (Digital Signal Processor) interfacing control, and transfer of PCM-based voice packets to and from the DSP subsystem or DIMM (DSP Interface Management Module). It coordinates with the DSPs to select voice coders (codecs) at startup when a particular vocoder is needed. When a call is received, telephony signaling is responsible for opening channels and PCM data streams to the DSPs to process the voice data.

The signaling controls provided by the Telephony Signaling functions includes the following:

- Call Progress Tone and Tone Detection — detects individual in-band frequencies and converts them into tones or other signaling events, e.g., answer or busy signals.
- Dialing Timers — used to time incoming signaling events, e.g., how long to wait for a wink start, or how long to wait for another digit.
- E&M Signaling (Common, Wink Start, Immediate Start and Delay Start) — customizes attribute settings or parameters to match CPE. (Available only on VSD-T1.)
- Foreign Exchange Station (FXS) Loop Start — customizes attribute settings or parameters to match CPE.
- Foreign Exchange Office (FXO) Loop Start — customizes attribute settings or parameters to match CPE.
- Caller ID — looks for Caller ID information, e.g., calling party telephone number between first and second ring.

The telephony signaling configuration options for telephone signaling interfaces, e.g., ring delay and cadence (ringing rhythm), are assigned to the physical ports on the daughtercard, including the T1 and E1 line specifications. All options are defined at the channel level.

Parameters for telephony signaling and VoIP network preferences are pre-configured in text-based configuration files referred to as **vsmbboot.asc** files. The parameters are stored in the boot files and subsequently assigned to a daughtercard and/or its components. For more details on setting these parameters, see also Chapter 5, “VoIP Commands.”

Payload Packetization and Digital Signal Processing

Payload packetization is responsible for conversion between time-continuous telephony (analog or digital payload) at the telephony interface and Real Time Protocol (RTP) packets on the data network interface. It supports voice compression, echo cancellation, Fax and DTMF Relay (demodulation/modulation), modem data transport (up to 14400 baud), voice activity detection and comfort noise generation, as well as packet arrival de-jittering.

Physically, the payload packetization function is implemented on the DSPs (DIMMs), with control and configuration on the Motorola MPC860 processor. Configuration is performed through the `vsmboot.asc` file on the switch. Upon VoIP daughtercard activation, the configuration is transferred from the switch to the daughtercard. See Chapter 4, “Setup and Installation,” for more information.

The controls for voice interoperability provided by the payload packetization functions include the following:

- Codecs (see also Coding Profiles -- H.323 Call Capabilities) — provides encoding/decoding of H.323 packets.
- Voice Echo Cancellers — reduces echo on voice conversations.
- Fax or Modem over IP — allows fax/modem calls to be transmitted via H.323.
- Voice Activity and Silence Detection — detects voice conversation (or lack thereof) to reduce H.323 bandwidth requirements.
- Comfort Noise and Jitter Buffer — generates slight background noise (white noise) on the voice conversation, so callers do not think the connection has failed.

Digital Signal Processors, or DSPs as they are more commonly known, are math-intensive coprocessors used to convert and manipulate information, especially in telecommunications systems (systems that transmit all types of data including voice and video). They are also programmable chips, well-suited for VoIP as DSPs have the ability not only to convert but to compress analog signals into various digital formats, i.e., perform digital signal processing.

Although DSPs do not have any direct analog input/output since they are actually digital devices, they can accept digitized analog data rather than raw analog signals. As a result, DSPs are used in the digital and analog VoIP daughtercards developed by Alcatel to bring switch-enabled VoIP to enterprises; however, before the digitized and compressed voice signals can be delivered as “voice data” in a VoIP network, they must be packetized into H.323 packets.

Packetized voice is digitized voice compressed into finite bit stream of IP packets, that carry the “voice payload” between remote and distant locations, across the IP network and make processing VoIP calls in IP networks possible. Once compressed and packetized, periodic delays (jitter) to make the call sound smoother must be imposed on the transmission of these packetized “voice” conversations to mimic “real time voice” (resonating by nature in continuous “analog” waveform). DSPs are used further to reduce the delays from conversion and compression to ensure quality voice communications without affecting the real time voice processing and compression that occurs simultaneously.

To transmit the compressed data (digitized voice) across the IP network, the Real Time Protocol (RTP) is used. RTP streamlines and then transports voice packets, including interactive multimedia packets over IP, although it does so without any guarantees or quality of service provisioning.

◆ Note ◆

H.323 VoIP telephone calls automatically receive the highest priority in the VoIP network via the Quality of Service ToS bit. For more information, see the switch manual.

Voice packet transmissions, or the “payload,” are expedited by engaging the User Datagram Protocol (UDP) for faster delivery, packets which by necessity include the IP network call transport header information. Resultant jitter caused by delays imposed on the payload packets upon arrival to their destinations is also handled by the DSPs.



Voice Packet Transmission

UDP is needed by RTP to keep pace with “Real Time Voice” but lacks controls and error checking capability.

DSPs can monitor calls in progress, detect voice activity and handle echo cancellation (the filtering of unwanted transmission signals as specified in ITU algorithm standards G.160 and G.126); comfort level background (white) noise can also be generated on either the transmitting or receiving end.

Since digital signal processing affects nearly every operation in VoIP, numerous DSPs are incorporated adjacent to the supporting MPC860 CPU signaling controller in the voice switching daughtercards (normally used with voice switching modules), comprising the core of Alcatel’s enterprise VoIP on the call processing end. The DSPs and the Motorola MPC860 controlling processor work in unison to support the various protocols and interfaces that implement the enterprise VoIP telephony functions contained in software on the voice switching daughtercards. In a nutshell, the DSPs are the voice processors, and the MPC860 controller is the data communications processor on the daughtercards. Altogether, the above components provide T1, E1 and ISDN voice and data synthesis processing, with scalable versions of each bringing enterprises any-to-any switching functionality that now, with enterprise VoIP, includes least-cost call routing for VoIP Virtual Private Networks (VPNs).

Signal Recognition

Initially, digital signal processing involves DSP detection of an array of voice signaling types using Channel Associated Signaling (CAS) repetitive circuit-state signaling protocols (for T1 and E1 lines). Many forms of call signaling exist to set up and end calls, most of which result in the ringing of a phone or connection of a fax machine. These forms entail newer line signaling methods that use digital pulses (PCM, or Pulse Code Modulation), analog touch-tones such as DTMF (Dual Tone Multiple Frequency), and other much older analog signals in all their assortments, including but not limited to: Ear & Mouth (E&M), Loop Start, Ground Start, Foreign Exchange Subscriber (FXS) and Wink Start. Each signaling method was developed through the years by the telephone industry to provide Plain Old Telephone Service (POTS).

E&M signaling, of which there are five interface types, is the most widely used method for connecting calls to PBXs, telephone switching systems which use channelized T1 or E1 lines to transmit signals and multiplex digitized voice. T1 robbed bit signaling is an example of narrow or in-band signaling — where signaling tones are passed along the same circuit as someone’s voice.

ISDN (Integrated Services Digital Network), on the other hand, is another type of signaling wherein voice transmissions are digitized then placed on separate broad or out-of-band channels (so signaling tones are not passed along the same circuit as someone’s voice). This prevents signaling or other intrusions into the calls, and usually provides faster transmission.

ISDN is a common protocol in the Common Channel Signaling (CCS) network architecture used for exchanging information between out-of-band signaling networks and telecommuni-

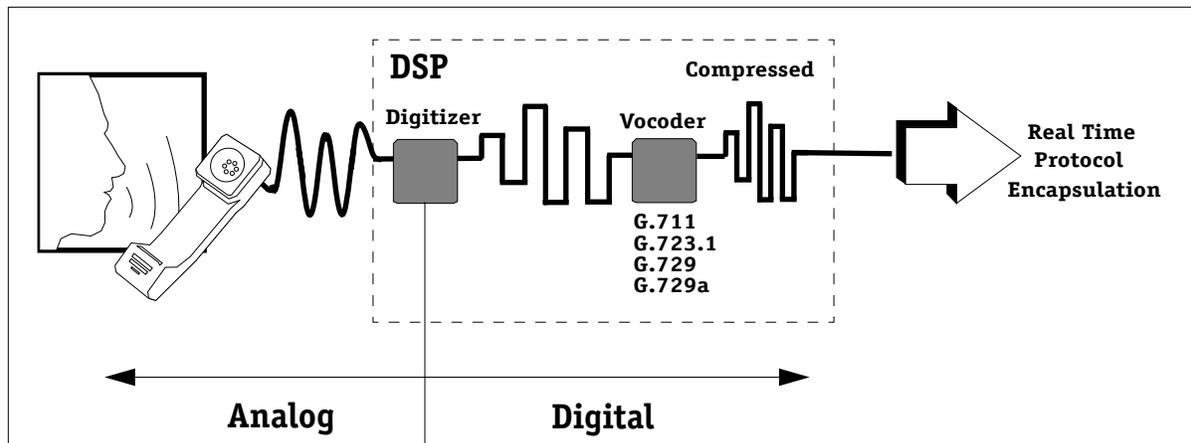
cations nodes in the network. ISDN does not use T1 (or DS-1) robbed bit signaling, where bits are taken from voice data to carry signaling. Alcatel H.323 gateways support in-band and out-of-band signaling.

Encoding

Once signaling types are determined they are analyzed and converted by the appropriate DSP voice coder (vocoder) into digital signals, which are ultimately converted and expanded back (re-modulated) into real voice. More specifically, after signal recognition and analysis, DSPs convert (encode) the amplitude of incoming analog signals into digital form using codecs, or CODer/DECoders.

The basic encoding schemes, or companding methods in use today, are for PCM which “encodes” analog signals into digital signals. Although the PCM companding methods used for T1, which follows Mu-Law, and E1 which follows A-Law, differ mainly in their algorithms, their purpose is much the same. (Companding is a contraction for compression and expansion.) However, A-Law and Mu-Law are incompatible. They use different methods, for example, to sample analog signals.

Next, the digitally encoded signals are compressed using industry standard vocoders. These are devices that use speech compression/decompression algorithms to analyze and convert analog waveforms into digital signals and reduce related bandwidth requirements.



Voice/Data Encoding and Call Compression

Compression

The appropriate vocoder used for VoIP calls is then negotiated by the H.323 VoIP gateway prior to call placement. As an added bonus, but with some variations in protocols, the same DSP technology that is used for voice compression also works with fax modems. (For that reason, it can be assumed that references to voice signal packets inherently include “fax” packets.)

The codecs and vocoders used in enterprise VoIP adhere to the ITU recommendations that fall under the H.323 IP network call control umbrella of interoperability standards for multimedia communications over packet-switched local area networks (part of the Series H Recommendations for Audiovisual and Multimedia Systems). The ITU’s H.323 suite of specifications includes the H.245 in-band call control specifications. For the signaling vocoders (G.711, PCM; G.723.1, Internet Speech; G.729/G.729a; Standard Telephone Speech), the algorithms in the Series G Recommendations for Transmission Systems and Media, Digital Systems and Networks are used.

H.323 Call Control and Network Interoperability (Convergence Features)

The ability to accommodate voice (also fax and modem) traffic compressed into data form via payload packetization for transport across data networks is achieved through the use of H.323 call controls and Alcatel VoIP Network Dialing Schemes (AVNDS) as described.

H.323 Network Call Control

H.323 network call controls are responsible for the procedures and protocols necessary to establish/tear down VoIP calls across the IP network. The VoIP gateway implements the H.323 network call control standards, which include the following:

- The H.225/Q.931 protocol that performs call establishment and tear down by establishing a reliable call signaling channel.
- The H.245 protocol that establishes a reliable H.245 in-band channel for communications between all endpoints or terminals, i.e., gateways, for capability exchange and other messages.
- The registration, admission and status (RAS) protocol that creates a RAS channel to carry RAS messages between an endpoint and a gatekeeper.
- The H.323 IP network call control standards that support multimedia communications over local area networks.

The H.323 network call controls and capability provided by the H.323 call control functions includes the following:

- H.323 gateway (VoIP Switch)
- H.323 gatekeeper (e.g., RADVision Server) (Discovery, Configuration and Operation)
- H.323 call capabilities (Coding Profiles or Codecs, Voice Network Delay Buffers)

The voice network configuration options include general network information, H.323, H.225, and H.245 configuration settings, gateway, gatekeeper and registration parameters, and Real Time Transport Protocol (RTP) session parameters that must be specified for IP network communications. Voice network call control parameters are configured at the VoIP daughter-card level.

Alcatel VoIP Network Dialing Schemes (AVNDS)

The AVNDS are responsible for the operations and configuration of the VoIP daughtercard and/or voice switching module, e.g, VSX. AVNDS are implemented on the Motorola MPC860 processor, and the switch.

AVNDS are responsible for providing the interface to configure and maintain all VoIP daughtercards (H.323 gateways) on the entire VoIP network. Additionally, both standard packet Management Information Bases (MIBs) and proprietary voice packet MIBs are supported.

The AVNDS are used to store information contained in VSM (Voice Switching Module) configuration boot files (**vsmboot.asc**) concerning the configuration of the VoIP network, particularly the following:

- Destinations (H.323 endpoints, H.323 local channel destinations)
- Phone Groups (e.g., strip digits and extensions)
- Numbering Plans (hunt methods and hunt groups)

H.323 VoIP Gateway Voice and Convergence Features

The AVNDS handle inbound/outbound calls routing to/from the VoIP network and local ports. AVNDS are also used to set up calls and translate IP addresses to telephone numbers, and can be used with, or in lieu of, H.323 VoIP gatekeepers.

VoIP configuration boot files and profiles simplify VoIP configuration of Alcatel's H.323 VoIP gateways (VoIP daughtercards) by using sets of pre-configured parameters that can be assigned to the various manageable components. Various configuration elements, e.g., profiles, have a user-defined name associated with it. VoIP daughtercard configurations are stored in the switch.

Destinations which consist of remote network and local calling gateways, including H.323 gatekeepers, allow Network Administrators to configure a destination IP address and its specific protocol. Local channel destinations are considered subdestinations. Destinations, which are appended to hunt methods, are configured at the daughtercard level.

Phone groups are used to indicate what telephone numbers are available. They also define digits to be stripped and forwarded. Phone groups are configured at the daughtercard level.

Voice numbering plans use hunt methods to arrange telephone lines so that when calls come into the network they will ring in a certain order. For example, to use PSTN fallback, all phone groups must be set up with the last group element indicating the local destination or gateway to fall back on when a call cannot be placed over the VoIP network.

Hunt methods in voice numbering plans are configured at the daughtercard level. Hunt methods dictate what to do if the first line tried is busy, i.e., hunt methods are used to track down lines in a certain order until an available line is located. Phone line destinations can be grouped as desired in user-defined groups, such as by divisions or departments, location, or some other meaningful grouping.

For information on setting up and using the AVNDS (Alcatel VoIP Network Dialing Schemes), see Chapter 3, "Network Dialing Schemes," Chapter 4, "Setup and Installation," and Chapter 5, "VoIP Commands."

Switch Backplane Interface

The switch backplane interface is responsible for the payload packet transport, and VoIP daughtercard management message transport between the VoIP daughtercard and the host switch. Physically, the interface consists of a 100-pin connector between the VoIP daughtercard and the motherboard. All functions of the H.323 VoIP gateway are implemented on the MPC860 controllers on the daughtercards, the VSX (OSR configurations only) and the switch.

VoIP Standards for Development

Alcatel's H.323 VoIP gateway is designed to function in accordance with the following IP Telephony and Internetworking standards currently available as briefly summarized below.

International Telecommunication Union (ITU) Standards

ITU-T technical, operational, and tariff recommendations are used for standardizing telecommunications on a worldwide basis. ITU H.323 IP network call control standards apply to VoIP. These standards define the major components, namely, Telephone Terminal Equipment, Gateways, Gatekeepers and Multipoint Control Units (MCUs) for H.323-based communications systems:

Series H.323 — Series H: Audiovisual and Multimedia Systems (Infrastructure of audiovisual services — systems and terminal equipment for audiovisual services). H.323: this standard is specifically concerned with recommendations for real time audio, video and/or data and facsimile transmissions over H.323, Packet-based Multimedia Communications Systems. In reference to Alcatel's enterprise VoIP, the series as a whole relates to gateway devices used in VoIP to handle audio, video, data and facsimile transmission over IP or packet networks. The standard specifically includes the newer ITU recommendations for the Internet facsimile protocol (T.38) that is used to 1) exchange messages and data between facsimile gateways connected via an IP network, and 2) message transport (depending upon bandwidth availability) using either TCP/IP or UDP/IP network protocols. (T.38 is incorporated into the first release of Alcatel's enterprise VoIP).

Series H.225 — Series H: Transmission of Non-Telephone Signals (Infrastructure of audiovisual services — Transmission multiplexing and synchronization. H.225: this standard is specifically concerned with recommendations for narrowband visual telephone services defined in H.200/AV.120-Series transmission paths for local area networks (LANs) providing non-guaranteed quality of service (QoS) which is less than that of ISDN PRI protection and recovery mechanisms. This recommendation describes how non-guaranteed QoS LANs provide conversational services for audio, video, data and control information in H.323 equipment. The series relates to gatekeeper devices used in VoIP to provide services across LANs. Gatekeepers are centralized network devices performing IP address translations and bandwidth management. Alcatel's H.323 VoIP gateway uses AVNDS to work with, and in lieu of, gatekeepers. (Note: Standard includes codec support for call synchronization.)

Series H.245 — Series H: Audiovisual and Multimedia Systems (Infrastructure of audiovisual services — Communications procedures), Control protocol from multimedia communication. H.245: this standard specifies syntax and semantics of terminal information messages, particularly receiving and transmitting capabilities, mode preferences from the receiver, including logical channel signaling, Control and Indication messages. Signaling acknowledgements are specified to ensure reliable audiovisual and data communications. The series relates to Multipoint Control Units used in VoIP to provide signaling and coding for call synchronization.

Codec Support (G.711, G.723.1, G.729a)

G.711 (PCM Encoding)

This is the ITU recommendation for an algorithm designed to transmit and receive A-Law and Mu-Law PCM voice at digital bit rates of 48, 56, and 64 Kbps. It applies to digital telephone sets on digital PBS (cellular) and ISDN channels. Support for this algorithm is required for ITU-T compliant videoconferencing (the H.320/H.323 standard).

A-Law and Mu-Law are processes needed to compand digital signals. A-Law is used in most countries except for the U.S., Canada and Japan where Mu-Law is more common. Companding is the process of compressing the amplitude range of a single signal, and then expanding them at the receiving end back to their original form. Although it is impossible to exactly reproduce an analog signal digitally, companding greatly improves the accuracy of this process. PCM uses two different companding processes. For this reason, PCM A-Law is used for international networks.

A-Law: The PCM coding and companding standard used in Europe and in areas outside of North America. A-Law encoding samples audio waveforms used in the 2.048 Mbps, 30-channel PCM system (E-carrier).

Mu-Law (E-Law): The PCM voice coding and companding standard used in Japan and North America. A PCM encoding algorithm where analog voice signals are sampled 8,000 times per second with each sample represented by an eight-bit value, and a raw 64 Kbps transmission rate. All sample bits are inverted before transmission.

◆ **Note** ◆

A-Law and Mu-Law are incompatible. For example, a signal sent with A-law cannot be received by a system using Mu-Law.

G.723.1

This is the ITU-T algorithm recommendation used for compressed digital audio over Plain Old Telephone Service (POTS) lines. It is the voice part of H.324 (POTS video conferencing). This algorithm runs at 6.3 or 5.3 kbps (20 bytes per 30ms interval) and uses linear predictive coding and dictionaries, which help provide smoothing. The smoothing process is CPU-intensive during real time based activities.

G.729a

This is the ITU's standard voice algorithm – CS-ACELP (Conjugate Structure Algebraic Code Excited Linear Predictive for the encoding/decoding of speech at 8 Kbps using conjugate-structure, algebraic-code excited linear predictive method. G.729 is supported by inter alia (among other things), American Telephone and Telegraph, France Telecom and Japan's Nippon Telephone and Telegraph.

VON (Voice on the Net) Developments

The VON (Voice on the Net) Coalition is concerned with developments in Internet Telephony and IP Telephony around the world. It is an incorporated, non-profit U.S. organization working with the government, business and other groups and individuals on regulations that affect this technology and its use. Alcatel's H.323 VoIP gateway was designed with the considerations of the VON Coalition in mind. Compression techniques and DSPs improve the quality of VON transmissions and minimize problems associated with IP packet delays.

VoIP and VLANs

Alcatel VoIP (VSD, VSB, VSA) modules *cannot* not be in a Virtual LAN (VLAN) with non-voice ports (i.e. data ports), IP phone ports, etc. All voice traffic *must* route in and out of the VoIP VLAN.

2 VoIP Daughtercards

Introduction

This chapter describes the voice switching daughtercards that can be installed in Alcatel switches to provide H.323 VoIP gateways in VoIP networks. Using ITU H.323 IP telephony standards, the H.323 VoIP gateway converts telephone or fax calls between the circuit switched Public Switched Telephone Network (PSTN) and packet-switched VoIP networks. Alcatel's H.323 VoIP gateways are typically used to handle VoIP calls as such placed across local and wide area networks between branch offices in remote enterprises, although the gateways are suitable for use in carrier applications, too. See Chapter 1, "VoIP Overview" for a more in-depth description of Alcatel's VoIP H.323 gateway operations.

Different VoIP daughtercards, as described below, are required depending on the telephony interface required to transmit and receive calls in the VoIP network. Furthermore, to digitize the VoIP calls, the daughtercards utilize digital signal processors (DSPs) containing a specified number of channels, which in turn determine the maximum number of calls that can be placed at one time on the card.

The VoIP daughtercards are referred to as voice switching daughtercards and, when installed in the switch, they are sometimes referred to as Voice Switching Modules (VSMs). Currently, the VoIP daughtercards can be installed in either the OmniAccess 512 or Omni Switch/Router. A blade installed in an OSR containing either one or two VoIP daughtercards of the same type is referred to specifically as a VSX or VSX switching module (for details see *VSX Switching Module* on page 2-25). For information on configuring either of these switches, refer to the appropriate switch user manual.

This chapter also depicts the port pinouts and jumper settings for all of the VoIP daughtercards where necessary, as well as the Deadman switch, and Cross-Over toggle switches available on certain voice daughtercards. The front panels for the VoIP daughtercards, including the front and bottom views of the cards, are shown to illustrate certain components relative to important operations of the H.323 VoIP gateway in the switch.

All VoIP daughtercards can be field-installed. For details on installing the cards, see also Chapter 4, "Setup and Installation." For details on configuring the switch to run VoIP, see Chapter 3, "Network Dialing Schemes" and Chapter 5, "VoIP Commands."

VoIP Daughtercard Types

There are two types of VoIP daughtercards: digital and analog. The digital voice switching daughtercards includes the VSD daughtercard used for digital calls placed in either North America and/or Europe, and the VSB daughtercard used specifically for digital calls placed in Europe. The analog voice switching daughtercard (VSA) is used only in North America for placing analog POTS (Plain Old Telephone Service) calls, e.g., to the PSTN. The basic VoIP daughtercards, which allow the switch to make these various types of phone connections, are listed and described below.

- VSDs (T1, or E1 QSIG and E1 ISDN PRI *Digital*) (North America and Europe)
- VSBs (Euro BRI ISDN *Digital*) (Europe)
- VSAs (*Analog*) (North America and Europe)

VSD — The digital voice switching daughtercards (VSDs) have two physical port connections which can be either T1 or E1 (called Dual T1 or Dual E1). Associated with each of the digital physical ports there can be either 48 channels for T1 connections, or 60 channels for E1 connections. The VSD card supports the following protocols or voice port interface connections for VoIP networks in North America: T1. The VSD card supports the following protocols or voice port interface connections for VoIP networks in Europe: E1 (QSIG) or E1 ISDN PRI (QSIG is another name for ITU Q.931). The VSD card does not support the following protocols: T1 ISDN PRI, T1 QSIG or Euro BRI ISDN (E1 ETSI). The VSD-60CH T1/E1 card is considered a high-end VoIP daughtercard as it provides the most channels. Reliable non-digitized voice processing is available only between two ports of the same interface type on a single daughtercard, and not between daughtercards. See *Voice Switching Daughtercard — Digital* on page 2-6 for more details. For more information on the voice port interface types for the digital VoIP daughtercards see also the digital port configuration commands in Chapter 5, “VoIP Commands.”

VSB — The other type of digital voice switching daughtercard is normally referred to as a VSB since it provides Euro BRI ISDN (E1 ETSI) protocol or interface port connections for VoIP networks in Europe. It differs mainly in that it has four ports, each with two (B) bearer channels and one (D) data channel. B-channels carry voice and data content, whereas D-channels are dedicated to carry control signals or call processing data for the B-channels. Each B-channel contains one DS0 voice channel. The VSB card does not support the following protocols: T1 ISDN PRI or T1 QSIG; T1 or E1 QSIG; E1 PRI or T1 BRI. See *Voice Switching Daughtercard — Euro BRI ISDN* on page 2-13 for more details.

VSA — The analog voice switching daughtercard (VSA) can contain an even number of analog ports from two to 16 depending on whether the card provides Foreign Exchange Station (FXS), e.g., telephone set (TelSet), or Foreign Exchange Office (FXO), e.g., Central Office (CO) port interface connections in an OmniAccess 512 or Omni Switch/Router. Each analog FXS port allows the connection of one off-the-shelf TelSet, or some other voice device, e.g., analog fax machine, analog phone answering machine, whereas each analog FXO port allows the connection of an FXO cable to a wall outlet or CO). See *Voice Switching Daughtercard — Analog* on page 2-19 for details.

◆ Note ◆

When used separately, the terms E1 and ETSI both entail European PRI and BRI interfaces. E1 ETSI used together as one term refers specifically to Euro PRI.

The table below shows the basic versions of the VoIP daughtercards, all of which were designed with various configurations in mind to fully support the wide range of features used in Voice over IP. See also Chapter 4, “Network Dialing Schemes,” for more configuration details. Also, not all configurations shown below may be currently available for purchase.

Voice Daughtercard		Description
VSD	VSD-12CH	Two (2) digital T1/E1 RJ-45 voice ports, 12 compressed voice channels
	VSD-24CH	Two (2) digital T1/E1 RJ-45 voice ports, 24 compressed voice channels
	VSD-36CH	Two (2) digital T1/E1 RJ-45 voice ports, 36 compressed voice channels
	VSD-48CH	Two (2) digital T1/E1 RJ-45 voice ports, 48 compressed voice channels
	VSD-60CH	Two (2) digital T1/E1 RJ-45 voice ports, 60 compressed voice channels
VSB	VSB	Four (4) digital E1 (Euro BRI) RJ-45 voice ports, TE (Terminal Equipment), NT (Network Terminator), Point to Point or Point to Multipoint NT, 8 compressed voice channels
VSA-FXS	VSA-4FXS	Four (4) analog RJ-11 voice ports
	VSA-8FXS	Eight (8) analog RJ-11 voice ports
VSA-FXO	VSA-2FXO	Two (2) analog RJ-11 voice ports
	VSA-4FXO	Four (4) analog RJ-11 voice ports
VSA-MIX	VSA-4FXS-2FXO	Four (4) analog FXS and (2) analog FXO RJ-11 voice ports

Digital Signal Processors (DSPs), DIMMs and Available Channels

All digital VoIP daughtercards have four vocoder channels available per DSP chip. DSPs are scalable in increments of three (in DSP DIMM modules) to better accommodate the needs of a VoIP network, and to reduce costs since the number of DSPs required is based on the number of simultaneous vocoder channels needed. DIMM stands for DSP Interface Management Module (*not* Dual Inline Memory Module).

Voice switching daughtercards, such as the VSD T1/E1 card, can contain up to 15 digital signal processors running at 100 MIPS (millions of instructions per second), providing voice processing functions for up to 60 DS-0s, or 60 Digital Service 0 channels (24 DS-0s are equal to one DS-1, or T1, channel).

The number of populated DIMMs (DSPs) on the digital voice daughtercards determines how many channels, which are bidirectional, are available on the card (e.g., 12 bidirectional channels per DIMM; or, a maximum of 60 bidirectional channels with 4 additional DIMMs). The number of simultaneous channels available on a particular digital voice switching daughtercard can be determined easily by counting the DIMMs on the card.

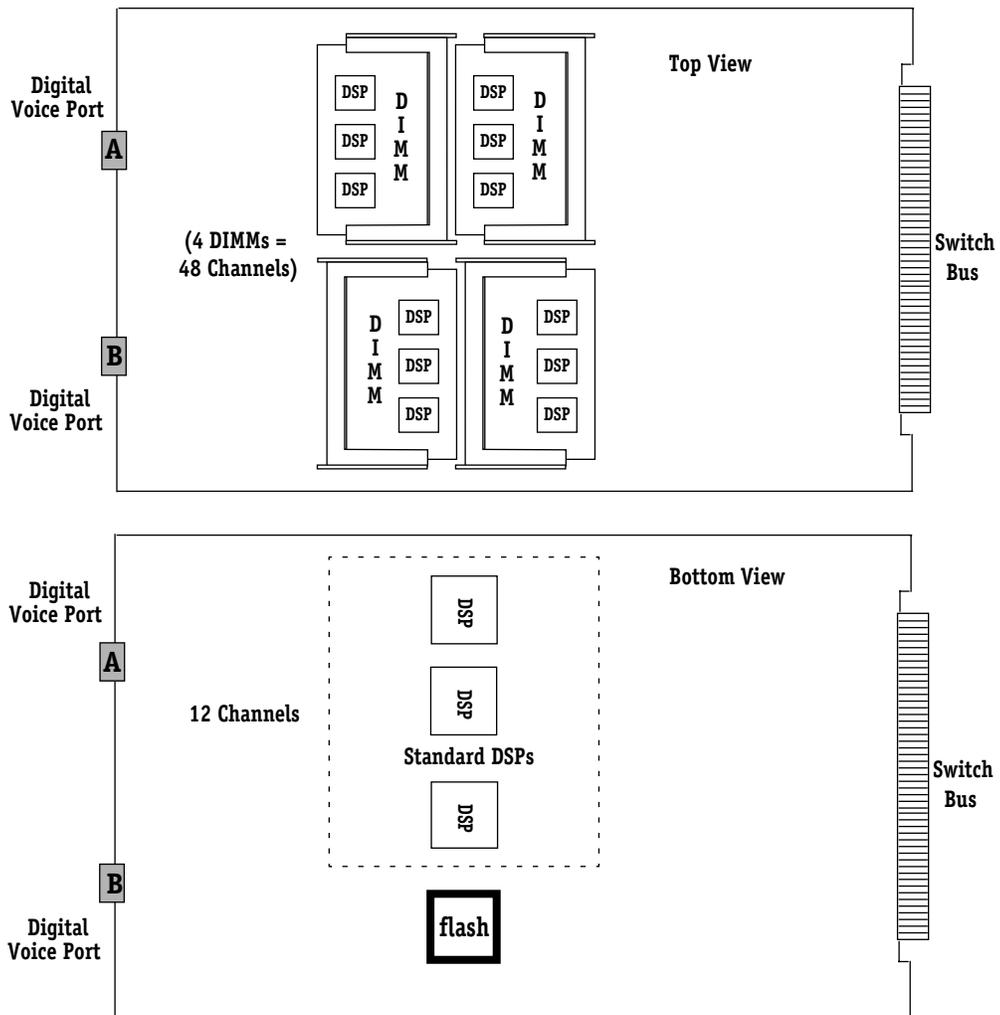
The illustration on the next page shows a VSD T1/E1 card fully-populated with standard DSPs and additional DIMMs. As a minimum configuration, three DSPs come standard on the digital cards to provide 12 channels (4 bidirectional channels per DSP); however, it is strongly recommended that only fully-populated (128 MB, 60 channel) VSD daughtercards be installed for the following reasons.

- Although a VSD with 12 channels (three std. DSPs) has two operational T1 ports that can provide up to 48 bidirectional channels, only the first 12 simultaneous calls can be handled per DSP; therefore, without additional DSPs (DIMMs) the 13th call and all subsequent calls will be ignored completely, i.e., no dial, busy signal or comfort noise will be generated, until either a channel becomes available, or additional DIMMs are installed.
- The installation of additional DIMMs, in effect, provides redundancy in the event of a DSP failure.

◆ Notes ◆

There are no DIMMs on the VSA daughtercard per se, only on the FXO or FXS grand-daughtercards with a maximum of one channel each per port. See *Voice Switching Daughtercard — Analog* on page 2-19 for more details.

DIMMs are *not* field upgradeable; however, the flash memory on the boards is field upgradeable. The flash memory must always match the image used or the daughtercard will not function properly if at all. Contact Alcatel's Customer Support for details on obtaining the appropriate flash and/or corresponding image upgrade.



Digital Voice Switching Daughtercard (VSD T1/E1) — DSPs/DIMMS Top and Bottom Views

Voice Switching Daughtercard — Digital

The digital Voice Switching Daughtercard (VSD) is used to provide digital telephone connections in Alcatel's H.323 VoIP gateways. There are two main types of digital voice switching daughtercards (VSDs) that can be used to provide VoIP: North American T1 or European E1 (QSIG or Euro ISDN PRI) and (VSBs) Euro BRI ISDN (E1 ETSI). Euro BRI ISDN; see also *Voice Switching Daughtercard — Euro BRI ISDN* on page 2-13. Each VSD contains two ports per daughtercard and up to 24 DS0 channels (T1) or 60 DS0 channels (E1) per port. A maximum of one daughtercard can be installed per OA-512 switch (see *VSD Front Panel* on page 2-7), and up to two daughtercards can be installed in a VSX in each available slot of an Omni Switch/Router.

All in all, there are five main daughtercard DSP/DIMM configurations on the VSD version of the digital cards:

- 12 channels (0 DIMM; only standard DSPs)
- 24 channels (1 DIMMs)
- 36 channels (2 DIMMs)
- 48 channels (3 DIMMS)
- 60 channels (4 DIMMS)

Each VSB contains four ports per daughtercard and two ISDN BRI B-channels and one D-channel per port. A maximum of one daughtercard can be installed per OA 512 switch (see *VSB Front Panel* on page 2-15), and up to two daughtercards can be installed in a VSX in each slot of an Omni Switch/Router. The VSB supports eight channels via two standard DSPs with four channels each, but does not support any add-on DIMM modules.

The OmniAccess 512 chassis provides one empty expansion slot (labeled as **S4**) reserved for use with features such as Voice Over IP (VoIP); it does not accept the VSX switching module used in Omni Switch/Routers. VSDs, VSBs and VSAs cannot be installed in the same slot in an OSR, and an MPX card is required in the OSR. See *VSX Switching Module* on page 2-25 for more information. Port numbers can vary depending on the VoIP switch configuration; see also *VoIP Daughtercard Port Numbering Schemes* on page 2-28.

All VSD ports are digital 8-pin, RJ-45 voice ports containing from one to 60 channels per port, depending on whether the voice port interface type is T1 or E1.

All VSD and VSB daughtercards require 32 MB Flash memory, and for OSR configurations, 64 MB DRAM memory on the MPX. For power requirements, see *VSX Switching Module* on page 2-25.

The following FCC Class B certifications for the VSD daughtercards have been obtained to date: OA-512-VSD-36T1, -36E1, -48T1, -48E1 and -60E1.

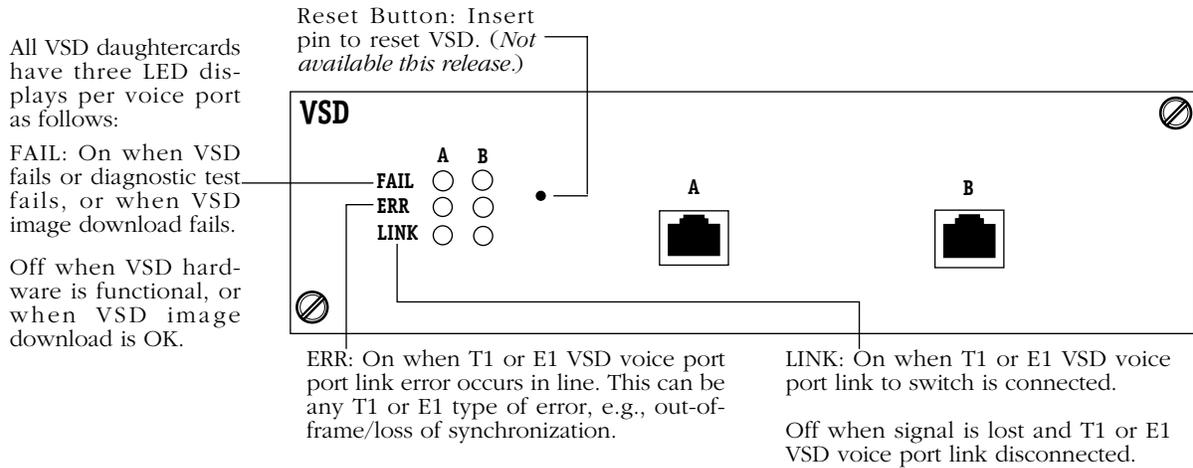
◆ Notes ◆

The number of simultaneous calls per card is dependent upon the number of available DSP channels. Two channels are used per call, e.g., with 12 channels six simultaneous calls can be connected.

Calls between channels on the same VSD or VSB card use PCM (Pulse Code Modulation) instead of the H.323 protocol to process digital calls, and require two DSP channels to make the calls.

VSD Front Panel

Each port has three corresponding LED indicators with link status displays as shown below.



VSD (Voice Switching Digital) T1 or E1 Daughtercard Front Panels

VSD Deadman Switch

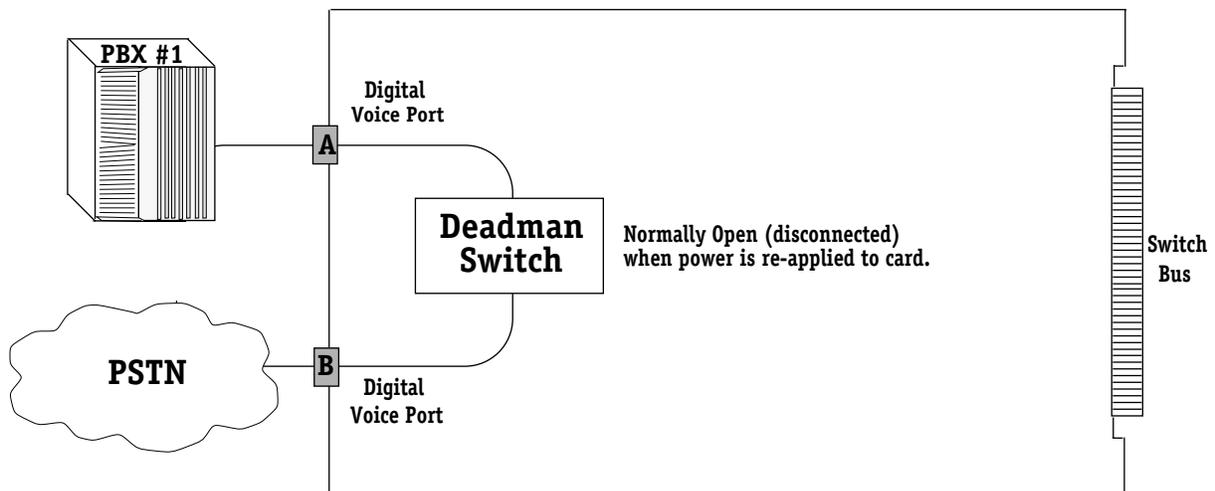
The two types of digital voice switching daughtercards (VSDs and VSBs) contain mechanical relay switches referred to as “Deadman” switches. The Deadman switch is a relay switch that allows two telephony ports using standard telephone connectors, such as the 8-pin RJ-45 jacks used for transmission lines, to be connected to each other in the event of a power failure until 1) power is re-applied to the daughtercard, and 2) the switch reboots to break the Deadman connection and allows VoIP calls to again be placed. For more information on the RJ-45 jacks, see *VSD RJ-45 Specifications* on page 2-11.

The Deadman switch, which resets after 200 ms, also contains a watchdog timer. Because the timer keeps the switch relays open when the Deadman switch disconnects the RJ-45s from each other, they can be connected immediately to the framers terminating the digital or analog telephone line. So, if power is lost to the VSD (or VSB), the Deadman switch keeps the PSTN connection alive by connecting the two telephony ports together before the signals reach the framers. In other words, for all new incoming calls, a connection is maintained between one port connected to a customer’s PBX and the second port connected to the PSTN, otherwise known as PSTN fallback.

No special configuration is required to use the Deadman switch on the VSD T1 cards; however, to use it on the VSD E1 cards, one port on the card must be set to be the **qmaster** and the other port must be set to be the **qslave**.

◆ Note ◆

On VSD E1 cards, setting both voice ports to **qmaster** (or both ports to **qslave**) will cause the two telephony switches to get alarms when the Deadman relay switch connects the two ports together. See the digital port configuration commands in Chapter 5, “VoIP Commands,” specifically the **voice port isdn protocol** command used to control the QSIG protocol settings.



Deadman Switch — PSTN Fallback Call Protection

VSD Cross-Over Toggle Switch

The Cross-Over toggle switch for digital voice daughtercards (VSDs only) can be used to correct communication link errors between a daughtercard in the switch and a PBX or keyset due to the transmit (TX) and receive (RX) pins of the cable connecting the VoIP daughtercard and the digital telephony device (PBX or other voice device). This will show up as the link LED not turning green (see *VSD Front Panel* on page 2-7).

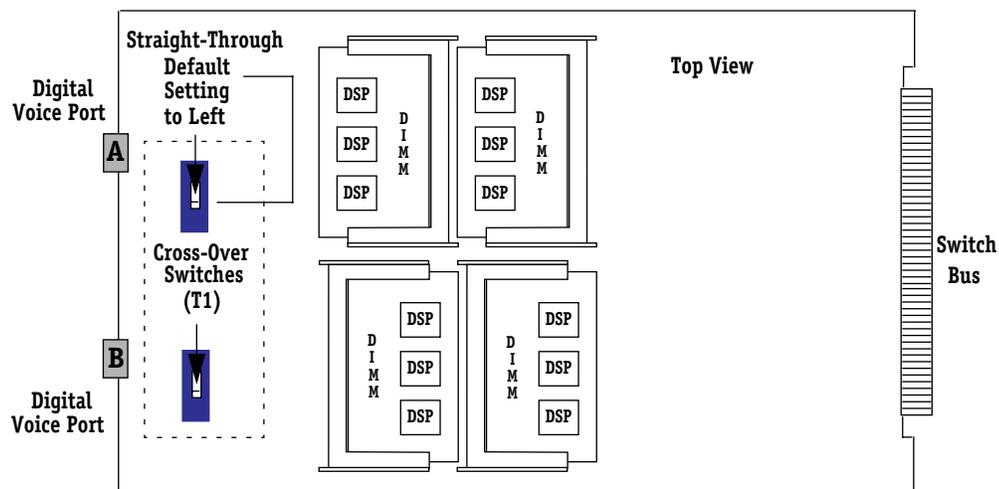
If a communication link error occurs between the switch and the PBX or Key Set as such, the blue Cross-Over toggle switch, as shown here on the top side of the board, can be flipped to a Cross-Over ON or OFF position after shutting down the VoIP switch and removing the affected daughtercard. This will swap the transmit and receive connections for the designated port. The default toggle position is to the left or OFF position.

Once the toggle switch has been flipped, the card can be reinstalled, and the ports on the voice daughtercard reconnected to the PBX or other voice device using either a Straight-Through or Cross-Over cable (*Straight-Through recommended*).

◆ Notes ◆

An amber cellophane tape may need to be peeled off the top of the Cross-Over toggle switch before the toggle switch can be flipped. When the tape is present it indicates the toggle switch is set to factory default.

The physical port always has 8 pins, but changes functionally depending on the cable in use. For more information on the RJ-45 jacks, see *VSD RJ-45 Specifications* on page 2-11.



Cross-Over Switches — Swapping Port Transmit/Receive Connections

Cabling

Of the common cable types (compatible with RJ-45 jacks) that can be used with VoIP switches, the Straight-Through (Ethernet) and Straight-Through (T1 Voice) are both acceptable, as well as the Cross-Over (T1) cable; however, due to the TX/RX pinout wiring, the Cross-Over (Ethernet) cable cannot be used with VoIP switches. If the Cross-Over (Ethernet) cable is used the LINK LED will not display.

◆ Note ◆

For E1 configurations, it is recommended that a balun connector always be used to connect a voice device (e.g., PBX) that uses an ITU G.703 interface (coaxial cables, BNC connectors) to a VSD (or any RJ-45 E1 port).

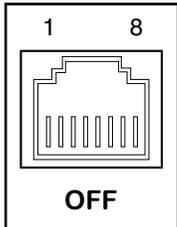
The balun converts the impedance of 120 Ohms on the RJ-45 port to 75 Ohms (G.723). The balun connector is not required when both ends have RJ-45 connections. Contact Alcatel's customer support for more details on balun connectors.

VSD Pinouts

The following illustration shows the pinouts for the digital voice switching daughtercard (VSD) 8-pin, RJ-45 jacks used to connect the voice ports on the card to voice devices in the VoIP network that support digital connections, e.g., PBX and Key Set.

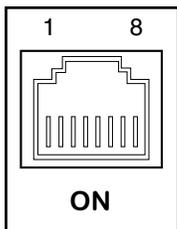
◆ **Note** ◆

The pinouts as shown indicate when the Cross-Over toggle switch is ON and OFF.



**Cross-Over
Toggle Switch**

VSD RJ-45 Specifications	
Pin Number	Standard Signal Name
1	Receive Data +
2	Receive Data -
3	
4	Transmit Data +
5	Transmit Data -
6	
7	
8	



**Cross-Over
Toggle Switch**

VSD RJ-45 Specifications	
Pin Number	Standard Signal Name
1	Transmit Data +
2	Transmit Data-
3	
4	Receive Data +
5	Receive Data -
6	
7	
8	

VSD Jumpers

The following jumpers are factory set on the VSD daughtercard and should not be changed by the customer unless under the direction of Customer Support. Note that, in general, only jumpers which can be set with shunts, or are associated with ports on the board are identified and described.

◆ Caution ◆

This information is being provided solely for the purpose of repositioning a shunt which may have been inadvertently removed, so as to prevent damage to the board, and/or possibly render the board or other components connected to the VSD inoperable.

Jumper No.	Shunt Position	Default	Port	Description
P3	no shunt (on Pins 1_2)	yes	B	RJ-45 connection
P3	no shunt (on Pins 3_4)	yes	A	RJ-45 connection
P7	no shunt (on Pins 1_2)	yes	B	RJ-45 connection
P7	no shunt (on Pins 3_4)	yes	B	RJ-45 connection
P7	no shunt (on Pins 5_6)	yes	A	RJ-45 connection
P7	no shunt (on Pins 7_8)	yes	A	RJ-45 connection
P10	Pins 1_2	yes	B	RJ-45 connection
P10	Pins 3_4	yes	A	RJ-45 connection

Other VSD Jumpers

Jumper No.	Shunt Position	Default	Description
P17	Pins 2_3	yes	Backplane interface (Hbus)

Voice Switching Daughtercard — Euro BRI ISDN

The Euro BRI voice switching daughtercard (VSB) is used to provide the European ISDN BRI voice port connections in Alcatel's H.323 VoIP gateways. Because many aspects of this card are similar to the VSD T1/E1 previously discussed; see also *VoIP Daughtercard Types* on page 2-2 and *Voice Switching Daughtercard — Digital* on page 2-6.

Digital Signal Processors (DSPs) and Available Channels

Unlike the VSD T1/E1 VoIP daughtercard shown previously, the VSB card does not support any additional DIMMs, only the two standard DSP which provide the VSB with eight channels. See also *Digital Signal Processors (DSPs), DIMMs and Available Channels* on page 2-4 for a more details.

VSB Deadman Switch

There are two Deadman switches on the VSB daughtercards. One Deadman switch is for ports A and B (1 and 2) and the other is for ports C and D (3 and 4); on a VSX switching module with two VSBs, the relays switches on the second card would be for ports A and B (5 and 6) and for ports C and D (ports 7 and 8). For more details on the Deadman switch, see *VSD Deadman Switch* on page 2-8. Port numbers can vary depending on the VoIP switch configuration; see also *VoIP Daughtercard Port Numbering Schemes* on page 2-28.

◆ Note ◆

For the deadman switch to operate properly on the VSB, ports 1 and 3 *must* be configured as TE, and ports 2 and 4 *must* be configured as NT (ports are TE, NT, TE, NT).

VSB NT (LT)/TE Cross-Over Toggle Switch

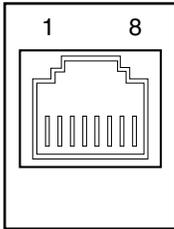
There are four NT (LT) / TE Cross-Over toggle switches on the VSB daughtercards. Each switch is factory set to **NT**. For more information on ISDN terminators, see *VSB Jumpers* on page 2-16, and also Chapter 3, "Network Dialing Schemes."

The NT (LT)/TE toggle switch is physically similar to the VSD Cross-Over toggle switch on the VSB, but functionally different as it is used to select either a network terminator or terminal equipment as an endpoint. For a description of the Cross-Over toggle switch on the digital voice switching daughtercards, see *VSD Cross-Over Toggle Switch* on page 2-9, and *Cabling* on page 2-10.

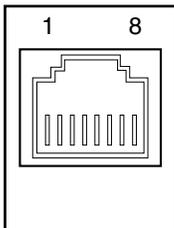
NT (LT) means that each NT port emulates the "network" side or "line terminator" point of the ISDN connections to the ISDN/PSTN network, e.g., connections to PBX, Key Set, BRI TelSet, Group 4 (ISDN) facsimile machine. TE means that each TE port emulates the "terminal" side of the ISDN connections to the ISDN network, e.g., PBX, Key Set, CO (Central Office) switch and ISDN telephone switch.

VSB Pinouts

The Euro BRI VoIP daughtercard (VSB) uses the 8-pin, RJ-45 jacks in the figures and tables on the following page to connect the voice ports on the card to voice devices in the VoIP network that support digital connections, e.g., PBX, Key Set, BRI TelSets, Group 4 (ISDN) facsimile machine, CO (Central Office) switch, and ISDN telephone switch. For more details on the VSB pinouts, see *VSD Pinouts* on page 2-11.



VSB Configured as TE RJ-45 Specifications	
Pin Number	Standard Signal Name
1	Unused
2	Unused
3	Tx +
4	Rx +
5	Rx -
6	Tx -
7	Unused
8	



VSB Configured as NT RJ-45 Specifications	
Pin Number	Standard Signal Name
1	Unused
2	Unused
3	Rx +
4	Tx +
5	Tx -
6	Rx -
7	Unused
8	

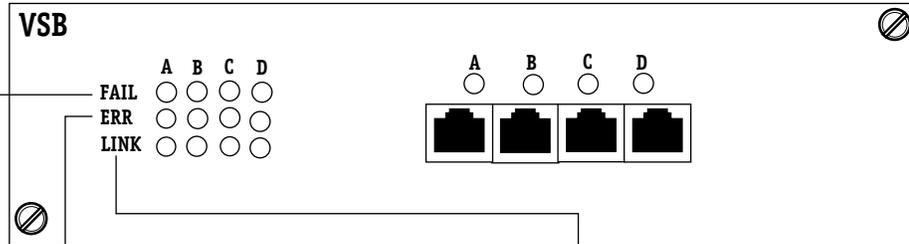
VSB Front Panel

Each port has three corresponding LED indicators with link status displays as shown in the following illustration. VSBs, VSDs and VSAs cannot be installed in the same slot in an OSR. An MPX must also be installed in the OSR. Port numbers can vary depending on the VoIP switch configuration; see also *VoIP Daughtercard Port Numbering Schemes* on page 2-28.

All VSB daughtercards have three LED displays per voice port as follows:

FAIL: On when VSB fails or diagnostic test fails, or when VSB image download fails.

LINK: Off when VSB hardware is functional, or when VSB image download is OK.



ERR: On when VSB voice port link error occurs in line. This can be any bearer or Data channel type of error, e.g., out-of-frame/loss of synchronization.

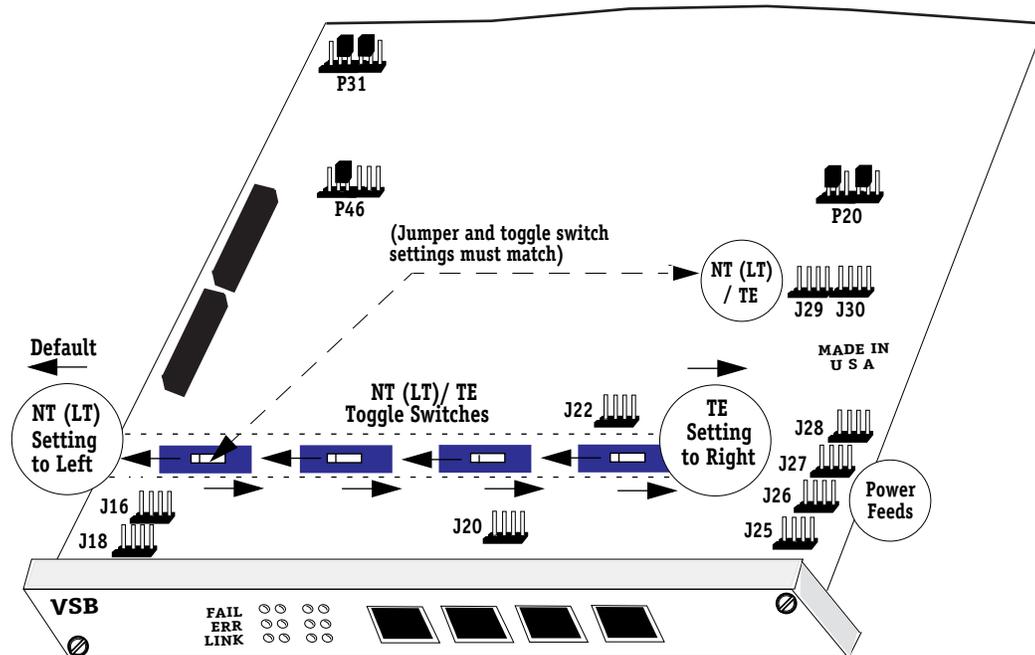
LINK: On when VSB voice port link to switch is connected.

Off when signal is lost VSB voice port link disconnected.

VSB (Voice Switching Digital) Euro BRI ISDN Daughtercard Front Panels

VSB Jumpers

The VSB daughtercard requires jumpers for Network Terminator (NT), also referred to as Line Terminator (LT), and Terminal Equipment (TE), impedance, and power feeds, be set as follows and in the order presented, e.g., NT (LT)/TE jumper/switch settings must be set on the VSB before any other jumpers on the board. In general, the jumpers are set on a per port basis. Locations of the jumpers are illustrated below.



VSB Jumper Settings — NT (LT)/TE, Impedance and Power Feeds

The jumpers are factory set on the VSB daughtercard, and should be changed by customers only under the circumstances as listed below for each jumper. Note that, in general, only jumpers which can be set with shunts are identified and described. Jumpers should be set in the order in which they are presented.

◆ Note ◆

All VSB daughtercards are factory set to the defaults, including but not limited to jumpers **J16** and **J18**, **J25** through **J28**, as well as **J29** and **J30**, and the corresponding **NT (LT)/TE** switches. This board is used only to provide Euro BRI ISDN (E1 ETSI) capabilities, and cannot be used in North America. See also *VSD Jumpers* on page 2-12.

Network or Line Terminator (NT/LT) / Terminal Equipment (TE)

The NT/LT and TE jumpers (**J29** and **J30**) are used to set the type of terminator on the VSB daughtercards. When the shunts are removed from these jumpers, NT (LT) is set as the terminator type; this is the default setting for both jumpers. When these jumpers and switches are set to either NT/LT or TE, the **voice port isdn protocol** command must also be set to the same, corresponding setting. For details on using this command, see Chapter 5, “VoIP Commands.”

◆ **Note** ◆

J29 and **J30** must match settings on **NT (LT)/TE** (Cross-Over toggle switches. The default setting for the toggle and corresponding jumpers is **NT** (see illustration for *VSB Jumpers* on page 2-16.)

Default (J29): **NT (LT)** on port 1; **no shunts** on **pins 1_2**.

Default (J29): **NT (LT)** on port 2; **no shunts** on **pins 3_4**.

Default (J30): **NT (LT)** on port 3; **no shunts** on **pins 1_2**.

Default (J30): **NT (LT)** on port 4; **no shunts** on **pins 3_4**.

TE (J29): TE on port 1; *requires shunts* on pins 1_2.

TE (J29): TE on port 2; *requires shunts* on pins 3_4.

TE (J30): TE on port 3; *requires shunts* on pins 1_2.

TE (J30): TE on port 4; *requires shunts* on pins 3_4.

For VSB jumpers **J29** and **J30**, only the following five NT (LT)/TE port configurations are allowed:

- All four ports can be configured as TE (ports are TE, TE, TE, TE).
- All four ports can be configured as NT (Ports are NT, NT, NT, NT).
- Ports 1, 2, and 3 configured as TE and port 4 configured as NT (ports are TE, TE, TE, NT).
- Ports 1 and 3 configured as TE, and port 2 and 4 configured as NT (ports are TE, NT, TE, NT).
- Port 1 configured as TE, and port 2, 3, and 4 configured as NT (ports are TE, NT, NT, NT).

Impedance

The impedance jumpers (**J16**, **J18**, **J20**, and **J22**) are used to set the resistance to the alternating current on the VSB daughtercards. When the shunts are removed from these jumpers, the standard impedance of 100 Ohms *will not* be fed to the designated port; this is the default setting for all four jumpers.

Default (J16): **no impedance** on port 1; no shunts on **pins 1_2** and **pins 3_4**.

Default (J18): **no impedance** on port 2; no shunts on **pins 1_2** and **pins 3_4**.

Default (J20): **no impedance** on port 3; no shunts on **pins 1_2** and **pins 3_4**.

Default (J22): **no impedance** on port 4; no shunts on **pins 1_2** and **pins 3_4**.

(J16): *add impedance* on port 1; *requires shunts* on *pins 1_2* and *pins 3_4*.

(J18): *add impedance* on port 2; *requires shunts* on *pins 1_2* and *pins 3_4*.

(J20): *add impedance* on port 3; *requires shunts* on *pins 1_2* and *pins 3_4*.

(J22): *add impedance* on port 4; *requires shunts* on *pins 1_2* and *pins 3_4*.

Power Feeds

The power feed jumpers (**J25**, **J26**, **J27** and **J28**) are used to set the power feeds on the VSB daughtercards. When the shunts are removed from these jumpers, *no power* is fed to the designated port; this is the default setting for all four jumpers.

◆ Caution ◆

Damage to the VSB daughtercards due to improper configuration of the power feeds is not covered by warranty.

For NT configurations only — The use of shunts to enable the power feeds on the VSB must be used with caution as they will substantially increase the voltage applied to the board, and may result in damage to the board and/or other components connected to the VSB. It is strongly recommended that Customer Support be contacted before installing shunts on jumpers **J25** through **J28**.

Default (J25): **no power** on port 1; **no shunts** on **pins 1_2** and **pins 3_4**.

Default (J26): **no power** on port 2; **no shunts** on **pins 1_2** and **pins 3_4**.

Default (J27): **no power** on port 3; **no shunts** on **pins 1_2** and **pins 3_4**.

Default (J28): **no power** on port 4; **no shunts** on **pins 1_2** and **pins 3_4**.

NT (J25): *power feed* on port 1; *optional shunts* on pins 1_2 and pins 3_4.

NT (J26): *power feed* on port 2; *optional shunts* on pins 1_2 and pins 3_4.

NT (J27): *power feed* on port 3; *optional shunts* on pins 1_2 and pins 3_4.

NT (J28): *power feed* on port 4; *optional shunts* on pins 1_2 and pins 3_4.

Other VSB Jumpers

The following jumpers are factory set on the VSB daughtercard and should not be changed by the customer unless under the direction of Customer Support.

Jumper No.	Shunt Position	Default	Description
P20	Pins 1_2, Pins 4_5	yes	Reserved
P31	Pins 2_3, Pins 4_5	yes	Backplane interface (Hbus)
P46	Pins 2_3	yes	Reserved

Voice Switching Daughtercard — Analog

The Analog Voice Switching (VSA) daughtercard is used to provide the analog voice port connections in Alcatel's H.323 VoIP gateways. Each VSA card includes either an FXS (Foreign Exchange Station) or FXO (Foreign Exchange Office) grand-daughtercard, or one of each depending on whether FXS signaling is needed to generate calls from POTS TelSets to a VoIP daughtercard, or FXO signaling is needed to generate calls to POTS TelSets from a VoIP daughtercard, or both. The FXS and FXO signaling protocols are used by the corresponding board types for which there are five VSA grand-daughtercard configurations:

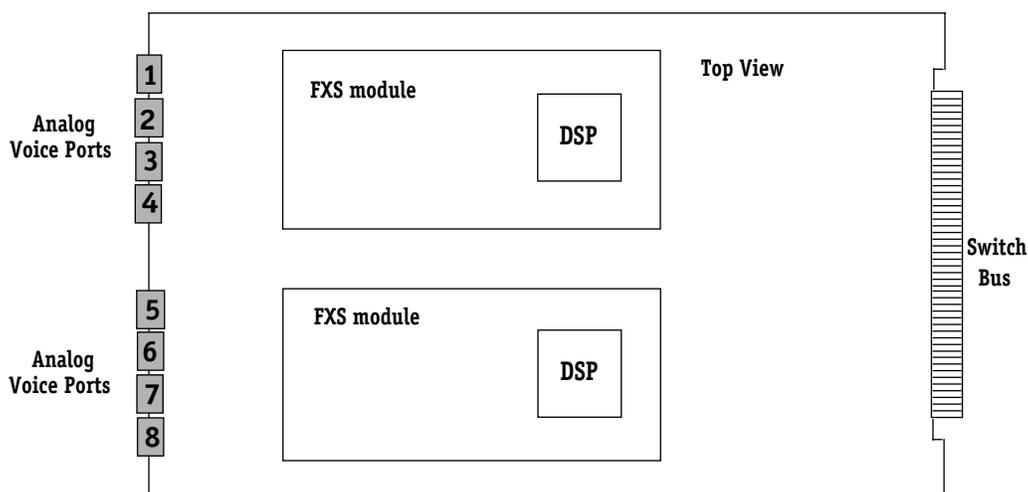
- FXS
- FXS/FXS (Dual)
- FXO
- FXO/FXO (Dual)
- FXS/FXO (Mixed)

Depending on the VSA FXS and/or FXO grand-daughtercards used there can be either 2, 4, 6, or 8 ports per daughtercard. As shown below, in a VSX-FXS-FXS (dual) configuration, up to eight POTS TelSets can be connected, and in a VSX-FXO-FXO configuration up to four POTS PSTN lines can be connected. FXS and FXO grand-daughtercards can be used together (referred to as a mixed configuration) on one VSA card.

Specifically, the VSA port range for the OmniAccess 512 is 1 to 2 (single FXO), 1 to 4 (dual FXO/FXO or single FXS), 1 to 6 (mixed FXS/FXO), or 1 to 8 (dual FXS/FXS grand-daughtercards). The same applies to the OSR except the range can be from 1 to 16 based upon the grand-daughtercards installed (OSR full capacity installations *not available this release*). An MPX must also be installed in the OSR. In general, VSAs, VSDs and VSBs are not be mixed in the same VSX card in the same slot of an OSR; however, there are exceptions. See also the *VSX Switching Module* on page 2-25 and the *VoIP Daughtercard Port Numbering Schemes* on page 2-28.

All VSA daughtercards require 32 MB Flash memory, and for OSR configurations, 64 MB DRAM memory on the MPX. For power requirements, see *VSX Switching Module* on page 2-25.

The following certifications for the VSA daughtercards have been obtained to date: FCC Class A for OA-512 and OSR VSA-FXO.



Analog Voice Switching Daughtercard (VSA-FXS-FXS, Dual) — Top View

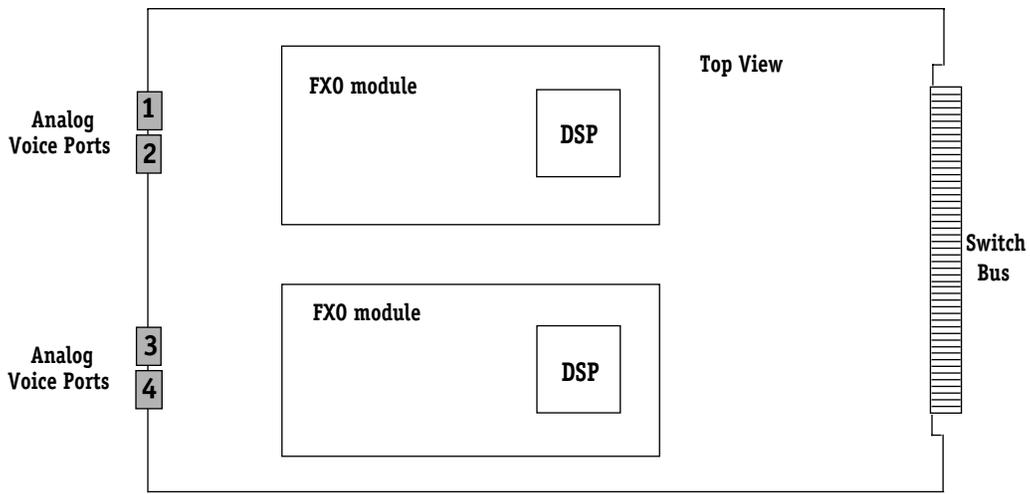
◆ Notes ◆

Analog voice switching daughtercards (VSAs) cannot be used without either installing an FXS or FXO grand-daughtercard (or module).

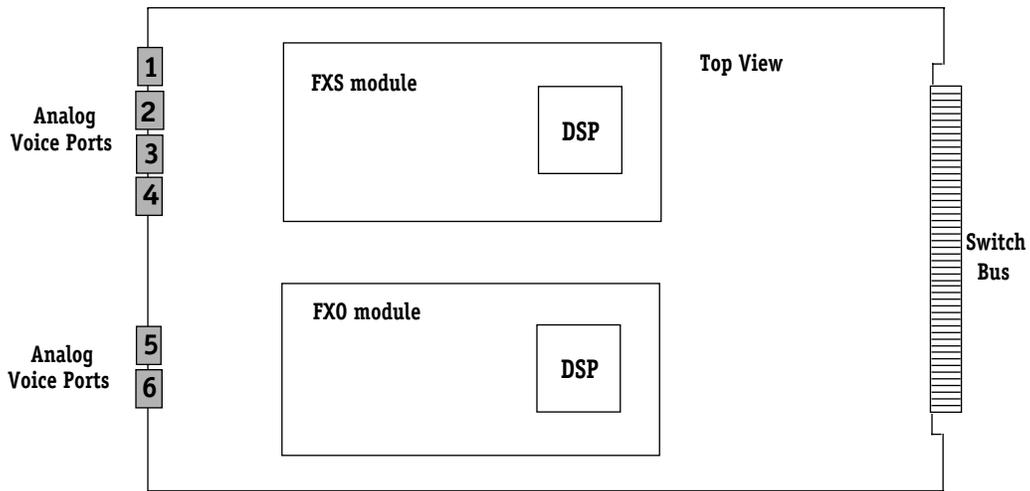
FXO grand-daughtercards cannot be installed as ports 1 and 2 when used with an FXS grand-daughtercard.

FXS and FXO grand-daughtercards are not field upgradeable.

Calls between channels on the same VSA card use PCM (Pulse Code Modulation) instead of the H.323 protocol to process analog calls, and require two DSP channels to make the calls.



Analog Voice Switching Daughtercard (VSA-FXO-FX0) — Top View



Analog Voice Switching Daughtercard (VSA-FXS-FX0, Mixed) — Top View

VSA Front Panel

Each channel (port) has one corresponding LED indicator with link status displays as shown in the following illustration. Port numbers can vary depending on the VoIP switch configuration; see also *VoIP Daughtercard Port Numbering Schemes* on page 2-28.

All VSA boards have one link LED per voice port.

All VSA Voice Port LEDs display as follows:

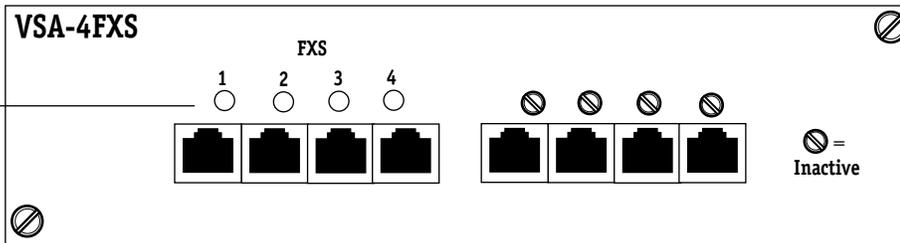
Green On: VSA port link to switch is connected.

Green Off: VSA port link is disconnected.

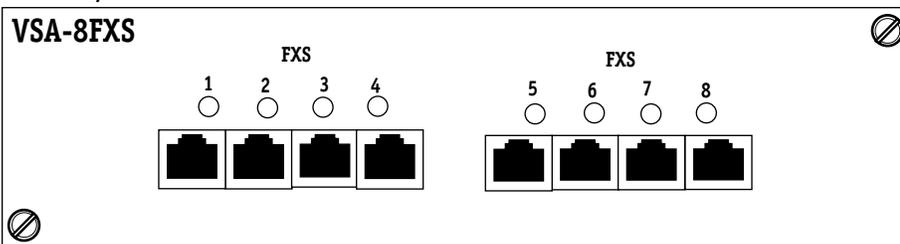
Green Blinking: Phone connected to VSA port ringing; No Blinking: Phone connected to VSA port off hook.

For the FXO submodules, only two voice ports are active.

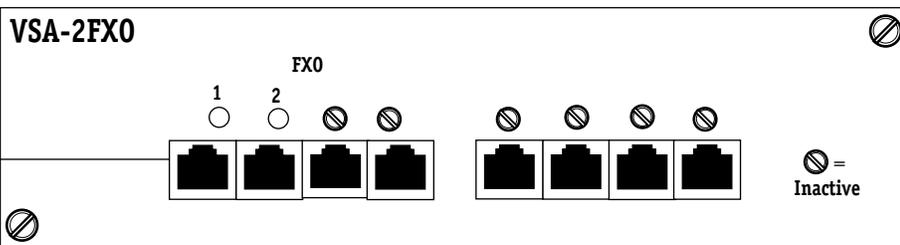
4 Ports / Single FXS module



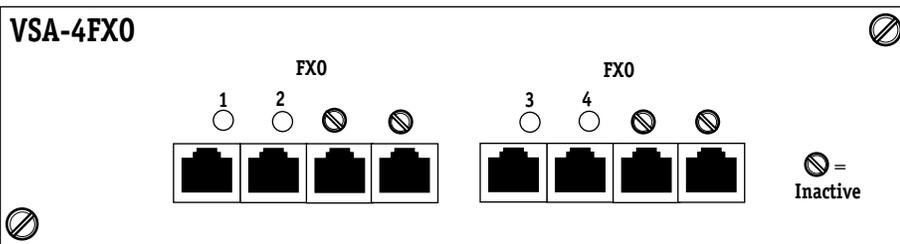
8 Ports / Dual FXS module



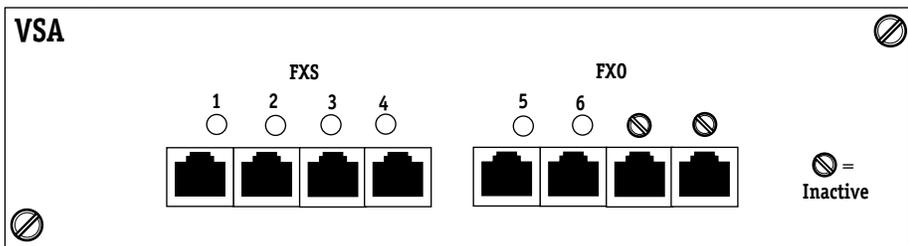
2 Ports / Single FXO module



4 Ports / Dual FXO module



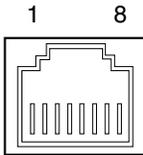
6 Ports / Mixed FXS and FXO module



VSA — Front Panels

VSA Pinouts

The following illustration shows the pinouts for the analog voice switching daughtercard (VSA) 8-pin, RJ-11 jacks used to connect the voice ports on the card to voice devices in the VoIP network that support analog connections, e.g., telephone and fax machine.



VSA TelSet RJ-11 Specifications	
Pin Number	Standard Signal Name
1	
2	
3	Ring
4	Tip
5	
6	
7	
8	

VSAs and Digital Signal Processors (DSPs), DIMMs and Available Channels

There are no DIMMs on the VSA, only DSPs on the FXS or FXO grand-daughtercards with a maximum of one channel per port. For more information on the DSPs, see *Digital Signal Processors (DSPs), DIMMs and Available Channels* on page 2-4 for more details.

VSAs and the Deadman Switch

Although VSA cards do not contain a Deadman switch, similar PSTN fallback call protection can be provided in the event of a power failure simply by plugging the analog VoIP switch into an Uninterruptable Power Supply (UPS). For a description of the Deadman switch on the digital voice switching daughtercards, see *VSD Deadman Switch* on page 2-8.

VSAs and Cross-Over Toggle Switches

There are no Cross-Over type toggle switches on the VSA daughtercards. For a description of the Cross-Over toggle switches on the digital voice switching daughtercards, see *VSD Cross-Over Toggle Switch* on page 2-9, and *VSB NT (LT)/TE Cross-Over Toggle Switch* on page 2-13.

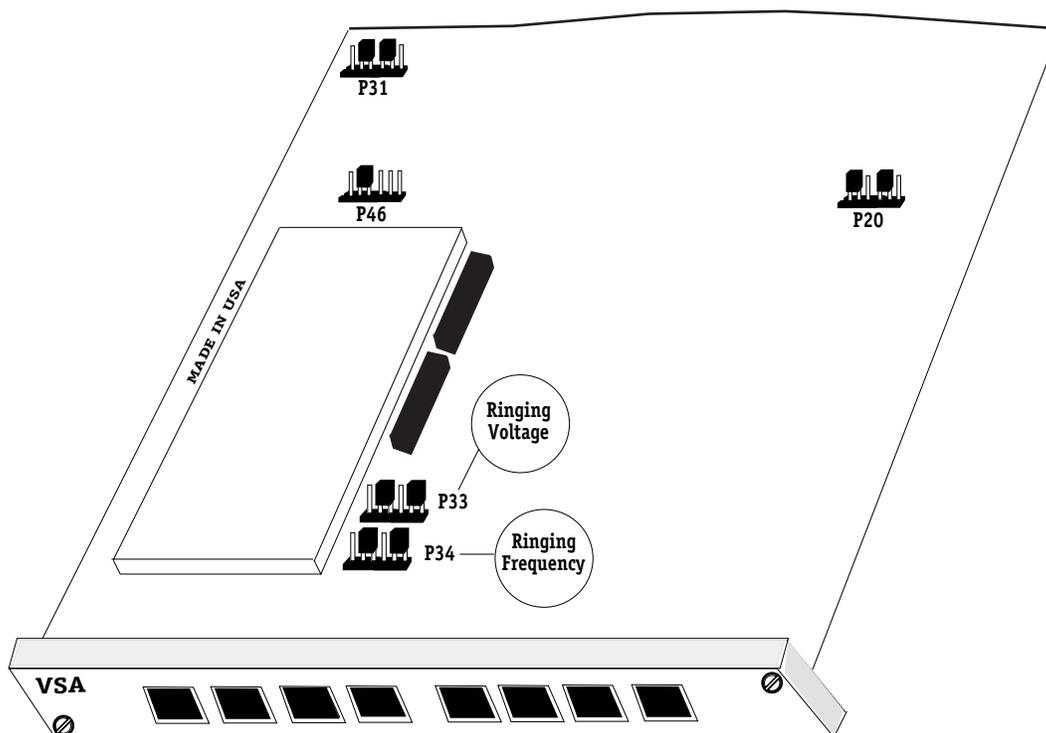
VSA Jumpers

The VSA daughtercard requires the jumpers for Ringing Voltage and Ringing Frequency to be set as follows. For additional details on the VSA ringing voltage or ringing frequency, see also Chapter 5, “VoIP Commands.” In general, the jumpers are set on a per port basis. Locations of the jumpers are illustrated below.

◆ Notes ◆

All VSA daughtercards are factory set to the defaults for jumpers **P33** and **P34**. The defaults are applicable to US and Europe versions of the board. See also *VSD Jumpers* on page 2-12.

Default jumper settings on VSA daughtercards can be used in Europe, but *ringing voltage and frequency must also match requirements as per specification of the equipment to be used with a VSA*, e.g., telephone, facsimile machine, terminal equipment (TE), PBX, etc.



VSA Jumper Settings — Ringing Voltage and Ringing Frequency

Ringing Voltage

The jumper for ringing voltage (**P33**) is used to set the continental ring tone in Vrms (voltage mean root square) on the VSA daughtercards.

Default: **75 Vrms**; requires shunts on **pins 2_3** and **pins 5_6** (No. Amer./European Spec.).

45 Vrms; requires shunts on **pins 2_3** and **pins 4_5** (European Specification).

86 Vrms; requires shunts on **pins 1_2** and **pins 5_6** (European Specification).

Ringling Frequency

The jumper for ringing frequency (**P34**) is used to set the frequency of the continental ring tone in Hertz (Hz) on the VSA daughtercards.

Default: **20 Hz**; requires shunts on **pins 2_3** and **pins 5_6** (No. Amer./European Spec.).

16 Hz; requires shunts on **pins 2_3** and **pins 4_5** (European Specification).

25 Hz; requires shunts on **pins 1_2** and **pins 5_6** (European Specification).

Other VSA Jumpers

The following jumpers are factory set on the VSA daughtercard and should not be changed by the customer unless under the direction of Customer Support.

Jumper No.	Shunt Position	Default	Description
P20	Pins 1_2, Pins 4_5	yes	Reserved
P31	Pins 2_3, Pins 4_5	yes	Backplane interface (Hbus)
P46	Pins 2_3	yes	Reserved

VSX Switching Module

As illustrated on the next two pages, the VSX switching module accepts up to two VoIP daughtercards, and can only be installed in the Omni Switch/Router as follows:

- A maximum of two (2) VoIP daughtercards can be installed per OSR switch.
- One VoIP daughtercard can be installed per VSX in a single slot.
- Two (2) VoIP daughtercards can be installed in **S3/1** (left) and **S3/2** (right) a VSX in a single slot.

With two digital daughtercards in a VSX in an OSR, for instance, it is possible to process up to 96 calls (T1) or 120 calls (E1) per VSX switching module. For port scalability, up to seven VSX switching modules with two VoIP daughtercards each of the same type can be installed per OSR (maximum capacity configuration *not available this release*). Pertinent specifications for the VSX switching module are as follows.

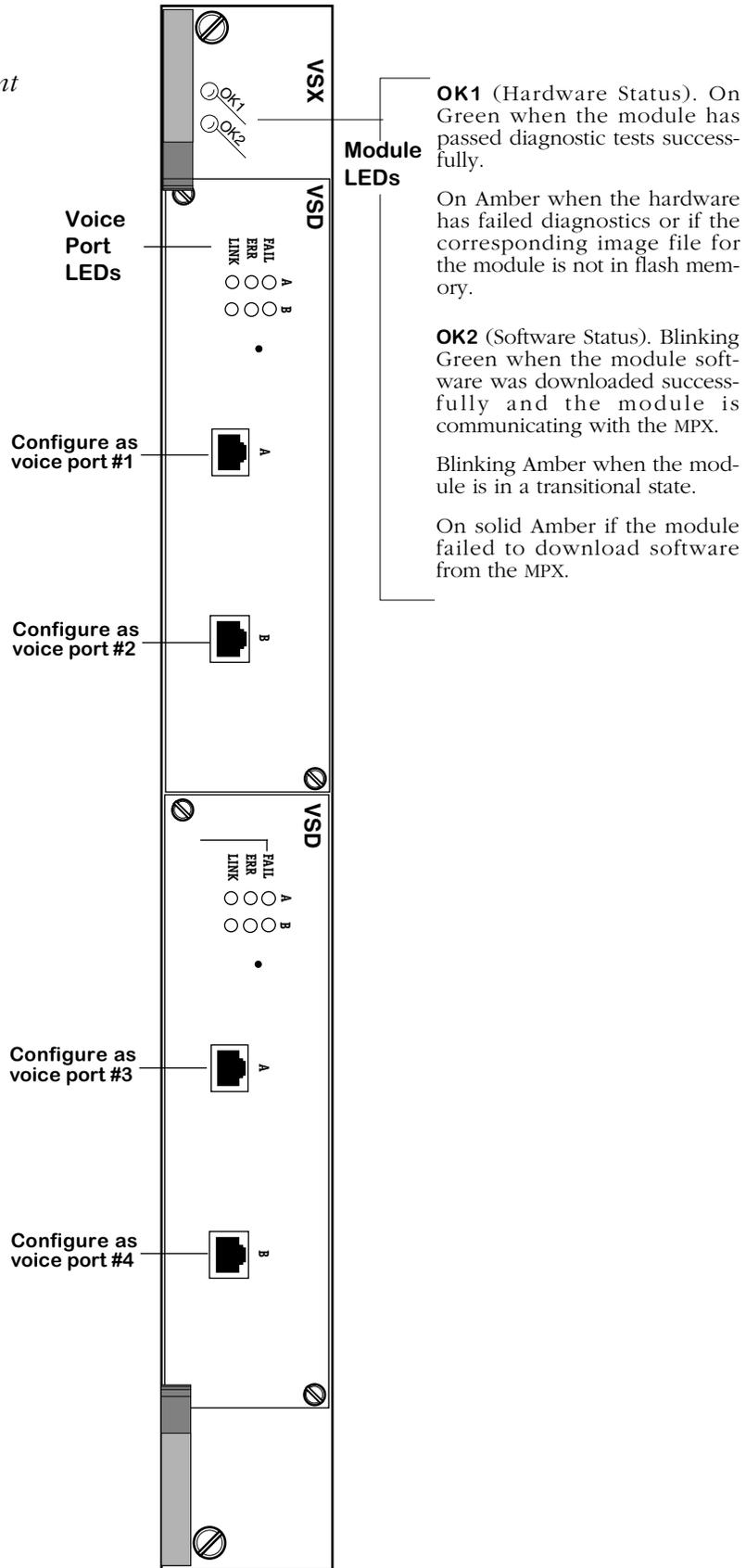
◆ Notes ◆

VoIP daughtercards cannot be installed in an HSX for interfacing with an OSR; however, two VoIP daughtercards (e.g., two VSBs) can be installed in a VSX (HSX-H) into an OSR with an MPX card, if necessary.

In OSR configurations, the use of an HRE-X device may be required. For more information, see the Omni Switch/Router user manual.

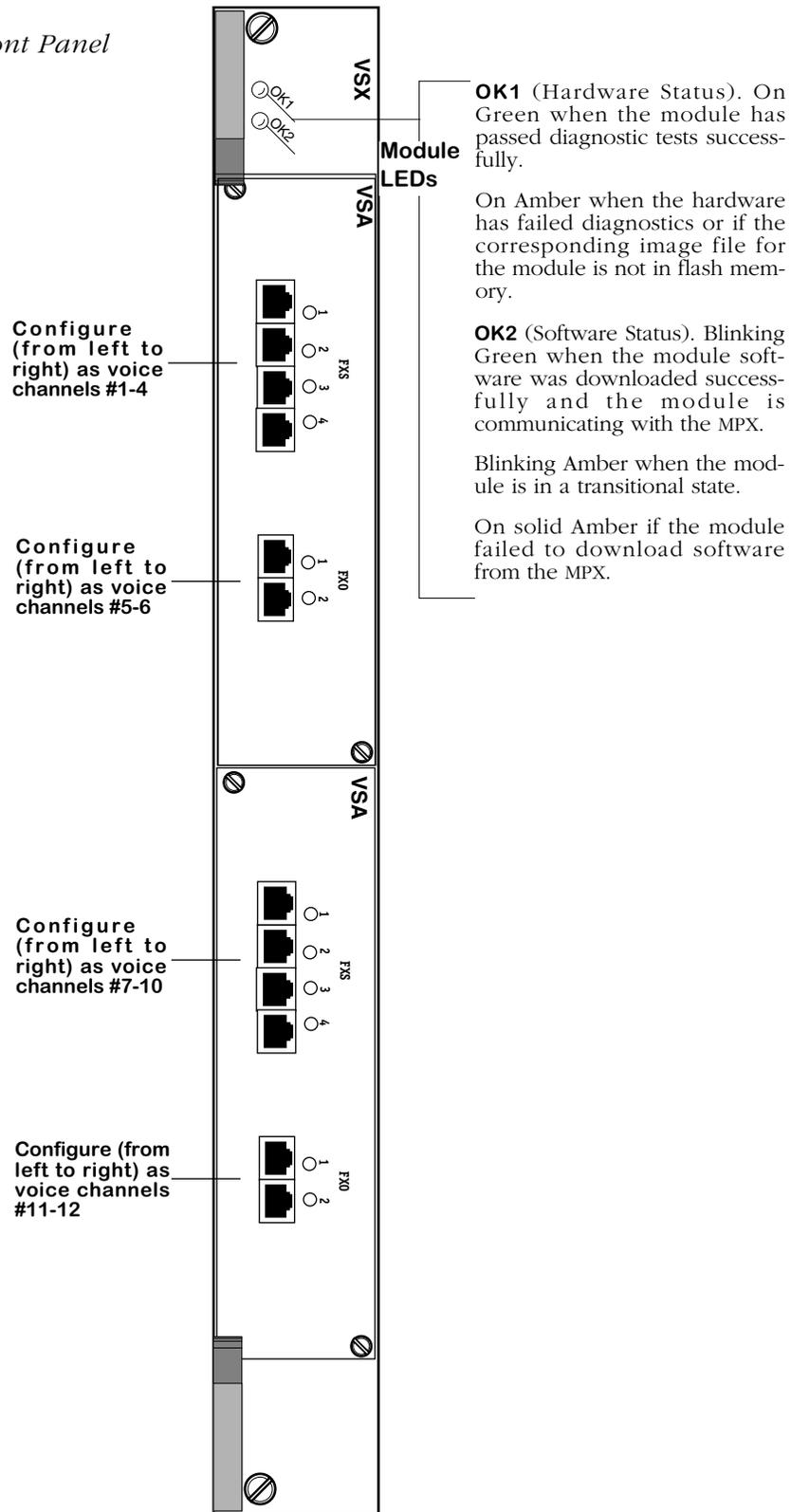
VSX Technical Specifications	
Data Rates Supported	1024 and 2048 Kbps
Clocking	Internal and External
MAC Addresses Supported	4096
H.323 IP Addresses Supported	2
Connections Supported	ISDN, QSIG master/slave, ETSI, FXS/FXO
Maximum Number of Simultaneous Voice/Data/Fax Channels Supported	120
Cable Supported	RJ-48C and Telset RJ-11
Power Consumption	5.50 amps Additional 2.0 amps per VSD or VSB Additional 3.55 amps per VSA daughtercard Additional 5.25 amps per VSA (with one FXS or FXO grand-daughtercard) Additional 7.0 amps per VSA (with two FXS or FXO grand-daughtercards)

See also *VSD Front Panel* on page 2-7



VSX Switching Module (with two VSDs)

See also *VSA Front Panel* on page 2-21.



VSX Switching Module (with two VSA-FXS-FX0, Mixed Modules)

VoIP Daughtercard Port Numbering Schemes

The following table is a representation of the port numbering schemes for all VoIP daughtercard configurations in either an OmniAccess 512 or an Omni Switch/Router. Although FXS and FXO installations must include a VSA daughtercard, for purposes of this table only, the VSA card is not specified; a maximum of two FXS or two FXO grand-daughtercards can be installed per VSA daughtercard. When determining valid port numbers, use this list as a reference:

- Each VSD has two ports.
- Each VSB has four ports.
- Each VSA FXS has four ports.
- Each VSA FXO has two ports.

◆ Notes ◆

OSR configurations are currently limited to a maximum capacity of two daughtercards per switch. Configuration options included one VXS module with two VoIP daughtercards, or two VXS modules and one VoIP daughtercard per VSX. Also, not all configurations shown below may be currently available for purchase.

See also *VSD Front Panel* on page 2-7, *VSB Front Panel* on page 2-15, *VSA Front Panel* on page 2-21; these drawings illustrate how VoIP daughtercard ports are numbered. For information on configuring the ports using CLI commands, see Chapter 5, “VoIP Commands”.

Switch	VSD	VSB	FXS	FXO	Valid Port/Configuration Numbers
OA-512	1	-	-	-	A-B (Note: Port numbers must be configured as 1 and 2; A=1 and B=2.)
OA-512	-	1	-	-	A, B, C, D (Note: Port numbers must be configured as 1, 2, 3 and 4; A=1, B=2, C=3, D=4.)
OA-512	-	-	1	-	1, 2, 3, 4 (FXS)
OA-512	-	-	2	-	1-4 (FXS), 5-8 (FXS)
OA-512	-	-	-	1	1, 2 (FXO)
OA-512	-	-	-	2	1-2 (FXO), 3-4 (FXO)
OA-512	-	-	1	1	1-4 (FXS), 5-6 (FXO)
OSR	1	-	-	-	A-B (Note: Port numbers must be configured as 1 and 2; A=1 and B=2.)
OSR	2	-	-	-	A-B (S3/1 position), A-B (S3/2 position); (Note: Port numbers in S3/1 must be configured as 1 and 2; A=1, B=2. In S3/2 port numbers must be configured as 3 and 4; A=3, B=4.)
OSR	-	1	-	-	A, B, C, D (Note: Port numbers must be configured as 1, 2, 3 and 4; A=1, B=2, C=3, D=4.)

VoIP Daughtercard Port Numbering Schemes

Switch	VSD	VSF	FXS	FXO	Valid Port/Configuration Numbers
OSR	-	2	-	-	A-D (S3/1 position), A-D (S3/2 position); (Note: Port numbers in S3/1 must be configured as 1, 2, 3, and 4; A=1, B=2, C=3, D=4. In S3/2 port numbers must be configured as 5, 6, 7, and 8; A=5, B=6, C=7, D=8.)
OSR	-	-	1	-	1-4 (FXS), (S3/1)
OSR	-	-	2	-	1-4 (FXS), (S3/1), 5-8 (S3/2)
OSR	-	-	3	-	1-12 (FXS-FXS-FXS)
OSR	-	-	4	-	1-16 (FXS-FXS-FXS-FXS)
OSR	-	-	-	1	1-2 (FXO), (S3/1)
OSR	-	-	-	2	1-2 (FXO), 3-4 (FXO), (S3/1)
OSR	-	-	-	3	1-2 (FXO), 3-4 (FXO), (S3/1), 5-6 (S3/2)
OSR	-	-	-	4	1-8 (FXO-FXO-FXO-FXO)
OSR	-	-	1	1	1-4 (FXS), 5-6 (FXO)
OSR	-	-	1	2	1-4 (FXS), 5-6 (FXO), 7-8 (FXO)
OSR	-	-	1	3	1-4 (FXS), 5-6 (FXO), 7-8 (FXO), 9-10 (FXO)
OSR	-	-	2	1	1-4 (FXS), 5-8 (FXS), 9-10 (FXO)
OSR	-	-	2	2	1-4, 5-6 (FXS-FXO), 7-10, 11-12 (FXS-FXO) or 1-8 (FXS-FXS), 9-12 (FXO-FXO)
OSR	-	-	3	1	1-4 (FXS), 5-6 (FXO), 7-14 (FXS) or 1-12 (FXS-FXS-FXS), 13-14 (FXO)

3 Network Dialing Schemes

Introduction

This chapter contains information on selecting and configuring Alcatel VoIP Network Dialing Schemes (AVNDS) which are used to translate dialed digits into IP addresses on the switch. At least one dialing scheme must be configured to support a Voice over IP network.

The dialing scheme examples discussed in this chapter are daughtercard centric, and typically consist of two PBXs with corresponding voice switching daughtercards each connected by one incoming only and one outgoing only trunk. In most cases the PBX is assumed to be trunked to the North American PSTN (Public Switched Telephone Network); however, some examples have voice daughtercards connected to the PSTN. It should be presumed also that all calls going to the PSTN are directed by Telco Central Offices. The WAN links between the switches (or some other device) are via the WSM or WSX modules which provide the ports, e.g., T1, E1, for data communications. The voice daughtercards (VSD, VSB and VSA) provide the telephony ports, e.g., T1, E1, Euro ISDN BRI, FXO and FXS for voice communications. Except where specified otherwise, it should be assumed that the VoIP daughtercards used in the examples are VSDs.

Variations to the dialing scheme configurations entail other likely scenarios in a VoIP network, including the use of hunt groups, site prefixes, strip digits, fax over IP and caller ID. Dialing schemes for special configurations, such as using VoIP in the switch with the OmniPCX 4400, are provided as well. All dialing schemes can be used in OmniAccess 512 and Omni Switch/Router configurations.

To simplify the configuration process, a VSM (Voice Switching Module) partial text-based ASCII configuration boot file (**vsmboot.asc**) has been created for each dialing scheme. Each partial boot file contains the specific CLI commands needed to implement a selected dialing scheme, and should be merged with the complete master boot file (**vsmboot_master.asc**), modified accordingly and then installed on the switch. Refer to Chapter 4, "Setup and Installation," for further details and an example boot file configuration. For specific details on the VoIP text-based command line interface (CLI) commands relative to the boot files and dialing schemes, see Chapter 5, "VoIP Commands".

◆ Caution ◆

Companies using Alcatel's VoIP feature are responsible for programming and testing all dialing schemes to reduce the likelihood or to eliminate the possibility of toll fraud from the PSTN and Emergency 911 processing.

The AVNDS consist of the following CLI command groups:

- H.323 endpoint destinations — these commands describe the IP address of every H.323 device in the H.323 VoIP network.
- H.323 local channel destinations — these commands describe every available channel on a voice daughtercard port in which to send calls.
- Phone Group — these commands describe every phone/fax number allowed on the H.323 VoIP network.
- Numbering Plan — these commands relate phone groups to destinations (H.323 endpoint and local channel destinations).

The AVNDS must include all CLI commands (approx. 20 commands) concerning voice destinations, phone groups and numbering plans to render a dialing scheme operational; however, some AVNDS will include other “non-AVNDS” commands in the matching **vsmboot.asc** files, as is the case with dialing scheme examples 12 and 17. For more details, see also Chapter 4, “Setup and Installation,” and Chapter 5, “VoIP Commands.”

How to Select a Network Dialing Scheme (AVNDS)

The tables below contain a list of the dialing scheme examples; use the decision criteria in the far right column of each table to determine the most appropriate dialing scheme to follow when configuring the network for VoIP in the switch. The dialing scheme examples, of which there are 26, are discussed in this chapter in numerical order, but are categorized into three distinct types:

- **VoIP Networks without PSTN** (Dialing Schemes 1-12)

Dialing scheme examples in this group do not connect to the PSTN. It is assumed that the PBX handles the routing of the call to the VoIP network. The first two examples are considered basic dialing schemes, while the remaining examples in this group demonstrate more complex VoIP dialing scheme concepts, such as how to use hunt groups (to multiply and split T1 lines), strip digits, or an H.323 gatekeeper.

- **VoIP Networks with PSTN** (Dialing Schemes 13-18)

Dialing scheme examples in this group connect the voice daughtercards to the North American PSTN, and cover the use of strip digits, fax over IP, and caller ID (forwarding and static). *International (ISDN) PSTN and Caller ID Forwarding not available this release.*

- **VoIP Networks with Interoperability** (Dialing Schemes 19-26)

Dialing scheme examples in this group allow VoIP networks to work with other functionally related equipment including H.323 gateways, H.323 endpoints, the OmniPCX 4400 and assorted PBXs.

No.	Dialing Scheme Examples / VoIP Networks without PSTN	Decision Criteria
1	Four Digit Extensions and Two Voice Daughtercards	Basic VoIP Network
2	Four Digit Extensions and Three Voice Daughtercards	Expanded VoIP Network
3	Hunt Groups — One Hunt Group (48 channels across two T1s)	One Hunt Group Per T1 Voice Daughtercard
4	Hunt Groups — One Hunt Group (60 channels across two E1s)	One Hunt Group Per E1 Voice Daughtercard
5	Hunt Groups — One Hunt Group (96 channels across four T1s)	One Hunt Group Across Two Voice Daughtercards
6	Hunt Groups — One Hunt Group (144 channels across six T1s)	One Hunt Group Across Three Voice Daughtercards
7	Hunt Groups — Four Hunt Groups (12 channels per group)	Fractional (1/2) T1 Hunt Groups
8	Hunt Groups — 48 Individual Hunt Groups (One channel per group)	Fractional (individual channel) T1 Hunt Groups
9	Strip Digits — Trunk Groups and Mixed Length Extensions	Unique mixed length extensions
10	Strip Digits — Trunk Groups and Two Strip Digits	Common extensions, Unique, two digit site prefix
11	Strip Digits — Trunk Groups and Eleven Digit Extensions (NANP-like)	Common extensions Eleven digit local extensions to simulate NANP dialing
12	H.323 Gatekeeper	Unique extensions Complex VoIP Network with H.323 gatekeeper

No.	Dialing Scheme Examples / VoIP Networks with PSTN	Decision Criteria
13	North American PSTN — Four Digit Extensions and Direct Inward Dial (DID)	Unique NANP extensions Unique Site Prefix DID
14	International (ISDN) PSTN — Four Digit Extensions and Direct Inward Dial (DID).	No. Amer and Intl. Sites DID
15	North American PSTN — Eleven Digit Extensions	Eleven Digit NANP Extensions with PSTN
16	North American PSTN — Fax over IP Network	Toll-Saving Fax Calls
17	North American PSTN — VSD/VSA Mixed	Mixed Digital and Analog Voice Daughtercards. PSTN at remote site.
18	North American PSTN — Caller ID (Static).	Analog Voice Daughtercard generating predetermined static caller ID

No.	Dialing Scheme Examples / VoIP Networks with Interoperability	Decision Criteria
19	H.323 Gateway — Microsoft NetMeeting (w/o FastStart)	VoIP to 3rd Party H.323 Software
20	H.323 Gateway — Cisco Routers	VoIP to 3rd Party H.323 Hardware
21	H.323 Gateway — OmniPCX 4400	VoIP to OmniPCX LIOE card
22	Omni PCX 4400 — E1 QSIG	Interoperating via E1 QSIG
23	Omni PCX 4400 — Euro PRI	Interoperating via Euro PRI
24	Other PBXs — T1	Interoperating with 3rd Party PBX via T1.
25	Other PBXs — Euro BRI	Interoperating with 3rd Party PBX via Euro BRI (E1 ETSI)
26	Other PBXs — European VSD/VSA Mixed	Interoperating with European Digital and Analog Voice Daughtercards.

All dialing schemes in this chapter can be modified to be used with the VSD, VSB and VSA voice switching daughtercards with the following exceptions:

- Dialing schemes No. 7 and 8 (Fractional T1 Hunt Groups) apply only to VSD and VSB daughtercards. Fractional type hunt groups do not apply to VSAs because analog channels can only be combined, not multiplied or split.
- Dialing schemes No. 17 (Mixed Digital and Analog) and No. 18 (Static Caller ID) apply to North American configurations using VSA daughtercards. Dialing scheme No. 26 (Mixed Digital and Analog) applies to a European VSA configuration.
- Dialing scheme No. 25 applies only to VSB daughtercards (European only).
- Except for when local channel destinations are used, all AVNDS commands function with the H.323 endpoints, e.g., OmniPCX, Cisco Routers, Microsoft NetMeeting.

Network Dialing Scheme VoIP Features

The table below lists dialing plans that use particular VoIP features. (See also *AVNDS Master List of Features by CLI Command* on page 3-67.) Descriptions of the dialing schemes in this chapter are intended to serve as guidelines in the development of enterprise-specific network VoIP dialing schemes.

VoIP Feature	Dialing Scheme
H.323 gateway to voice daughtercard (A)	All
H.323 gateway to H.323 gatekeeper (RADVision) (B)	12
H.323 gateway to H.323 device (C)	19, 21
Local channel — 48 individual hunt groups (One channel per group) (D)	8, 17, 18
Local channel — four hunt groups (12 T1 channels per group) (E)	7
Local channel — two hunt groups (24 T1 channels per group) (F)	1-3, 5, 7, 9-16, 19-24
Local channel — one hunt group (48 channels across two T1s) (G)	5, 6
Local channel — one hunt group (60 channels across two E1s) (H)	4
Site prefix — no site prefix (I)	1-9, 12, 14, 15, 19-24
Site prefix — single or multiple digits (J)	9-11, 13-19, 21
Voice phone group type — three digit local extensions (K)	9, 10
Voice phone group type — four digit local extensions (L)	1-8, 11, 12, 20
Voice phone group type — eleven digit local extensions (M)	11, 19, 21-24
Voice phone group type — NANP extensions (N)	13-18
Voice phone group type — INTL extension (O)	14, 22, 23
Voice phone group type — PSTN NANP (P)	13-18
Voice phone group type — PSTN International (INTL) (Q)	25
Strip digit length — no strip digits (R)	1-9, 12, 17-24
Strip digit length — 2 (T)	10
Strip digit length — 7 (V)	11, 13-16, 19-24
Digital Interface type — T1 (W)	1-3, 5-16, 19-24
Digital Interface type — E1 (QSIG) (X)	22
Digital Interface type — E1 ISDN PRI (Euro PRI) (Y)	4, 14, 23
Digital Interface type — Euro BRI (Z)	25
Digital Interface type — FXS (AA)	17, 18
Digital Interface type — FXO (AB)	17, 18

Trunk List for AVNDS Examples

The trunk list below is also provided as a general reference guideline to each of the diagrams used in the dialing scheme examples. Note that most examples use only one or two PBXs, and one or two voice daughtercards.

Lines **(A)**: Telephone/Fax lines off of PBX #1.

Lines **(B)**: Telephone/Fax lines off of PBX #2.

Lines **(C)**: Telephone/Fax lines off of PBX #3.

Trunk **(D)**: Inbound trunk to PBX #1.

Trunk **(E)**: Inbound trunk to PBX #2.

Trunk **(F)**: Inbound trunk to PBX #3.

Trunk **(G)**: Inbound trunk to Voice Daughtercard #1.

Trunk **(H)**: Inbound trunk to Voice Daughtercard #2.

Trunk **(I)**: Inbound trunk to Voice Daughtercard #3.

Trunk **(X)**: Inbound trunk to Voice Daughtercard #4.

Trunks **(J)**, **(K)**: Inbound/Outbound trunk (to Voice Daughtercard #1).

Trunks **(L)**, **(M)**: Inbound/Outbound trunk (to Voice Daughtercard #2).

Trunks **(N)**, **(O)**: Inbound/Outbound trunk (to Voice Daughtercard #3).

Trunks **(P)**, **(Q)**: Inbound/Outbound trunk (to Voice Daughtercard #4).

Trunk **(R)**, **(S)**: Inbound/Outbound trunk (between Voice Daughtercard #1 and PSTN)

Trunk **(T)**, **(U)**: Inbound/Outbound trunk (between Voice Daughtercard #2 and PSTN)

Trunk **(V)**: Inbound/Outbound trunk (between Voice Daughtercard #3 and PSTN)

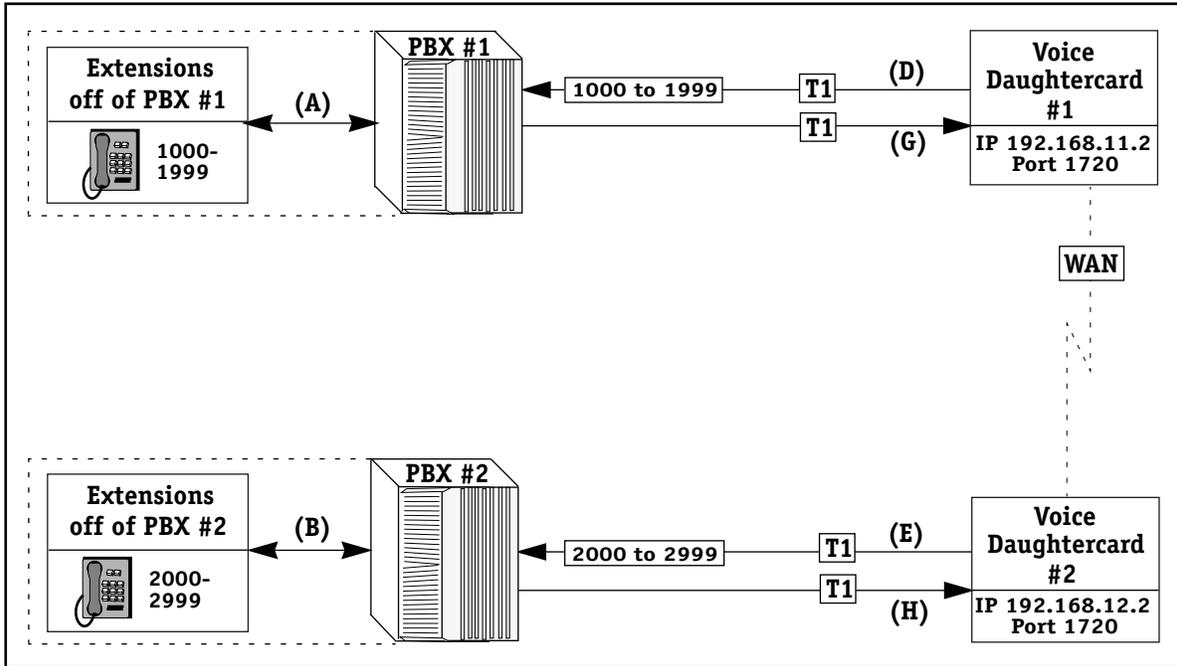
Trunk **(W)**: Inbound/Outbound trunk (between Voice Daughtercard #4 and PSTN)

Ethernet WAN **(Z)**: Inbound Ethernet (WAN) to H.323 Device

VoIP Networks without PSTN — Example 1

Four Digit Extensions and Two Voice Daughtercards

This is one of the simplest dialing schemes to implement in a VoIP network. It uses two voice switching daughtercards to translate four digit extensions. Extensions are unique across the entire enterprise network, and the PBX handles all calls to the PSTN. Since incoming and outgoing trunks are separated, this dialing scheme guarantees that no inseize collisions will occur.



Example 1 — Four Digit Extensions and Two Voice Daughtercards

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on trunk (D) and then uses these digits to route calls to lines (A).	— Expects to receive four digits on trunk (E) and then uses these digits to route calls to lines (B).
— Routes calls starting with 1 to lines (A).	— Routes calls starting with 2 to lines (B).
— Routes calls starting with 2 to trunk (G), and then the VoIP network uses these digits to route calls to trunk (E).	— Routes calls starting with 1 to trunk (H), and then the VoIP network uses these digits to route calls to trunk (D).

Remarks

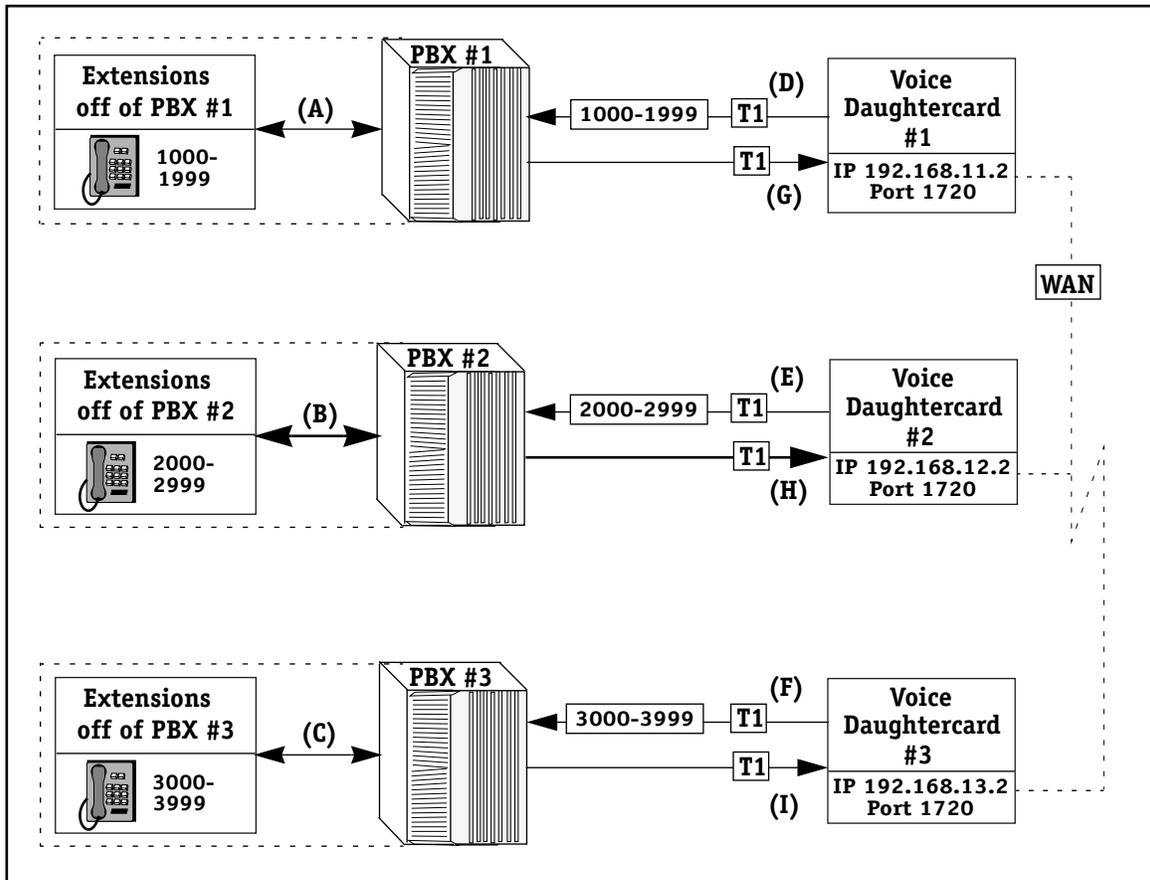
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks without PSTN — Example 2

Trunk Groups and Three Voice Daughtercards

This dialing scheme is used to set up additional sites on an existing VoIP network. Individual calls are routed to more than one possible destination, e.g., from Trunk G to Trunk E or Trunk F, depending on the number dialed. All telephone numbers in this example are unique across the VoIP network.



Example 2 — Trunk Groups and Three Voice Daughtercards

LEGEND for Diagram Components		
PBX #1 Configuration	PBX #2 Configuration	PBX #3 Configuration
— Expects to receive four digits on trunk (D) and then uses these digits to route calls to lines (A).	— Expects to receive four digits on trunk (E) and then uses these digits to route calls to lines (B).	— Expects to receive four digits on trunk (F) and then uses these digits to route calls to lines (C).
— Routes calls starting with 1 to lines (A).	— Routes calls starting with 2 to lines (B).	— Routes calls starting with 3 to lines (C).
— Routes calls starting with 2 or 3 to trunk (G), and then the VoIP network uses these digits to route calls to trunks (E) or (F), respectively.	— Routes calls starting with 1 or 3 to trunk (H), and then the VoIP network uses these digits to route calls to trunks (D) or (F), respectively.	— Routes calls starting with 1 or 2 to trunk (I), and then the VoIP network uses these digits to route calls to trunks (D) or (E), respectively.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

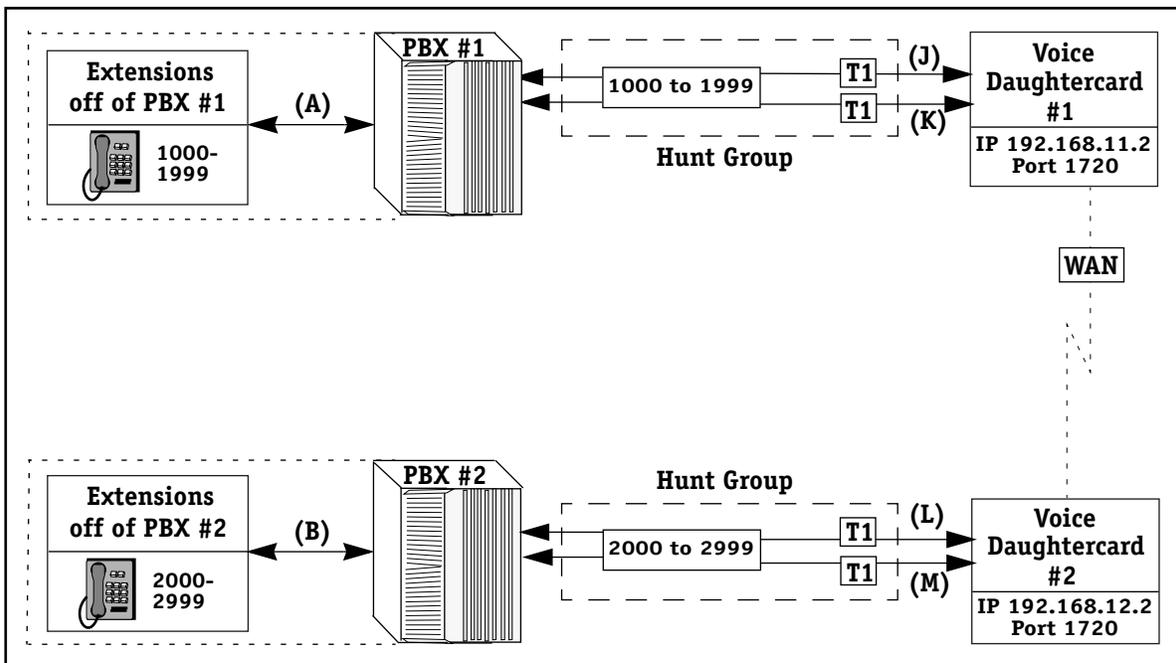
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks without PSTN — Example 3

One Hunt Group (48 Channels Across Two T1s)

This dialing scheme uses one hunt group spanning two T1 lines to make a single 48 channel trunk.

Hunt groups are called voice numbering plans in the AVNDS. Voice numbering plans relate phone groups and destinations. In this example, the voice daughtercards are using the top down hunt method to determine where new calls will be routed.



Example 3 — One Hunt Group (48 Channels Across Two T1s)

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on trunks (J) or (K) and then uses these digits to route calls to lines (A).	— Expects to receive four digits on trunks (L) or (M) then uses these digits to route calls to lines (B).
— Routes calls starting with 1 to lines (A).	— Routes calls starting with 2 to lines (B).
— Routes calls starting with 2 to trunks (J) or (K), and then the VoIP network uses these digits to route calls to trunks (L) or (M) according to the hunt method used	— Routes calls starting with 1 to trunks (L) or (M), and then the VoIP network uses these digits to route calls to trunks (J) or (K) according to the hunt method used.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

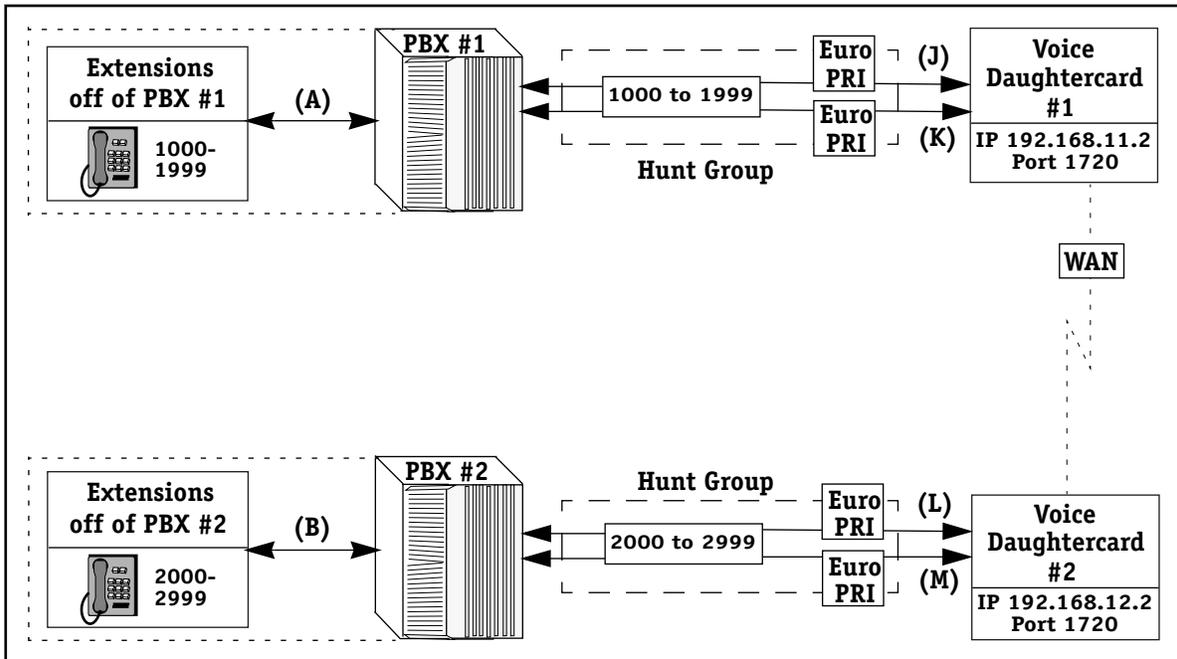
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks without PSTN — Example 4

One Hunt Group (60 Channels Across Two E1s)

This dialing scheme uses one hunt group spanning two E1 (Euro PRI) trunks to make a single 60 channel trunk.

Hunt groups are called voice numbering plans in the AVNDS. Voice numbering plans relate phone groups and destinations. In this example, the voice daughtercards are using the top down hunt method to determine where new calls will be routed.



Example 4 — One Hunt Group (60 Channels Across Two E1s)

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on trunks (J) or (K) and then uses these digits to route calls to lines (A).	— Expects to receive four digits on trunks (L) or (M) then uses these digits to route calls to lines (B).
— Routes calls starting with 1 to lines (A).	— Routes calls starting with 2 to lines (B).
— Routes calls starting with 2 to trunks (J) or (K), and then the VoIP network uses these digits to route calls to trunks (L) or (M) according to the hunt method used	— Routes calls starting with 1 to trunks (L) or (M), and then the VoIP network uses these digits to route calls to trunks (J) or (K) according to the hunt method used.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

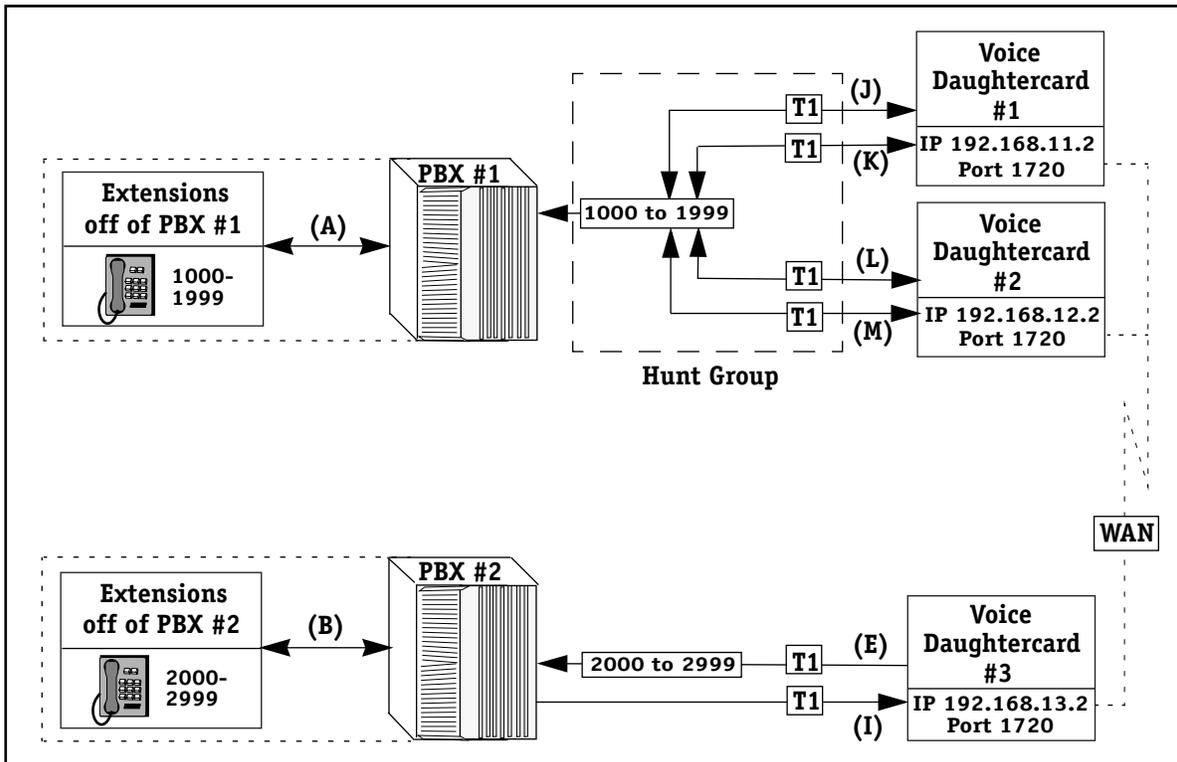
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — one hunt group (60 channels across two E1s) (H)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — E1 ISDN PRI (Euro PRI) (Y)	<i>voice port interface type</i> (page 5-34) <i>voice signaling companding</i> (page 5-77)

VoIP Networks without PSTN — Example 5

One Hunt Group (96 Channels Across Four T1s)

In this dialing scheme, one hunt group spans four T1 lines using two voice switching daughtercards (spanning two T1 lines each). In this example, the cards are installed in a single VSX motherboard in the same Omni Switch/Router to provide four T1 lines connected to a PBX.

Hunt groups are called voice numbering plans in the AVNDS. Voice numbering plans relate phone groups and destinations. In this example, the voice daughtercards are using the top down hunt method to determine where new calls will be routed.



Example 5 — One Hunt Group (96 Channels Across Four T1s)

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on trunks (J), (K), (L), or (M), and then uses these digits to route calls to lines (A).	— Expects to receive four digits on (E) and then uses these digits to route calls to lines (B).
— Routes calls starting with 1 to lines (A).	— Routes calls starting with 2 to lines (B).
— Routes calls starting with 2 to trunks (J), (K), (L), or (M), and then the VoIP network uses these digits to route calls to trunk (E).	— Routes calls starting with 1 to trunk (I), and then the VoIP network uses these digits to route calls to trunks (J), (K), (L), or (M) according to the hunt method used.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

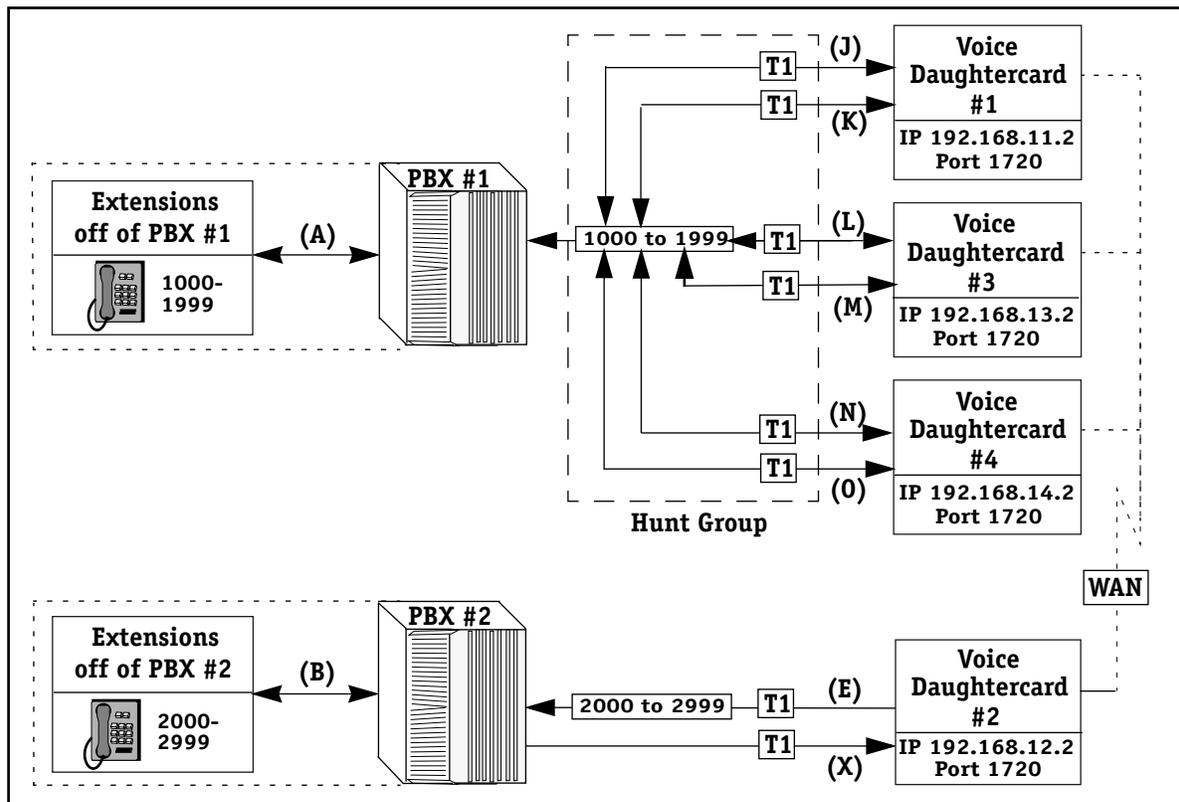
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Local channel — one hunt group (48 channels across two T1s) (G)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — Euro BRI (Z)	<i>voice port interface type</i> (page 5-34)

VoIP Networks without PSTN — Example 6

One Hunt Group (144 Channels Across Six T1s)

This dialing scheme has six T1 lines connected to one PBX. In this dialing scheme, one hunt group spans six T1 lines using three voice switching daughtercards (spanning two T1 lines each). In this example, the cards are installed in a two VSX motherboards in the same Omni Switch/Router to provide six T1 lines connected to a PBX.

Hunt groups are called voice numbering plans in the AVNDS. Voice numbering plans relate phone groups and destinations. In this example, the voice daughtercards are using the round robin hunt method to determine where new calls will be routed.



Example 6 — One Hunt Group (144 Channels Across Six T1s)

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on trunks (J), (K), (L), (M), (N) or (O), and then uses these digits to route calls to lines (A).	— Expects to receive four digits on trunk (E) and then uses these digits to route calls to lines (B).
— Routes calls starting with 1 to lines (A).	— Routes calls starting with 2 to lines (B).
— Routes calls starting with 2 to trunks (J), (K), (L), (M), (N) or (O), and then the VoIP network uses these digits to route calls to trunk (E).	— Routes calls starting with 1 to trunks (X), and then the VoIP network uses these digits to route calls to trunks (J), (K), (L), (M), (N) or (O) according to the hunt method used.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — one hunt group (48 channels across two T1s) (G)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

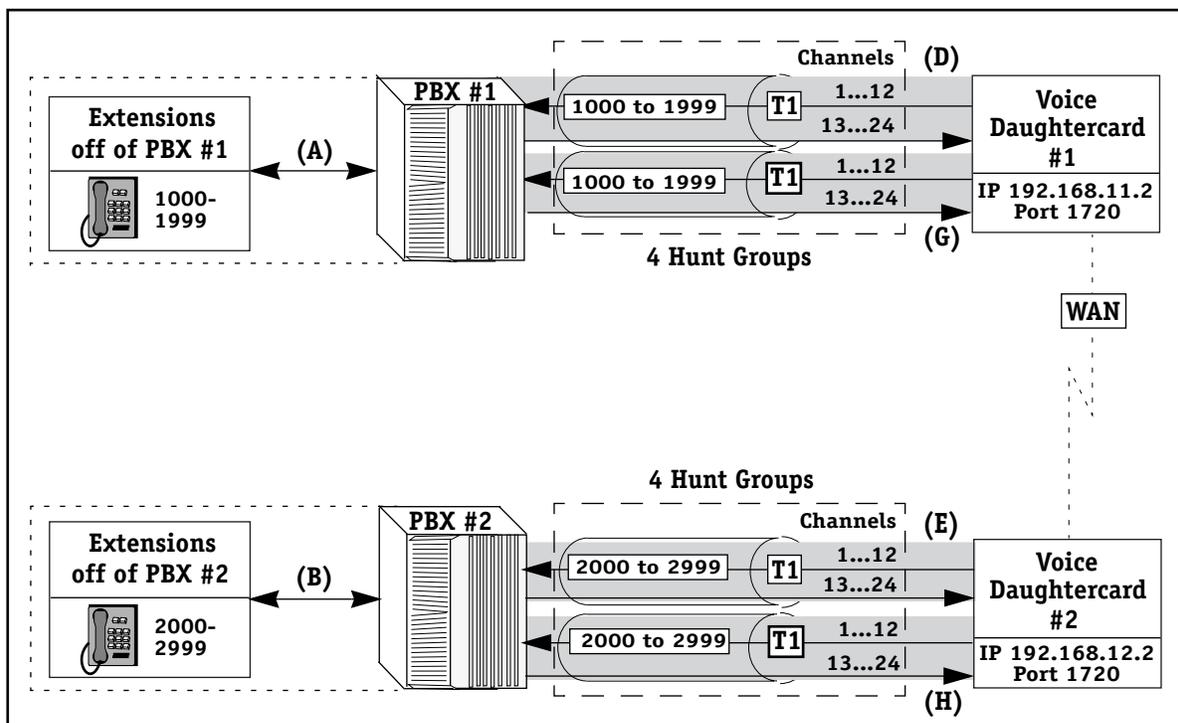
VoIP Networks without PSTN — Example 7

Four Hunt Groups (12 Channels Per Hunt Group)

This dialing scheme is used to split a T1 line in half, and demonstrates one way of having redundant T1 lines on one switch. In this example, one hunt group is half of a T1 line, or 12 channels. Each T1 trunk is split into an incoming and outgoing hunt group.

Dialing Scheme Examples 7 and 8 (showing fractional T1 hunt groups) apply only to digital (VSD) and Euro BRI (VSB) voice switching daughtercards. See Chapter 2, “VoIP Daughtercards” for a description of the various daughtercards, and Chapter 4, “Setup and Installation” for details on installation.

Hunt groups are called voice numbering plans in the AVNDS. Voice numbering plans relate phone groups and destinations. In this example, the voice daughtercards are using the round robin hunt method to determine where new calls will be routed.



Example 7— Four Hunt Groups (12 Channels Per Hunt Group)

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on channels 1 through 12 on trunks (D) or (G), and then uses these digits to route calls to lines (A).	— Expects to receive four digits on channels 1 through 12 on trunks (R) or (H), and then uses these digits to route calls to lines (B).
— Routes calls starting with 1 to lines (A).	— Routes calls starting with 2 to lines (B).
— Routes calls starting with 2 to channels 13 through 24 on trunks (D) or (G), and then the VoIP network uses these digits to route calls to trunks (E) or (H).	— Routes calls starting with 1 to channels 13 through 24 on trunks (E) or (H), and then the VoIP network uses these digits to route calls to trunks (D) or (G).

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

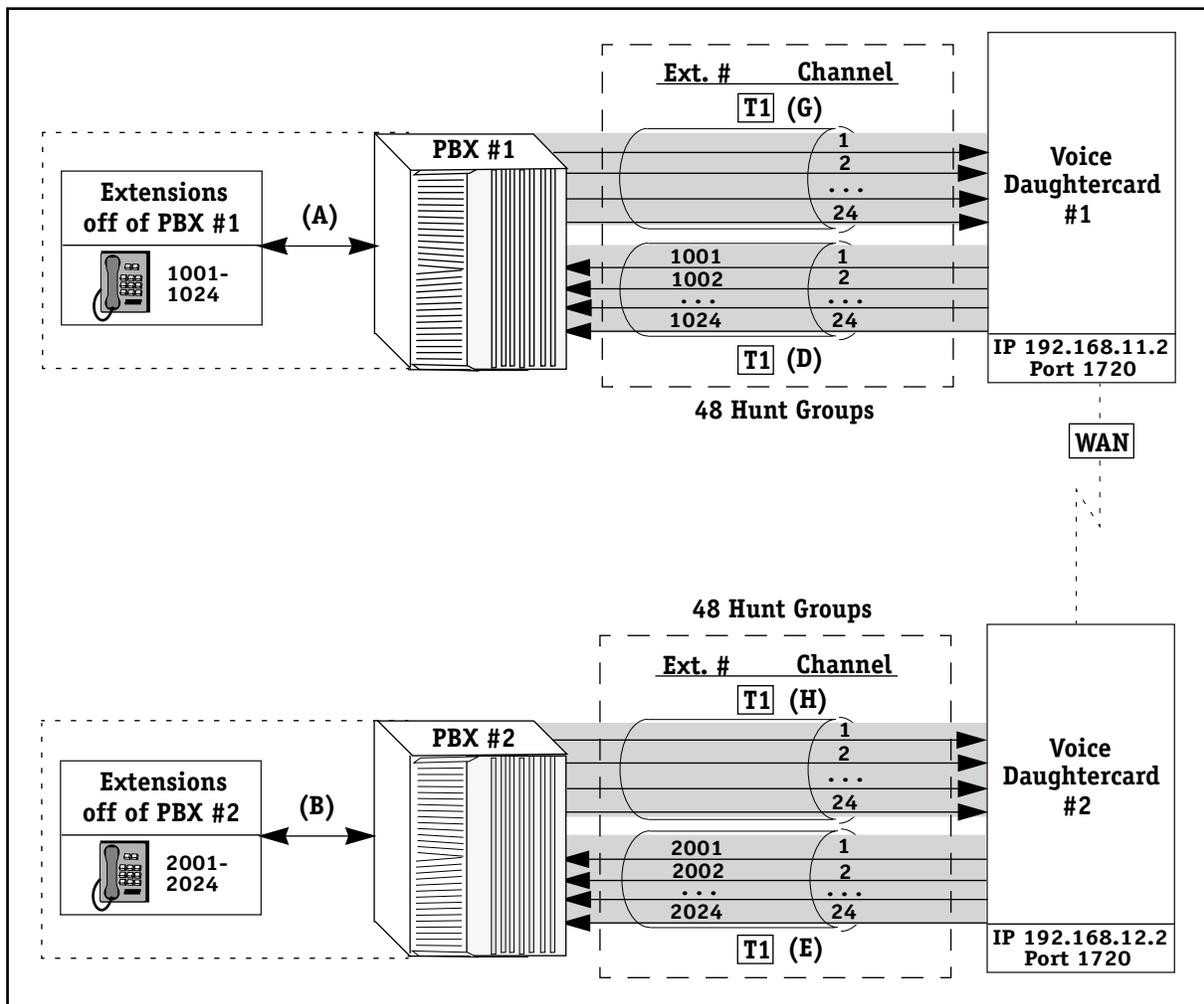
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — four hunt groups (12 channels per group/T1) (E)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks without PSTN — Example 8

48 Individual Hunt Groups (One Channel Per Group)

This dialing scheme shows how to divide a single T1 line into smaller (or fractional T1) trunk groups. Additionally, each channel has a unique telephone number and is also associated with a single telephone number and a single channel. Often this dialing scheme is used to test individual channels on a T1 line, but it can also be used to bypass hunt group behavior. Since each hunt group has only one channel, hunting is, in effect, disabled.

Dialing Scheme Examples 7 and 8 (showing fractional T1 hunt groups) apply only to digital (VSD) and Euro BRI (VSB) voice switching daughtercards. See Chapter 2, “VoIP Daughtercards” for a description of the various daughtercards, and Chapter 4, “Setup and Installation” for details on installation. Hunt groups relate phone groups and destinations. In the command line syntax, hunt groups are called voice numbering plans.



Example 8 — 48 Individual Hunt Groups (One Channel Per Group)

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on channels 1 through 24 on trunk (D) , and then uses these digits to route calls to lines (A) .	— Expects to receive four digits on channels 1 through 24 on trunk (E) , and then uses these digits to route calls to lines (B) .
— Routes calls starting with 1 to lines (A) .	— Routes calls starting with 2 to lines (B) .
— Routes calls starting with 2 to channels 1 through 24 on trunk (G) , and then the VoIP network uses these digits to route calls to trunk (E) .	— Routes calls starting with 1 to channels 1 through 24 on trunk (H) , and then the VoIP network uses these digits to route calls to trunk (D) .

Remarks

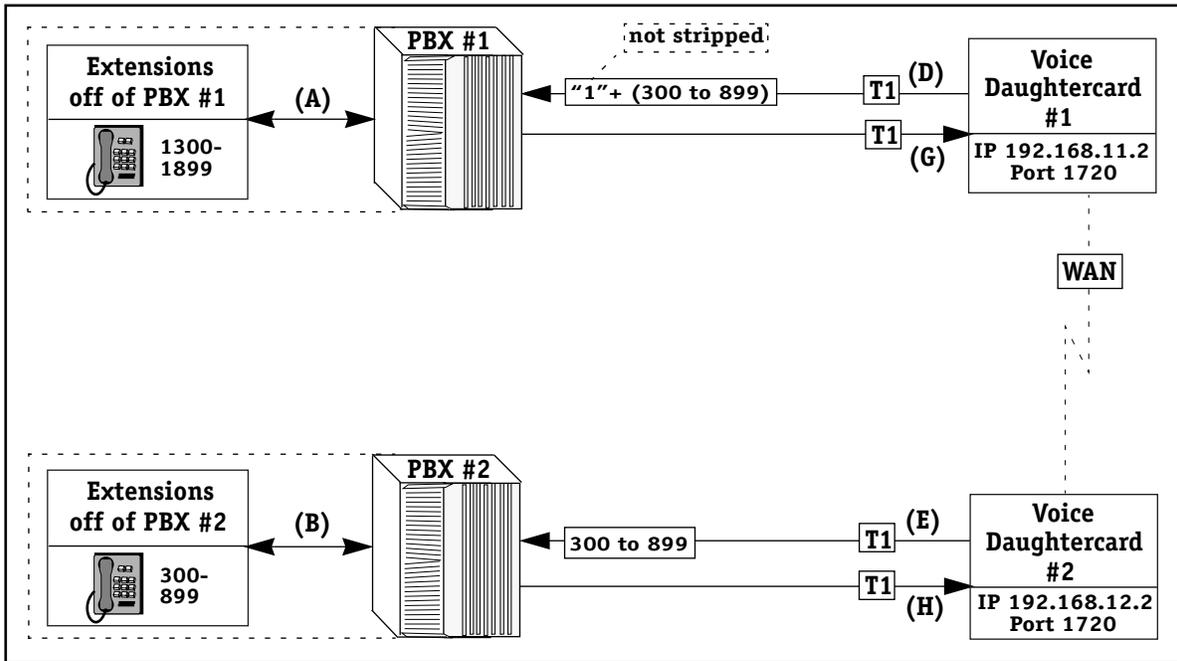
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — individual hunt groups (48 channels per group/T1) (D)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks without PSTN — Example 9

Trunk Groups and Mixed Length Extensions

This is another dialing scheme that is relatively simple to implement in a VoIP network, and demonstrates how to mix different extension lengths in one dialing scheme. It uses two voice switching daughtercards to translate a single digit trunk prefix and three digit extensions. The single digit site prefix, rather than the three digits extensions, are unique across the VoIP network. The site prefix digit is used to send the VoIP calls to the correct PBX node. When a caller dials a site prefix, e.g., 1, it routes the call to the corresponding PBX and then dials a prefix to get a specific trunk.



Example 9 — Trunk Groups and Mixed Length Extensions

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on trunk (D), and then uses these digits to calls to line (A).	— Expects to receive three digits on trunk (E), and then uses these digits to route calls to line (B).
— Routes calls starting with 1 to lines (A).	— Routes calls starting with 3 to lines (B).
— Routes calls starting with 3 to trunk (G), and then the VoIP network uses these digits to route calls to trunk (E).	— Routes calls starting with 1 to trunk (H), and then the VoIP network uses these digits to route calls to trunk (D).

Remarks

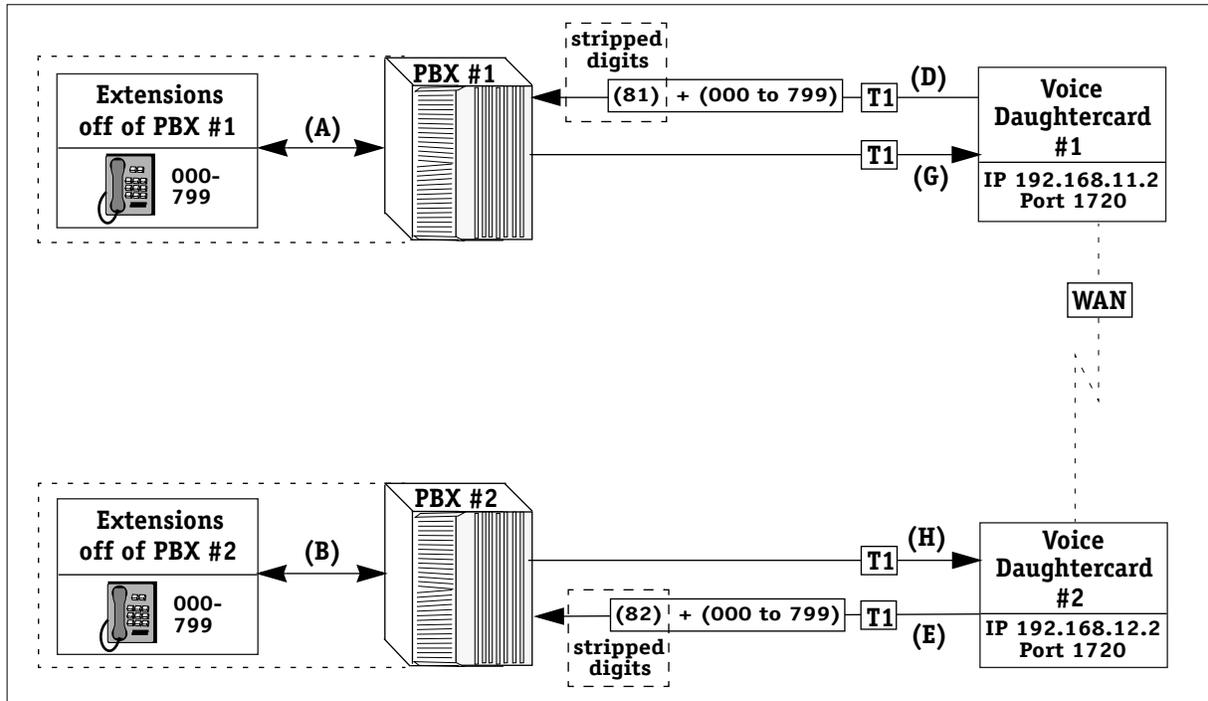
In the CLI commands, trunk groups are referred to as Site Prefix. Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — three digit local extensions (K)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks without PSTN — Example 10

Strip Digit Length (2)

In this dialing scheme, the PBX uses the first two digits received to route calls. The PBX first dials an “8” to go to the VoIP network. The 2nd digit dialed (“1” or “2”) determines the site (PBX) to which the call is sent. A “1” means the call goes to PBX # 1, and a “2” means the call goes to PBX #2. The two-digit prefix is stripped before the destination voice switching daughtercard sends the digits to the PBX.



Example 10 — Strip Digit Length (2)

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive three digits on trunk (D), and then uses these digits to route calls to lines (A).	— Expects to receive three digits on trunk (E), and then uses these digits to route calls to lines (B).
— Routes calls starting with 0-7 to lines (A).	— Routes calls starting with 0-7 to lines (B).
— Routes calls starting with 82 to trunk (G), and then the VoIP network uses these digits to route calls to trunk (E).	— Routes calls starting with 81 to trunk (H), and then the VoIP network uses these digits to route calls to trunk (D).

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — three digit local extensions (K)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — 2 (T)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

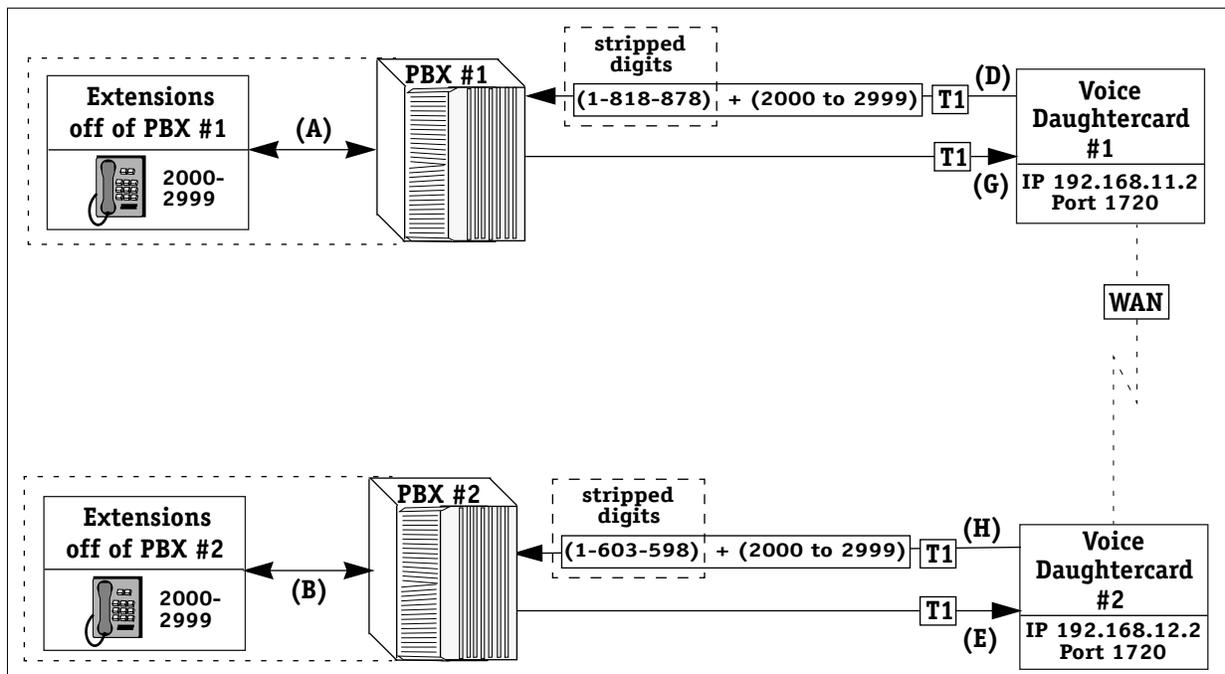
VoIP Networks without PSTN — Example 11

Trunk Groups and Eleven Digit Extensions

This dialing scheme can be used with two voice daughtercards to translate seven digit trunk prefixes and four digit extensions. The seven digit site prefix, rather than the four digits extensions, are unique across the VoIP network. This enables each PBX site to handle the same or overlapping phone extensions.

The site prefix digits are used to send the VoIP calls to the correct PBX node. When a caller dials a specific site prefix, e.g., 1-603-598, it routes the call to the corresponding PBX and then dials a prefix to get a specific trunk. The user would dial 1-603-598-2xxx to call an extension off of PBX #2, for example. The VoIP daughtercard will strip the first seven digits and forward the last four digits to the PBX.

Any number used as a site prefix cannot be used for the first digit of any valid extension.



Example 11— Trunk Groups and Eleven Digit Extensions

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on trunk (D) , and then uses these digits to route calls to lines (A) .	— Expects to receive four digits on trunk (H) , and then uses these digits to route calls to lines (B) or trunk (E) .
— Routes calls starting with 2 to lines (A) .	— Routes calls starting with 2 to lines (B) .
— Routes calls starting with 1-603-598 to trunk (G) , and then the VoIP network uses these digits to route calls to trunk (H) .	— Routes calls starting with 1-818-878 to trunk (E) , and then the VoIP network uses these digits to route calls to trunk (D) .

Remarks

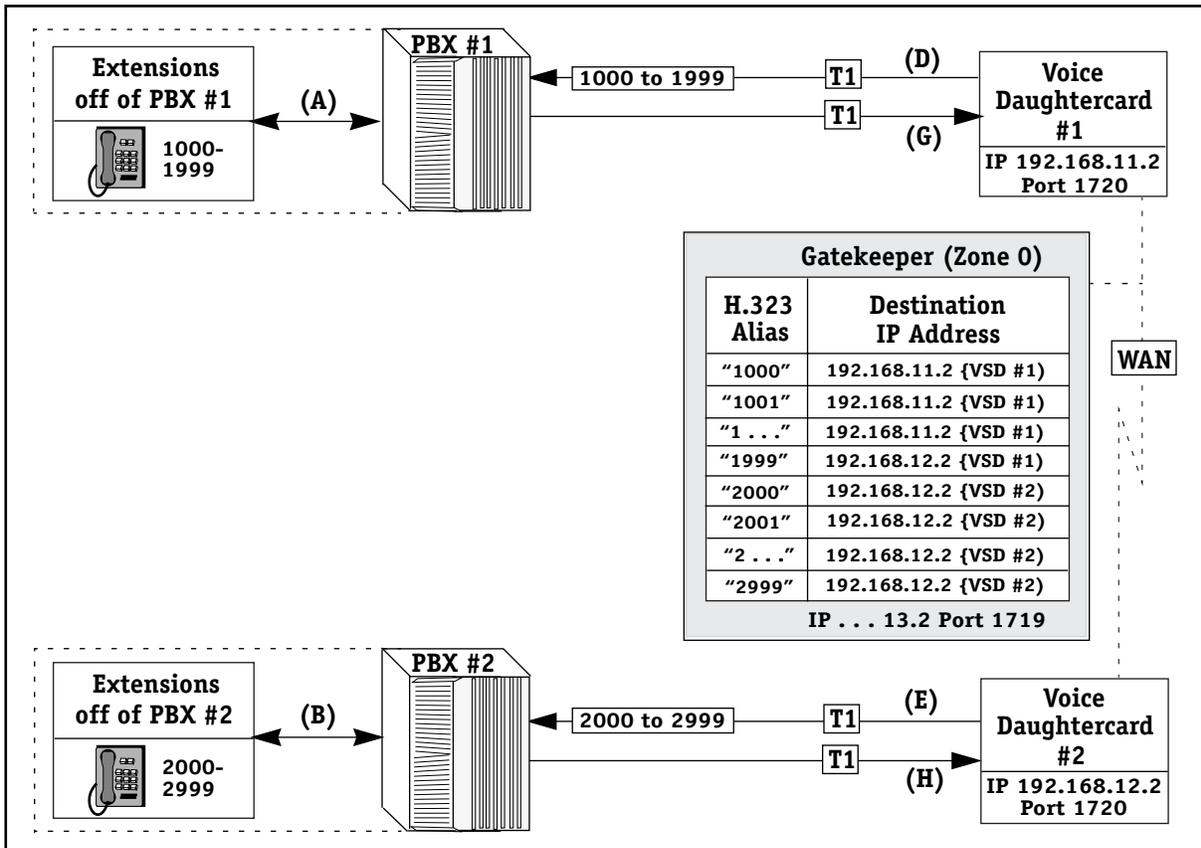
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — eleven digit local extensions (M)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — 7 (V)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks without PSTN — Example 12

H.323 Gatekeeper

This dialing scheme is used to connect voice switching daughtercards to an external H.323 NT100 RADVision (or other third-party) gatekeeper software application installed on a server or workstation. The phone groups for the individual daughtercard must be associated with the card in order for it to generate the H.323 alias telephone numbers for the gatekeepers. Configure the AVNDS for the voice daughtercards so that the H.323 destination is the gatekeeper instead of Trunks D or E. Most third-party gatekeepers should be compatible with the Alcatel's H.323 gateway providing the gatekeepers are also H.323 compliant.



Example 12 — H.323 Gatekeeper

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on trunk (D) and then uses these digits to route calls to lines (A).	— Expects to receive four digits on trunk (E) and then uses these digits to route calls to lines (B).
— Routes calls starting with 1 to lines (A).	— Routes calls starting with 2 to lines (B).
— Routes calls starting with 2 to trunk (G). The VoIP network then routes the call to the gatekeeper, and the gatekeeper uses these digits to route calls to trunk (E).	— Routes calls starting with 1 to trunk (H). The VoIP network then routes the call to the gatekeeper, and the gatekeeper uses these digits to route calls to trunk (D).

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
H.323 gateway to H.323 gatekeeper (RADVision) (B)	<i>voice network card b.323 gatekeeper control</i> (page 5-217) <i>voice network b.323 gatekeeper mode</i> (page 5-218) <i>voice network b.323 gatekeeper address</i> (page 5-219) <i>voice network b.323 gatekeeper associate</i> (page 5-223)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks with PSTN — Example 13

North American PSTN and VoIP Calls

This dialing scheme is used for calls going through the North American PSTN and the VoIP Network. The following four diagrams are used to demonstrate how these calls are handled:

- North American PSTN Calls — Overview
- North American PSTN Calls — Outbound
- North American PSTN Calls — Inbound
- North American PSTN Calls — VoIP Network

Note that DID (Direct Inward Dial) is used only on inbound calls, and that all calls are connected using the North American Numbering Plan (NANP) that also includes Canada. See Dialing Scheme Example 15 for details on North American and International calls going through the PSTN or the VoIP network.

To use this dialing scheme, the voice phone group type must be set to NANP extensions. The site prefix digits and strip digits must be set to seven digits. NANP PSTN numbers depend on the configuration of the local exchange, of which there are three types: 1) those that support 7 digit dialing to get calls across the street (nxx-xxxx), and 2) those that support 10 digit dialing to get calls across the street (npa-nxx-xxxx) where NPA is the area code, and 3) those that support 11 digit dialing to get calls across the street (1-npa-nxx-xxxx) where npa is the area code.

When using voice phone group type NANP PSTN, the PSTN functionality is supported with the following exceptions:

- DID extensions (add numbers) and the NANP PSTN cannot be the same numbers.
- DID extensions (add numbers) and the site prefix digits cannot be the same numbers.
- Site prefix digits and NANP PSTN numbers cannot be the same numbers.

The phone group type NANP PSTN may be substituted by using site prefixes (forces callers to dial one or more digits, e.g., 9, to place an external call. For more information on using site prefixes, see also *VoIP Networks without PSTN — Example 10* on page 3-25.

◆ Cautions ◆

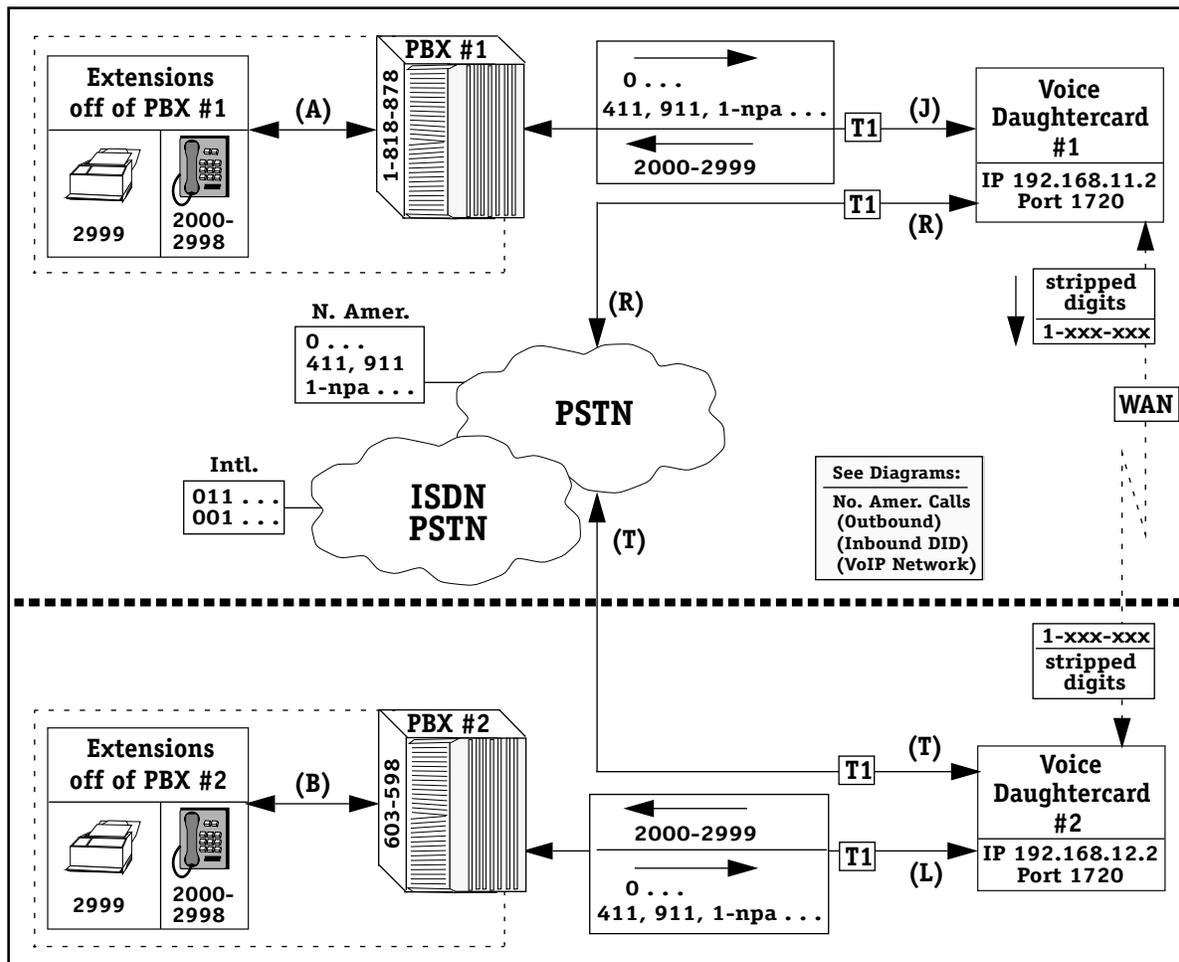
Voice daughtercards using either voice phone group type of NANP PSTN or International PSTN dialing schemes cannot handle PBXs with extensions starting with 0, 1, 411 or 911. The following extensions are also not allowed: 0000 to 0999, 1000 to 1999, 4110 to 4119, and 9000 to 9999. (For more details, see Chapter 5, “VoIP Commands.”)

Companies using Alcatel’s VoIP feature are responsible for programming and testing all dialing schemes to reduce the likelihood or to eliminate the possibility of toll fraud from the PSTN.

North American PSTN Calls — Overview

In this dialing scheme two voice switching daughtercards are used to translate area codes and telephone numbers. All eleven digits in the telephone numbers are unique across the VoIP network, and the voice switching daughtercards are responsible for all telephone number routing. 411 and 911 calls can be handled as well using this dialing scheme. Calls that are not routed across the VoIP network will be dropped and inserted, referred to as “Drop and Insert,” into the PSTN. Minimal to no PBX re-configuration is required; however, due to less than 99.995% reliability of Voice over IP networks, this dialing scheme is not recommended unless “passthrough” is used on some channels. (Passthrough *not available this release*).

To call a 2000 extension off of PBX#1, the caller dials an eleven digit NANP telephone number. In the overview diagram below, the voice daughtercard strips off the first seven digits, and then forwards the last four digits of the dialed number.



Example 13 — North American PSTN Calls (Overview)

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits on trunk (J) and then uses these digits to route calls to lines (A).	— Expects to receive four digits on trunk (L) and then uses these digits to route calls to lines (B).
— Routes all calls starting with 2 to lines (A).	— Routes all calls starting with 2 to lines (B).

LEGEND for Diagram Components	
— Routes all 0 . . . , 411, 911 and 1-npa . . . calls to trunk (J) , and then the VoIP network uses these digits to route calls to trunk (L) , or trunk (R) if non-PSTN extensions.	— Routes all 0 . . . , 411, 911 and 1-npa . . . calls to trunk (L) , and then the VoIP network uses these digits to route calls to trunk (J) , or trunk (T) if non-PSTN extensions.

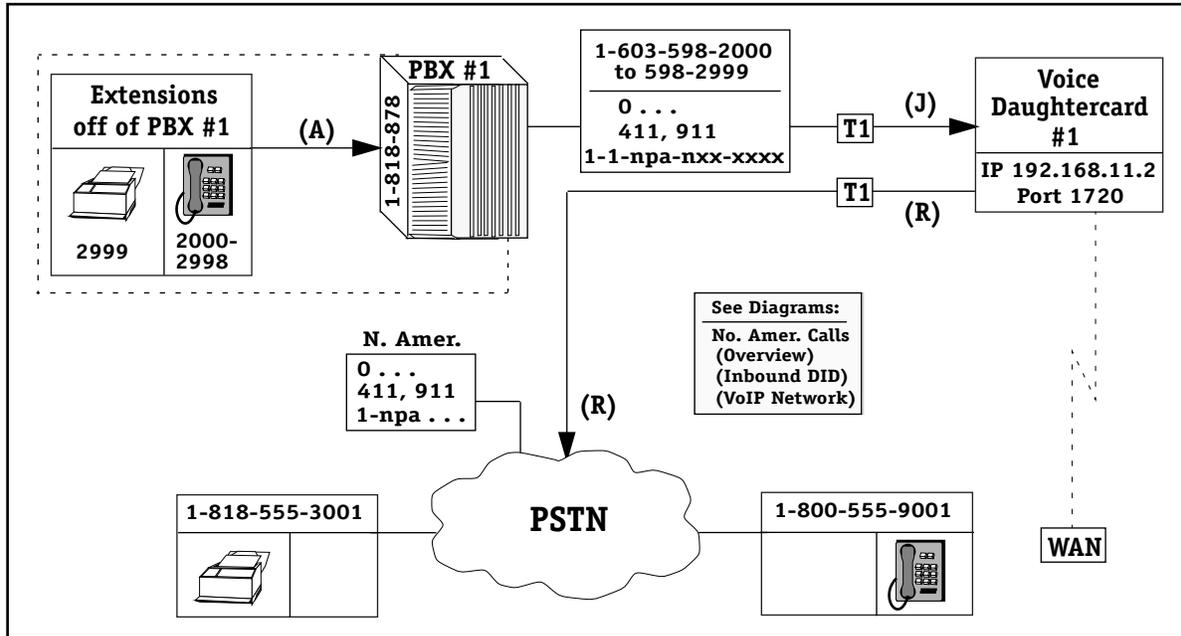
Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows, and are applicable to the outbound, inbound DID and North American VoIP calls.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — NANP extensions (N)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — PSTN NANP (P)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — 7 (V)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

North American PSTN Calls — Outbound

This diagram demonstrates how outbound North American PSTN calls are sent to the PSTN.

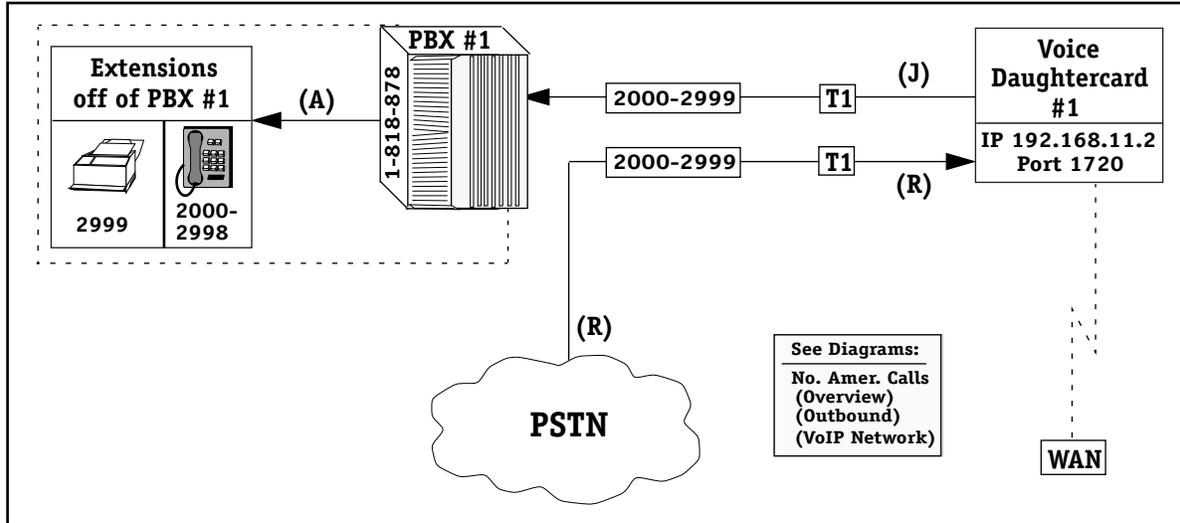


Example 13 — North American PSTN Calls (Outbound)

LEGEND for Diagram Components	
PBX #1 Configuration	Voice Daughtercard #1 Configuration
— Routes all calls to trunk (J) and sends four digits to voice daughtercard. See next diagram (No. Amer. PSTN Inbound) trunk (J) information.	— Expects to receive one to 24 digits on trunk (J), and then uses these digits to route calls to either trunk (R) or the VoIP network.

North American PSTN Calls — Inbound DID (Direct Inward Dial)

This diagram demonstrates how inbound North American DID calls from the PSTN are handled.



Example 13 — North American PSTN Calls (Inbound DID)

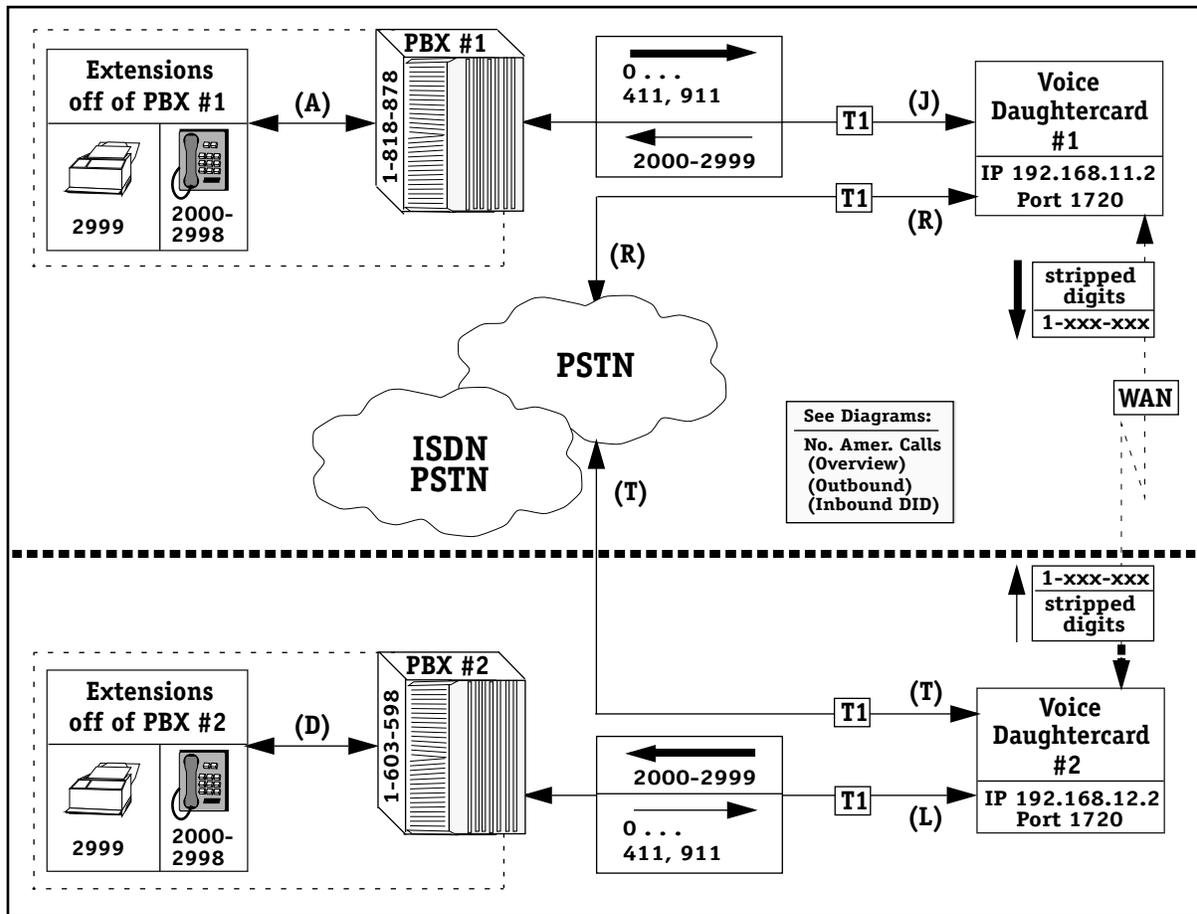
LEGEND for Diagram Components	
PBX #1 Configuration	Voice Daughtercard #1 Configuration
— Expects to receive four digits on trunk (J) and then uses these digits to route calls starting with 2 to lines (A) . See previous diagram (No. Amer. PSTN Outbound) trunk (J) information.	— Expects to receive four digits on trunk (R) , and then uses these digits to route calls to either trunk (J) or VoIP network.

North American PSTN Calls — VoIP Network

This diagram demonstrates how North American calls are handled in a VoIP network.

When a 1-818-878-2000 extension is called from a 1-603-589-2000 extension, the PBX routes calls to Trunk J and then sends 1-603-589-2000 number to Voice Daughtercard #1.

The Voice Daughtercard #1 strips the 1st seven digits and forwards the last four digits across the WAN to Voice Daughtercard #2. These four digits are then forwarded to Trunk L. PBX #2 receives the four digits and then routes the call to the appropriate extension.



Example 13 — North American PSTN Calls (VoIP Network)

LEGEND for Diagram Components	
Voice Daughtercard #1 Configuration	Voice Daughtercard #2 Configuration
— Expects to receive one to 24 digits on trunk (J) and then uses these digits to route calls to voice daughtercard #2.	— Expects to receive one to 24 digits on (L) and then uses these digits to route calls to voice daughtercard #1.
— Routes all 1-603-589 calls to trunk (L).	— Routes all 1-818-878 calls to trunk (J).
— Routes all 1-npa-nxx-xxxx PSTN calls to trunk (R).	— Routes all 1-npa-nxx-xxxx PSTN calls to trunk (T).
— Routes all PSTN calls to trunk (R).	— Routes all PSTN calls to trunk (T).

VoIP Networks with PSTN — Example 14

North American PSTN, International PSTN and VoIP Calls

This dialing scheme is used for calls going through the North American PSTN, International PSTN and the VoIP network. The following four diagrams are used to demonstrate how these calls are handled:

- North American PSTN, International PSTN and VoIP Calls — Overview (*Not available this release*)
- North American PSTN and International PSTN Calls — Outbound (*Not available this release*)
- North American PSTN and International PSTN Calls — Inbound (*Not available this release*)
- North American PSTN and International PSTN Calls — VoIP Network (*Not available this release*)

See Dialing Scheme Example 13 for details on calls going only through the North American PSTN and the VoIP network.

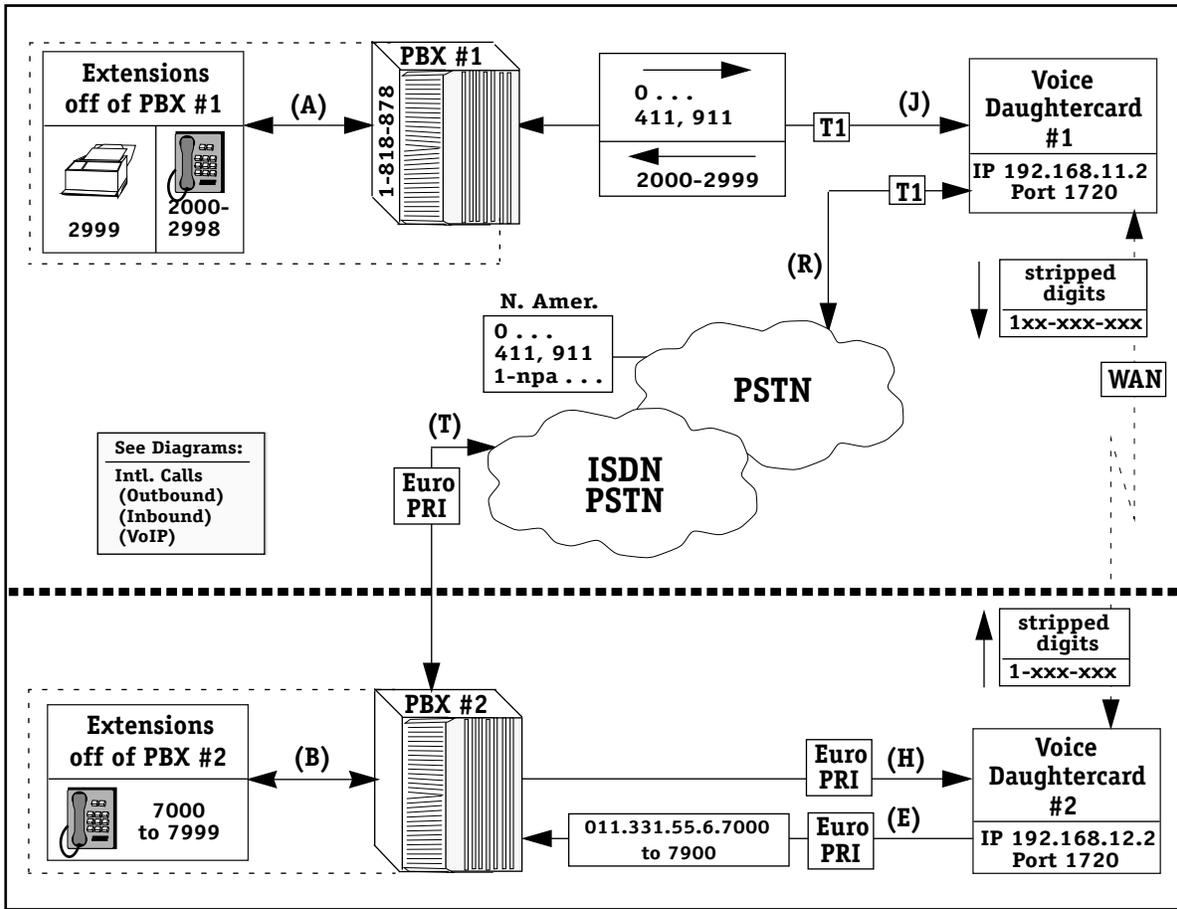
Voice daughtercards using either North American or International PSTN dialing schemes cannot handle PBXs with extensions starting with 0, 1, 411 or 911. The following extensions are also not allowed: 0000 to 0999, 1000 to 1999, 4110 to 4119, and 9110 to 9119.

◆ Cautions ◆

Companies using Alcatel's VoIP feature are responsible for programming and testing all dialing schemes to reduce the likelihood or to eliminate the possibility of toll fraud from the PSTN.

International PSTN Calls — Overview

This dialing scheme is used to handle international telephone calls. *Not available this release.*



Example 14 — International PSTN Calls (Overview)

LEGEND for Diagram Components (<i>not available this release</i>)	
PBX #1 Configuration	PBX #2 Configuration

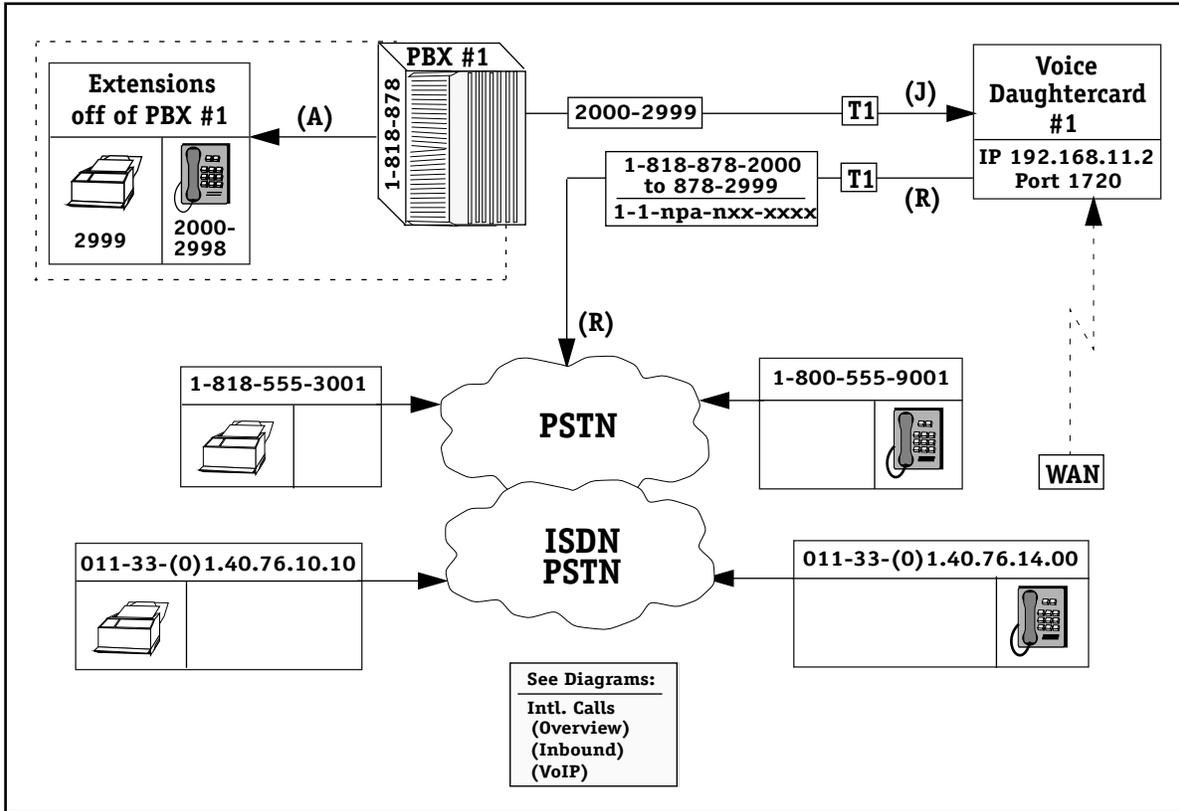
Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows, and are applicable to the outbound, inbound and VoIP PSTN International calls.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — NANP extensions (N)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — INTL extension (O)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — PSTN NANP (P)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — 7 (V)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)
Digital Interface type — E1 ISDN PRI (Euro PRI) (Y)	<i>voice port interface type</i> (page 5-34) <i>voice signaling companding</i> (page 5-77)

International PSTN Calls — Outbound

This dialing scheme is used to handle outbound international telephone calls. *Not available this release.*

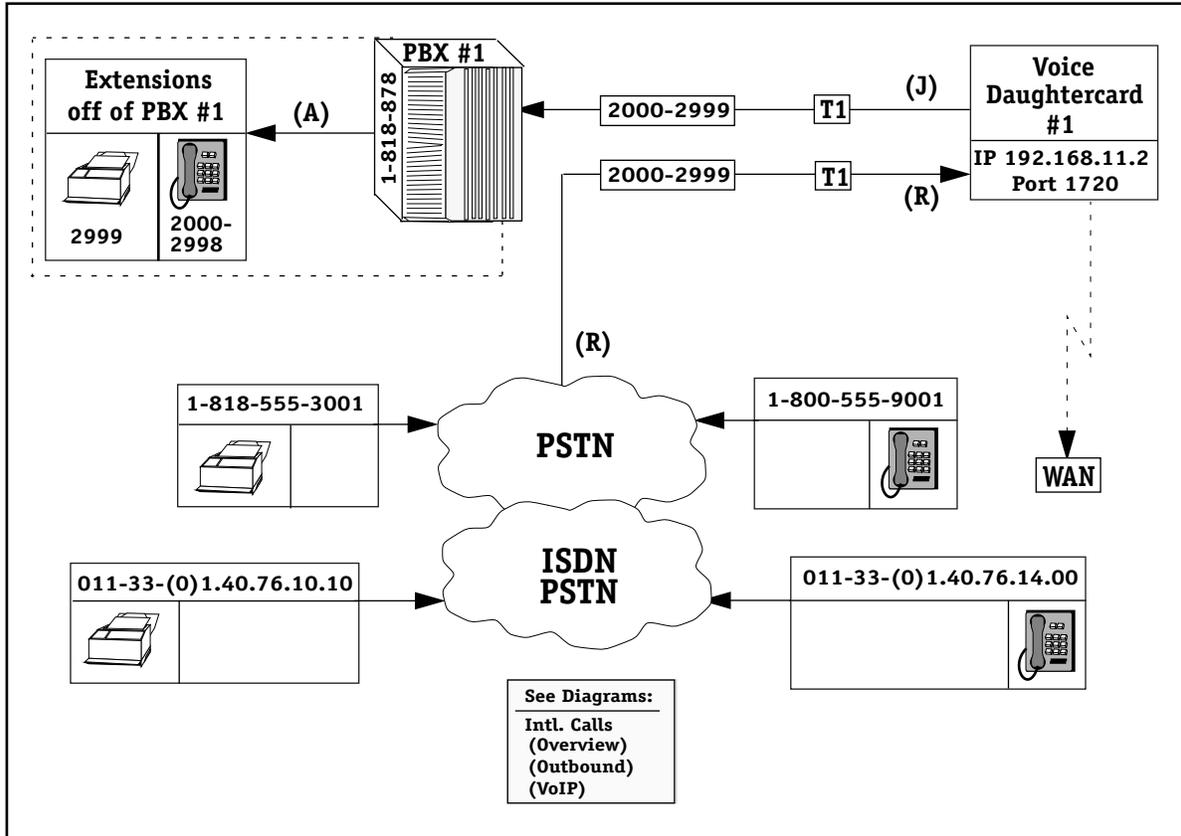


Example 14 — International PSTN Calls (Outbound)

LEGEND for Diagram Components (<i>not available this release</i>)	
PBX #1 Configuration	Voice Daughtercard #1 Configuration

International PSTN Calls — Inbound

This dialing scheme is used to handle inbound international telephone calls. *Not available this release.*

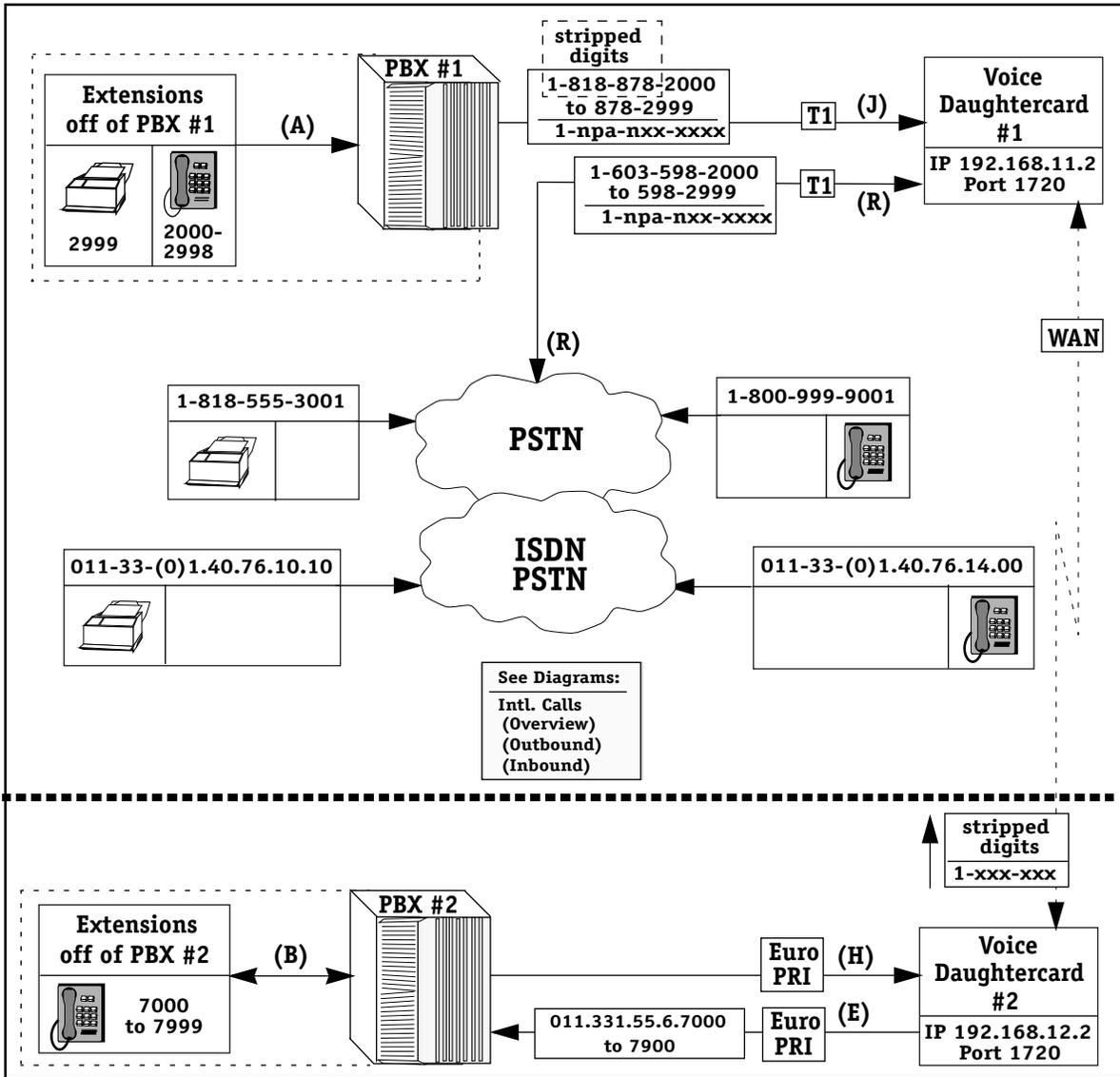


Example 14 —International PSTN Calls (Inbound)

LEGEND for Diagram Components (<i>not available this release</i>)	
PBX #1 Configuration	Voice Daughtercard #1 Configuration

International PSTN Calls — VoIP Network

This diagram demonstrates what happens on international calls handled between voice daughtercards. *Not available this release.*



Example 14 — PSTN International Calls (VoIP Network)

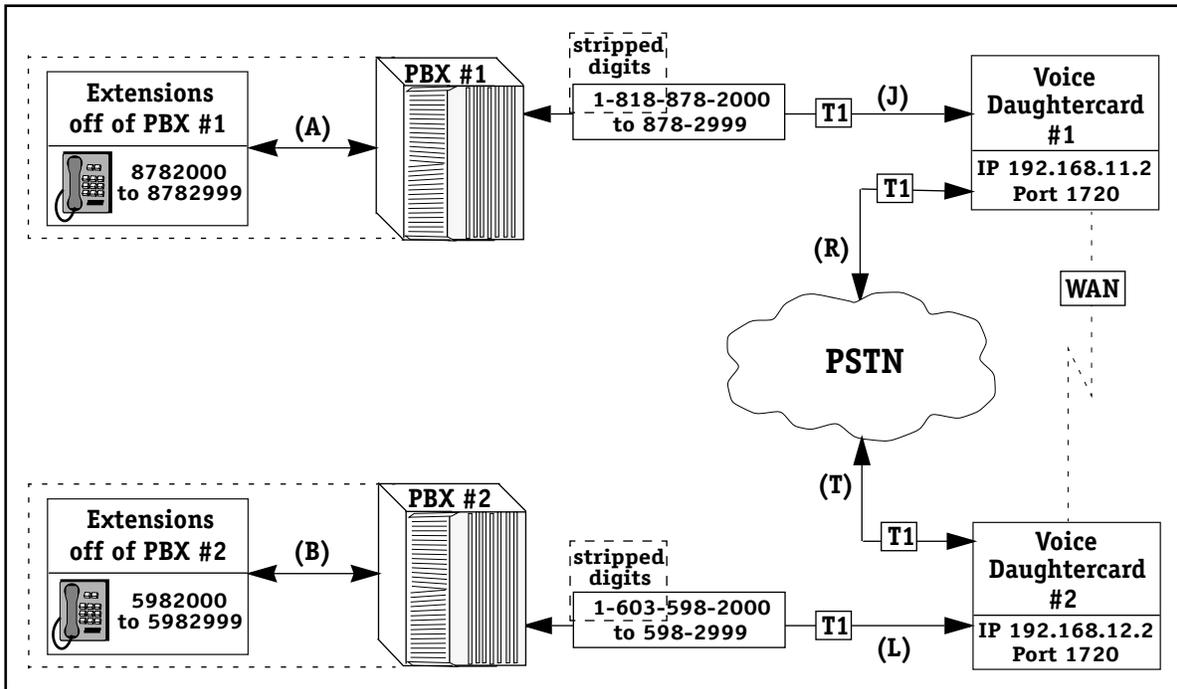
LEGEND for Diagram Components (<i>not available this release</i>)	
PBX #1 Configuration	PBX #2 Configuration

VoIP Networks with PSTN — Example 15

PSTN and Eleven Digit Extensions

This dialing scheme demonstrates using eleven digit extensions with the PSTN, although only seven digits are actually forwarded to the PBXs.

Any number used as a site prefix cannot be used for the first digit of any valid extension.



Example 15 — PSTN and Eleven Digit Extensions

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive seven on trunk (J) and then uses these digits to route calls to lines (A) .	— Expects to receive seven digits on trunk (L) and then uses these digits to route calls to lines (B) .
— Routes calls starting with 878-2xxx to lines (A) .	— Routes calls starting with 1-598-2xxx to lines (B) .
— Routes calls starting with 1-603-598-2xxx to trunk (J) , and then the VoIP network uses these digits to route calls to trunk (L) .	— Routes calls starting with 1-818-878-2xxx to trunk (L) , and then the VoIP network uses these digits to route calls to trunk (J) .

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

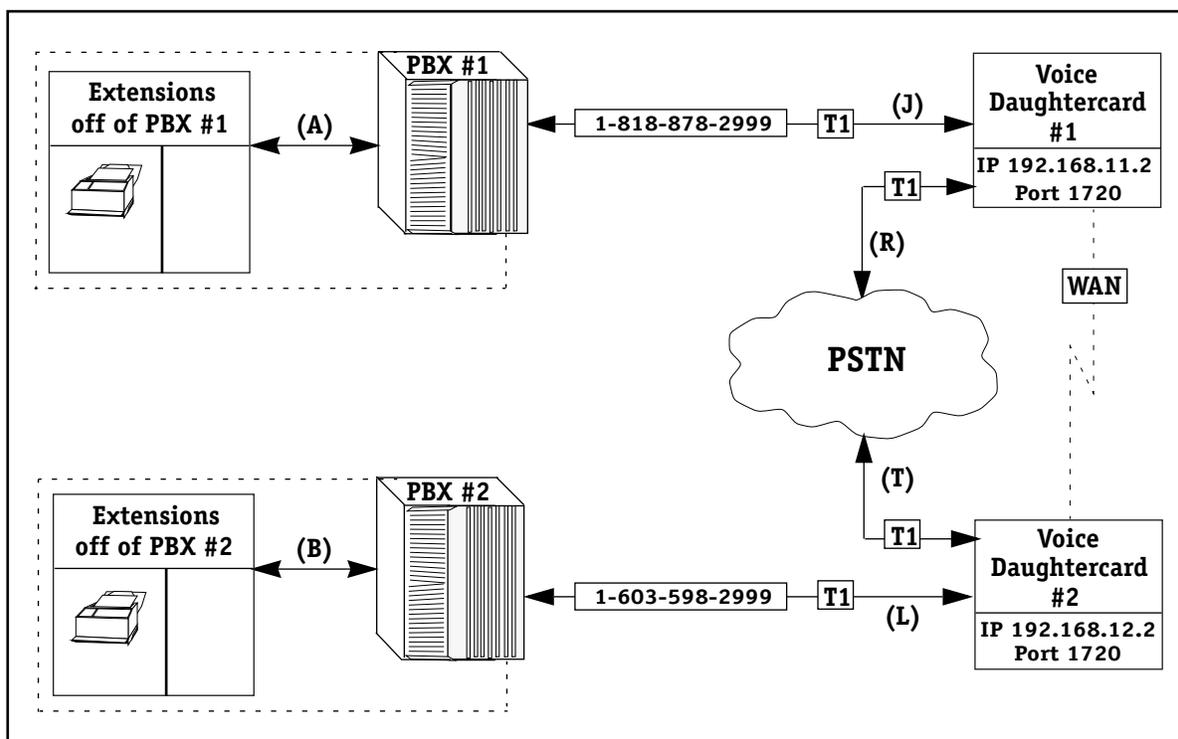
Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — NANP extensions (N)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — PSTN NANP (P)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — 4 (U)	<i>voice phone group strip digit length</i> (page 5-244)
Strip digit length — 7 (V)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks with PSTN — Example 16

FAX over IP Network

This dialing scheme is used to create a homogeneous Fax Over IP network, i.e. a network wherein no voice traffic per se is processed by the VoIP network. In the event of a digital voice daughtercard power failure, the Deadman relay switch on the card goes into PSTN fallback mode to allow new, incoming calls to be placed. For more details on the Deadman switch, see Chapter 2, “Voice Daughtercards.”

To use voice and fax on the VoIP network simultaneously, the fax/modem switchover feature must be turned ON via the **voice coding profile switchover** command; otherwise, voice calls will fail. See Chapter 5 “VoIP Commands,” for details.



Example 16 — Fax over IP Network

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive eleven digits on trunk (J), and then uses these digits to route calls to lines (A).	— Expects to receive eleven digits on trunk (L), and then uses these strip digits to route calls to lines (B).
— Routes calls starting with 1-603-598 to trunk (J), and then the VoIP network uses these digits to route calls to trunk (L). If the voice daughtercard loses power, calls will be routed using PSTN fallback to trunk (R).	— Routes calls starting with 1-818-878 to trunk (L), and then the VoIP network uses these digits to route calls to trunk (J). If the voice daughtercard loses power, calls will be routed using PSTN fallback to trunk (T).

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — NANP extensions (N)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — PSTN NANP (P)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — 7 (V)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

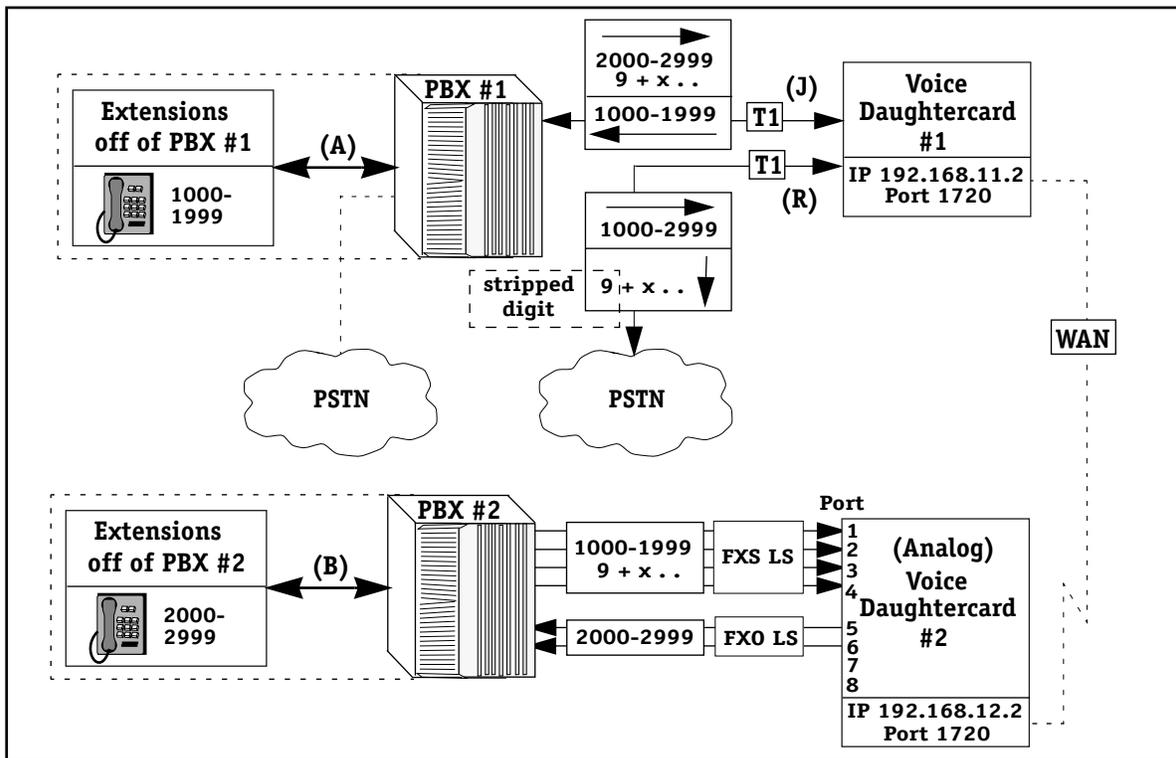
VoIP Networks with PSTN — Example 17

Mixed Digital and Analog Voice Daughtercards

This dialing scheme illustrates how to connect a digital voice switching daughtercard (VSD) to an analog voice switching daughtercard (VSA) to send and receive calls. It also depicts a remote PSTN connection as part of a VoIP network in which a VSA providing analog connections using Plain Old Telephone Service (POTS) is connected to the PSTN via a remote VSD.

PBX #2 can send four simultaneous PSTN calls (on FXS LS lines) to the VoIP network; however, it can only receive two simultaneous calls (on FXO LS lines) from the VoIP network.

On Voice Daughtercard #2, the Foreign Exchange Station (FXS) Loop Start lines are used to send calls to the VoIP network from the PBX via analog trunks, whereas the Foreign Exchange Office (FXO) Loop Start lines are used to receive calls from the VoIP network and send them to the PBX.



Example 17 — Mixed Digital and Analog Voice Daughtercards

LEGEND for Diagram Components	
Voice Daughtercard #1 Configuration	Voice Daughtercard #2 Configuration
— Expects to receive prefix digit 9 followed by at least one digit, and up to 22 digits or a "2" followed by any three digits on trunk (J).	— Expects to receive prefix digit 9 followed by at least one digit, and up to 22 digits or a "2" followed by any three digits on ports 1, 2, 3, or 4.
— Expects to receive a "1" or "2" followed by any three digits on trunk (R).	— Does not expect to receive any in-seizes (outbound calls from PBX #2) on ports 1, 2, 3 or 4.

LEGEND for Diagram Components	
— Routes calls starting with prefix digit 9 to PSTN (port 2 of Voice Daughtercard #1), after stripping first digit (“9”) to trunk (R) .	— Routes calls starting with prefix digit 9 to PSTN (port 2 of Voice Daughtercard #1) via the VoIP network (WAN).
— Routes calls starting with “1” followed by any three digits to trunk (J)	— Routes calls starting with “1” followed by any three digits to trunk (J) via the VoIP network (WAN).
— Routes calls starting with “2” followed by any three digits to ports 3 or 4 of Voice Daughtercard #2 via the WAN.	— Routes calls starting with “2” followed by any three digits to ports 5 or 6 of Voice Daughtercard #2 via the VoIP network.

Remarks

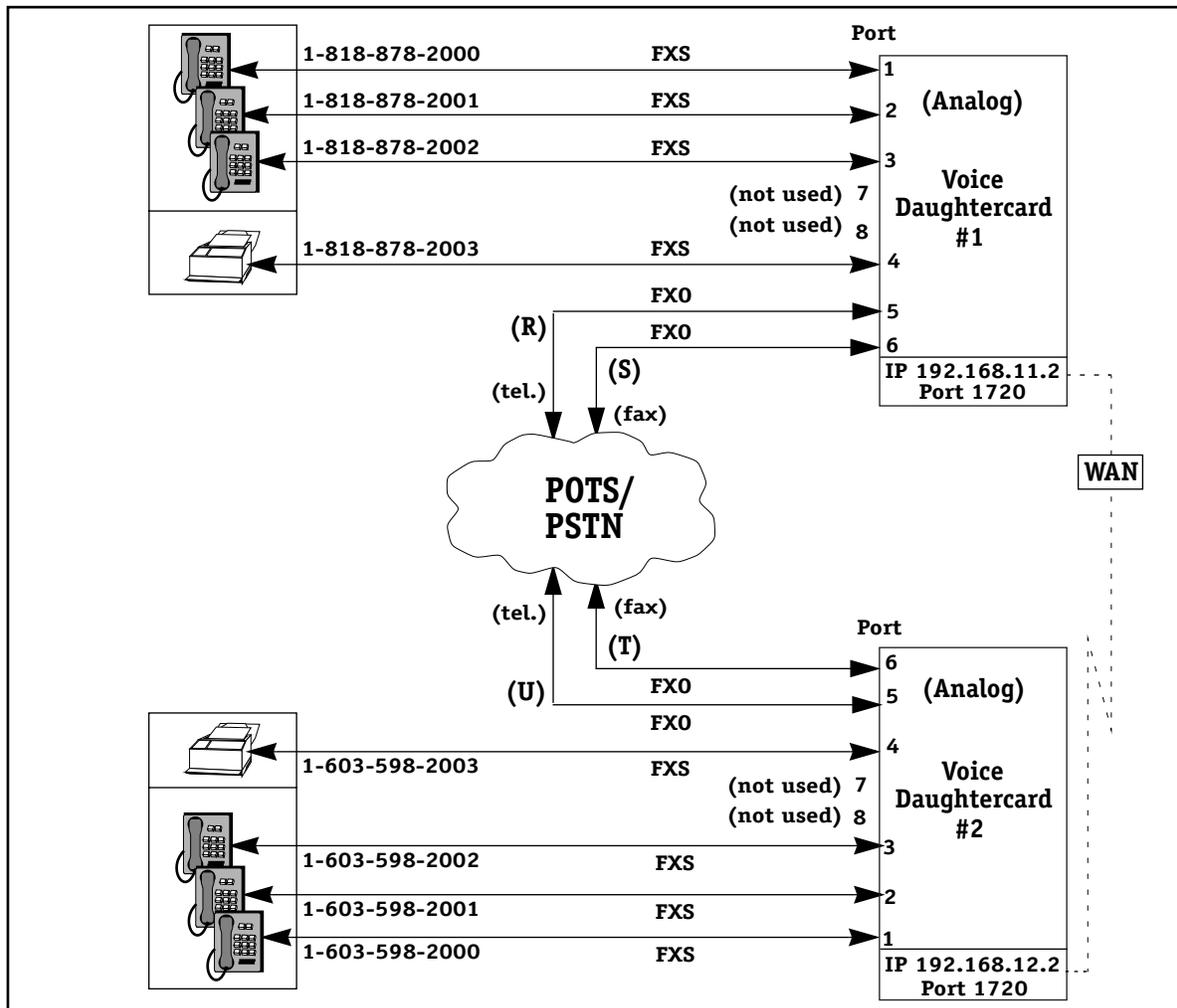
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — individual hunt groups (48 channels per group/T1) (D)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — NANP extensions (N)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — PSTN NANP (P)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Strip digit length — 1 (S)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)
Digital Interface type — FXS (AA)	<i>voice port interface type</i> (page 5-34)
Digital Interface type — FXO (AB)	<i>voice port interface type</i> (page 5-34)

VoIP Networks with PSTN — Example 18

Caller ID (Static)

This dialing scheme demonstrates how to use static caller ID with an analog voice switching daughtercard (VSA). Note that the static caller ID name and number always overrides inbound FXO caller ID name and number. See Chapter 2, “VoIP Daughtercards” for a description of the various daughtercards, and Chapter 4, “Setup and Installation” for details on installation.



Example 18 — Caller ID (Static)

LEGEND for Diagram Components

Voice Daughtercard #1 Configuration	Voice Daughtercard #2 Configuration
— Expects to receive in-seize signal on trunk (POTS line) (R) and then uses Private Line Automatic Ring-down (PLAR) to route calls to ports 1, 2, or 3.	— Expects to receive in-seize signal on trunk (POTS line) (U) and then uses Private Line Automatic Ring-down (PLAR) to route calls to ports 1, 2 and 3.
— Expects to receive in-seize signal on trunk (POTS line) (S) and then uses Private Line Automatic Ring-down (PLAR) to route calls to port 4.	— Expects to receive in-seize signal on trunk (POTS line) (T) and then uses Private Line Automatic Ring-down (PLAR) to route calls to port 4.
— Routes all POTS PSTN calls to trunks (R) or (S) .	— Routes all POTS PSTN calls to trunks (T) or (U) .

Remarks

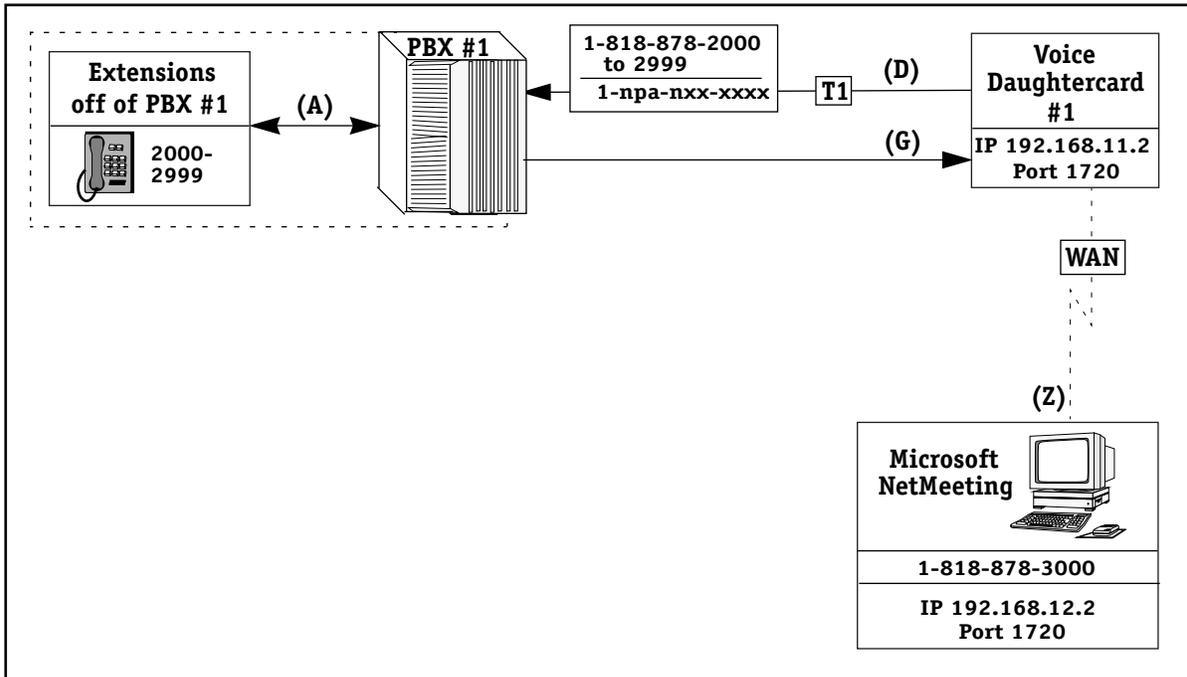
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — individual hunt groups (48 channels per group/T1) (D)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — NANP extensions (N)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — PSTN NANP (P)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — FXS (AA)	<i>voice port interface type</i> (page 5-34)
Digital Interface type — FXO (AB)	<i>voice port interface type</i> (page 5-34)

VoIP Networks with Interoperability — Example 19

H.323 Gateway to Microsoft NetMeeting (without FastStart)

This dialing scheme example is used to connect a voice daughtercard to a Microsoft NetMeeting H.323-compliant terminal with microphone.



Example 19 — H.323 Gateway to Microsoft NetMeeting

LEGEND for Diagram Components	
PBX #1 Configuration	NetMeeting Configuration
— Expects to receive eleven digits on trunk (D) and then uses these digits to route calls to lines (A).	— Expects to receive eleven digits in H.323 packets on ethernet WAN (Z).
— Routes calls starting with 2 or 1-818-878-2xxx to lines (A).	— Routes calls starting with 1-818-878-2xxx to voice daughtercard #1 via ethernet WAN (Z) using the VoIP gateway's IP address.
— Routes calls starting with 1-818-878-3000 to trunk (G), and then the VoIP network uses these digits to route calls to Microsoft NetMeeting via ethernet WAN (Z).	

Remarks

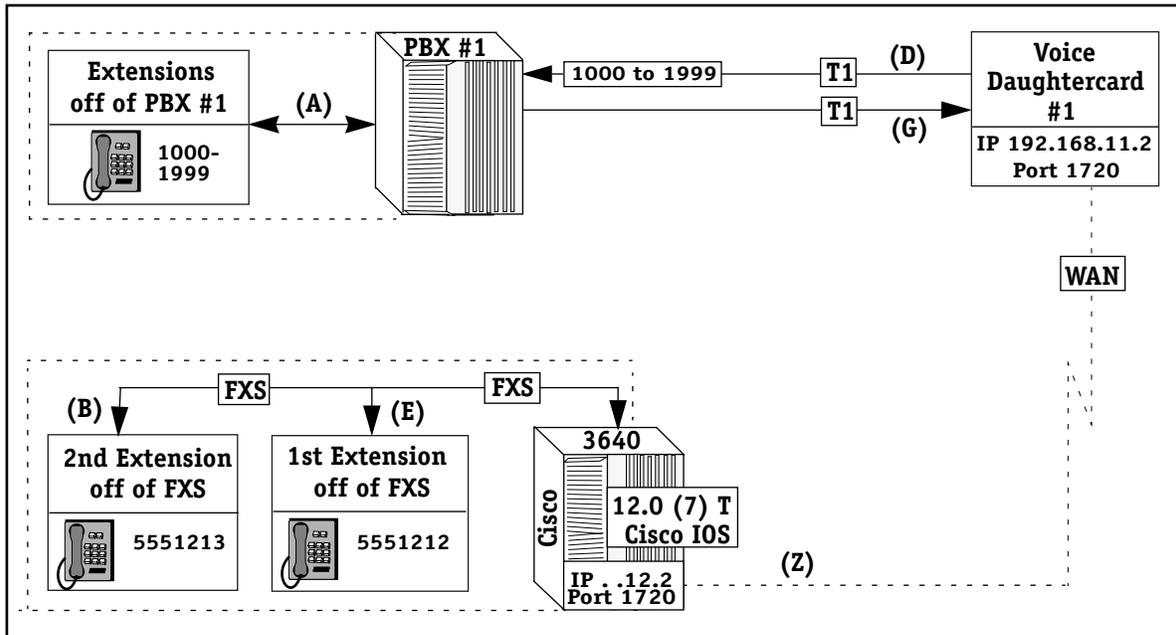
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
H.323 gateway to H.323 device (C)	<i>voice destination h.323 endpoint</i> (page 5-230)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — eleven digit local extensions (M)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Strip digit length — 7 (V)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks with Interoperability — Example 20

H.323 Gateway to Cisco Router

This dialing scheme is used to connect a voice daughtercard to a Cisco 3640 router running Cisco IOS (7) 12.0.



Example 20 — H.323 Gateway to Cisco Router

LEGEND for Diagram Components	
PBX #1 Configuration	Cisco 3640 Configuration
— Expects to receive four digits on trunk (D) and then uses these digits to route calls to lines (A).	— Expects to receive seven digits in H.3232 packet on ethernet WAN (Z), and then uses these digits to route calls to lines (B) or lines (E).
— Routes calls starting with 1 to lines (A).	— Routes 555-1213 calls to lines (B) and all 555-1212 calls to lines (E).
— Routes calls starting with 555-121x to trunk (G), and then the VoIP network uses these digits to route calls to the Cisco 3640 router via ethernet WAN (Z).	— Routes calls starting with 1 to voice daughtercard #1, and then the VoIP network uses these digits to route calls to trunk (D).

Remarks

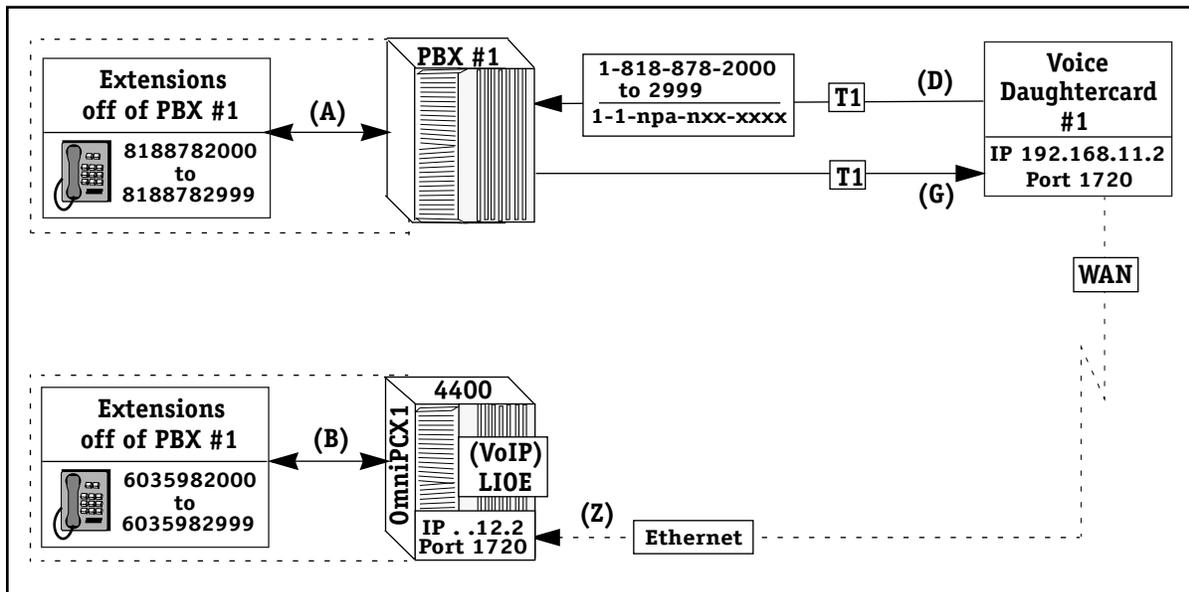
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
H.323 gateway to H.323 device (C)	<i>voice destination h.323 endpoint</i> (page 5-230)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Strip digit length — 4 (U)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks with Interoperability — Example 21

H.323 Gateway to OmniPCX 4400

This dialing scheme is used to connect an H.323 Gateway to an OmniPCX 4400 via an ethernet port (LIOE; Link Optimizer Ethernet card), and requires full compliance with H.323 Version 1 (v1); however, only H.323 v1 voice capability is currently supported by the OmniPCX. The OmniPCX does not support Codec Fax T.38. The OmniPCX 4400 also requires installation of an LIOE voice card to provide VoIP. Refer to the Alcatel OmniPCX 4400 Operations Manual for more information.



Example 21 — H.323 Gateway to OmniPCX 4400

LEGEND for Diagram Components	
PBX #1 Configuration	OmniPCX #1 Configuration
— Expects to receive eleven digits on trunk (D) and then uses these digits to route calls to lines (A).	— Expects to receive eleven digits in H.3233 packet on ethernet WAN (Z) and then uses these digits to route calls to lines (B).
— Routes calls starting with 1-818-878-2xxx to lines (A).	— Routes calls starting with 1-603-598-2xxx to lines (B).
— Routes calls starting with 1-603-598 to trunk (G), and then the VoIP network uses these digits to route calls to ethernet WAN (Z).	— Routes calls starting with 1-818-878 to voice daughtercard #1 via ethernet WAN (Z), and then the VoIP network uses these digits to route calls to trunk (D).

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

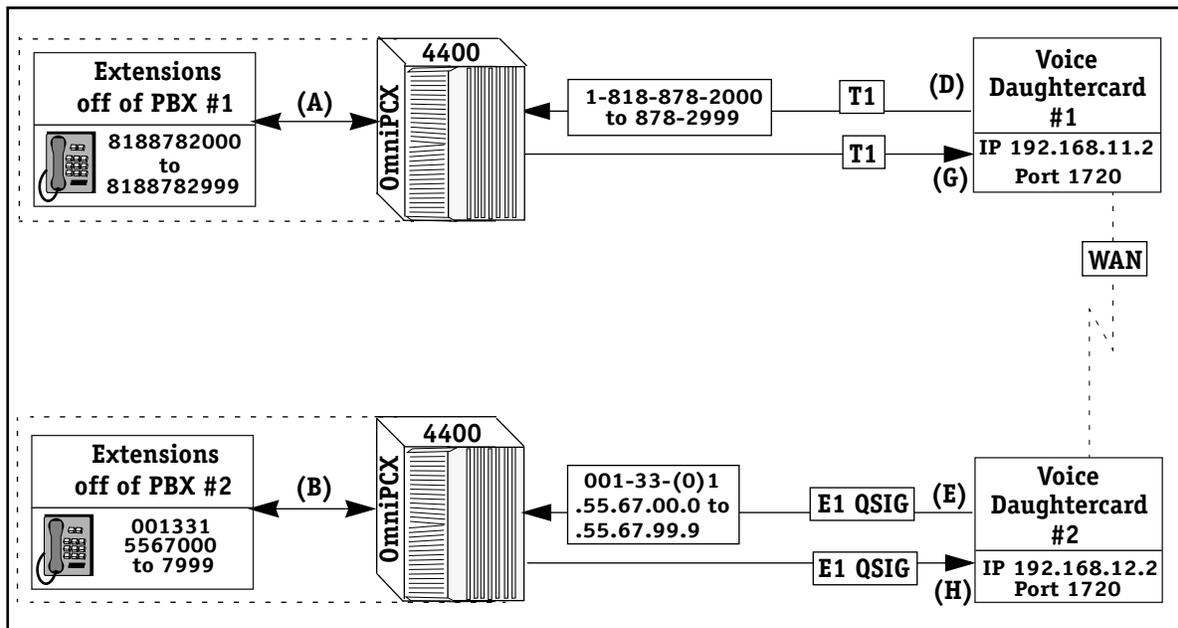
To connect to the LIOE card on the OmniPCX 4400, the voice coding profile must be set to “cp5”, the VIF (voice information field) set to 192, and the VPI (voice packet interval) set to 30 ms. See Chapter 5, “VoIP Commands,” for more details on these settings.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
H.323 gateway to H.323 device (C)	<i>voice destination h.323 endpoint</i> (page 5-230)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Site prefix — single digit (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — eleven digit local extensions (M)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

VoIP Networks with Interoperability — Example 22

OmniPCX 4400 and E1 QSIG

This dialing scheme is used to connect an OmniPCX 4400 to an E1 QSIG port on a voice daughtercard.



Example 22 — OmniPCX 4400 and E1 QSIG

LEGEND for Diagram Components	
OmniPCX #1 Configuration	OmniPCX#2 Configuration
— Expects to receive eleven digits on trunk (D) and then uses these digits to route calls to lines (A).	— Expects to receive one to 23 digits on trunk (E) and then uses these digits to route calls to lines (B).
— Routes calls starting with 1-818-878-2xxx to lines (A).	— Routes calls starting with 011-33-(0)-1.556-7xxx to lines (B).
— Routes calls starting with 001-33-(0)1.55.6 to trunk (G), and then the VoIP network uses these digits to route calls to trunk (E).	— Routes calls starting with 1-818-878 to trunk (H), and then the VoIP network uses these digits to route calls to trunk (D).

Remarks

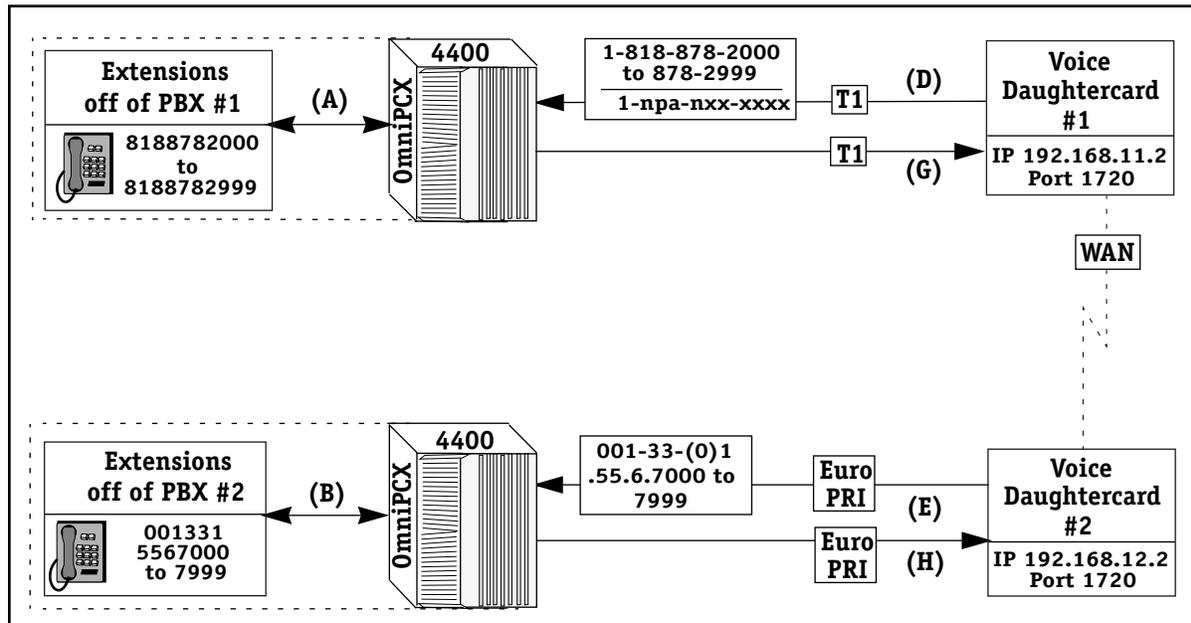
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — eleven digit local extensions (M)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — INTL extension (O)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)
Digital Interface type — E1 (QSIG) (X)	<i>voice port interface type</i> (page 5-34) <i>voice signaling companding</i> (page 5-77)

VoIP Networks with Interoperability — Example 23

OmniPCX and Euro PRI

This dialing scheme is used to connect a voice daughtercard to an OmniPCX using Euro PRI.



Example 23 — OmniPCX 4400 and Euro PRI

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive eleven digits on trunk (D) and then uses these digits to route calls to lines (A) .	— Expects to receive one to 23 digits on trunk (E) and then uses these digits to route calls to lines (B) .
— Routes calls starting with 1-818-878-2xxx to lines (A) .	— Routes calls starting with 011-33-(0)-1.556-7xxx to lines (B) .
— Routes calls starting with 001-33-(0)1.55.6 to trunk (G) , and then the VoIP network uses these digits to route calls to trunk (E) .	— Routes calls starting with 1-818-878-2xxx to trunk (H) , and then the VoIP network uses these digits to route calls to trunk (D) .

Remarks

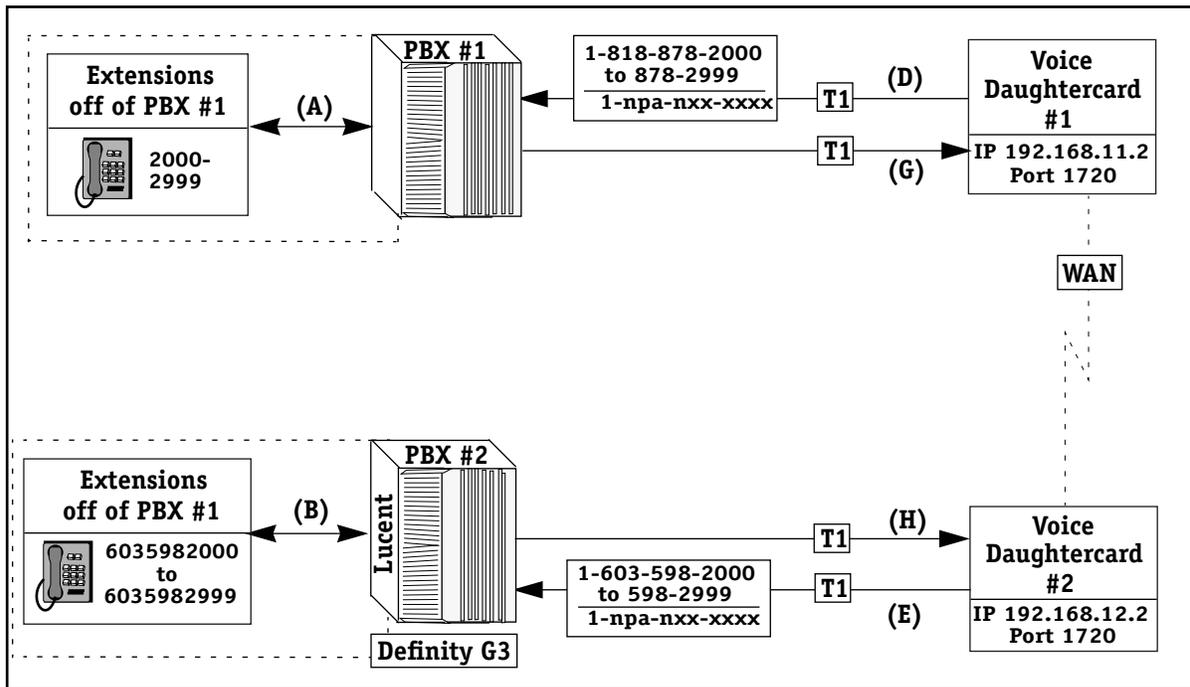
Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — eleven digit local extensions (M)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — INTL extension (O)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)
Digital Interface type — E1 ISDN PRI (Euro PRI) (Y)	<i>voice port interface type</i> (page 5-34) <i>voice signaling companding</i> (page 5-77)

VoIP Networks with Interoperability — Example 24

Other PBXs with T1

This dialing scheme is used to connect a voice daughtercard to a Lucent Definity G3 running the following operating system: System G3siV4, Software Version G34.i.04.0.054.0.



Example 24 — Other PBXs with T1

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive eleven digits on trunk (D) and then uses these digits to route calls to lines (A).	— Expects to receive eleven digits on trunk (E) and then uses these digits to route calls to lines (B).
— Routes calls starting with 1-818-878-2xxx to lines (A).	— Routes calls starting with 1-603-598-2xxx to lines (B).
— Routes calls starting with 1-603-598-2xxx to trunk (G), and then the VoIP network uses these digits to route calls to trunk (E).	— Routes calls starting with 1-818-878-2xxx to trunk (H), and then the VoIP network uses these digits to route calls to trunk (D).

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — eleven digit local extensions (M)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Strip digit length — 7 (V)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)

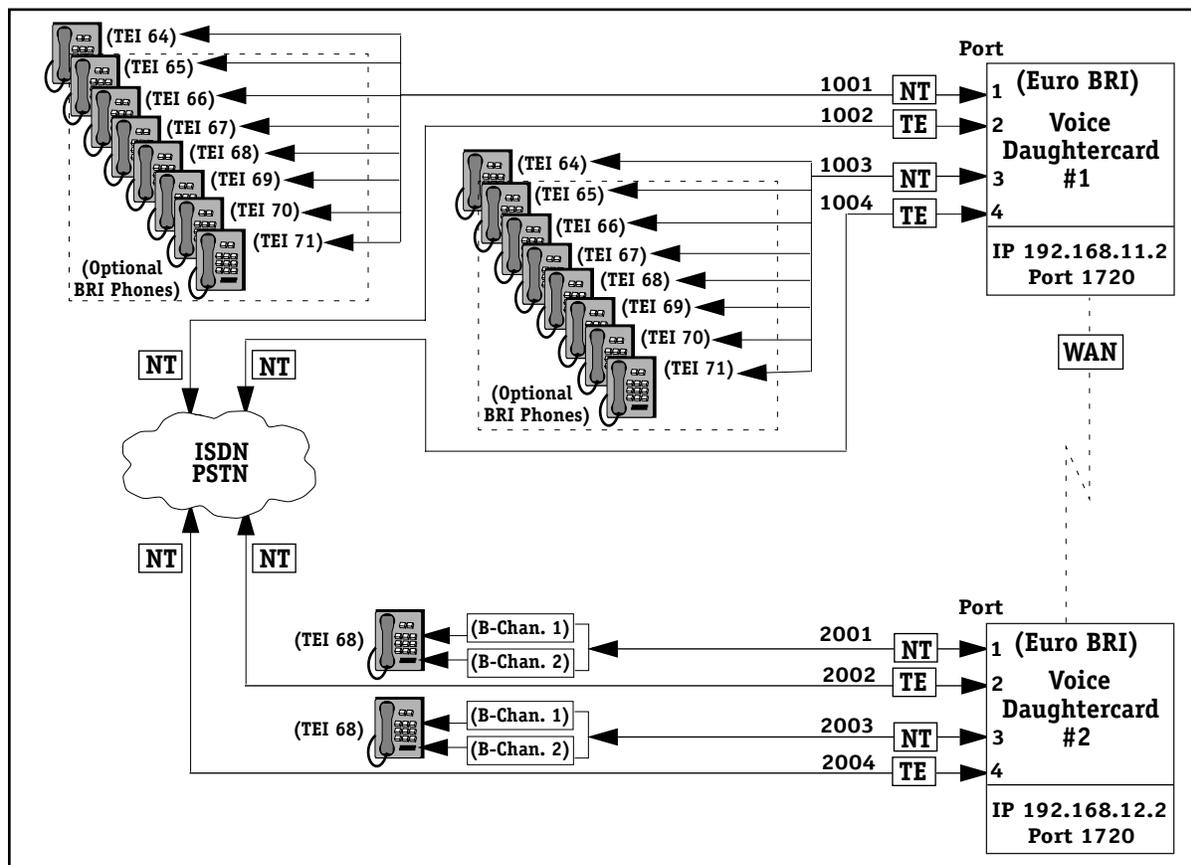
VoIP Networks with Interoperability — Example 25

Other PBXs with Euro BRI

This dialing scheme is used to connect a VSB daughtercard to a generic Euro ISDN device using Euro BRI (E1 ETSI). Note that the two NT ports on Voice Daughtercard #1 are configured as Point to Multipoint, whereas the two NT ports on Voice Daughtercard #2 are configured as Point to Point. Each Euro BRI (Point to Point NT) interface connects only to one telephone line containing two B-channel lines.

The VSB daughtercard does not support static TEI assignments. The VSB daughtercard automatically assigns TEI numbers, e.g., TEI 64, to BRI TE devices; these numbers will vary between 64 and 126. As per ITU-T Recommendation E.164 (1991), this VSB dialing scheme does not route based upon the ISDN subaddress supplementary service. Rather, the BRI devices connected to the VSB are responsible for ISDN subaddress routing. Subaddressing provides expanded addressing capacity beyond the standard ISDN number.

Installation of digital voice switching daughtercards, i.e., Euro BRI VSBs, must be installed to use this dialing scheme. VSB daughtercards also contain jumpers which must be configured in order to process VoIP calls using Euro BRI. See Chapter 2, “VoIP Daughtercards,” for details, and Chapter 4, “Setup and Installation” for further details on installation.



Example 25 — Other PBXs with Euro BRI

LEGEND for Diagram Components	
Voice Daughtercard #1 Configuration	Voice Daughtercard #2 Configuration
— Expects to receive four-digit ISDN address, e.g., 2001, on ports 1, 2, 3, or 4, and then uses these digits to route calls via the VoIP network.	— Expects to receive four-digit ISDN address, e.g., 1001, on ports 1, 2, 3, or 4, and then uses these digits to route calls via the VoIP network.
— Routes calls starting with 1001 to port 1, and calls starting with 1003 to port 3, and then the TE side of the connection routes the call to the specific TEI of a device.	— Routes calls starting with 2001 to port 1 (B-Channel 1), and calls starting with 2003 to port 3 (B-Channel 1). If B-Channel 1 is busy, then the calls are automatically routed to B-Channel 2 on the corresponding port.
— Routes calls starting with 1002 to port 2, and calls starting with 2004 to port 4.	— Routes calls starting with 2002 to port 2, and calls starting with 2004 to port 4.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — individual hunt groups (48 channels per group/T1) (D)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — Euro BRI (Z)	<i>voice port interface type</i> (page 5-34) <i>voice port isdn protocol</i> (page 5-51)

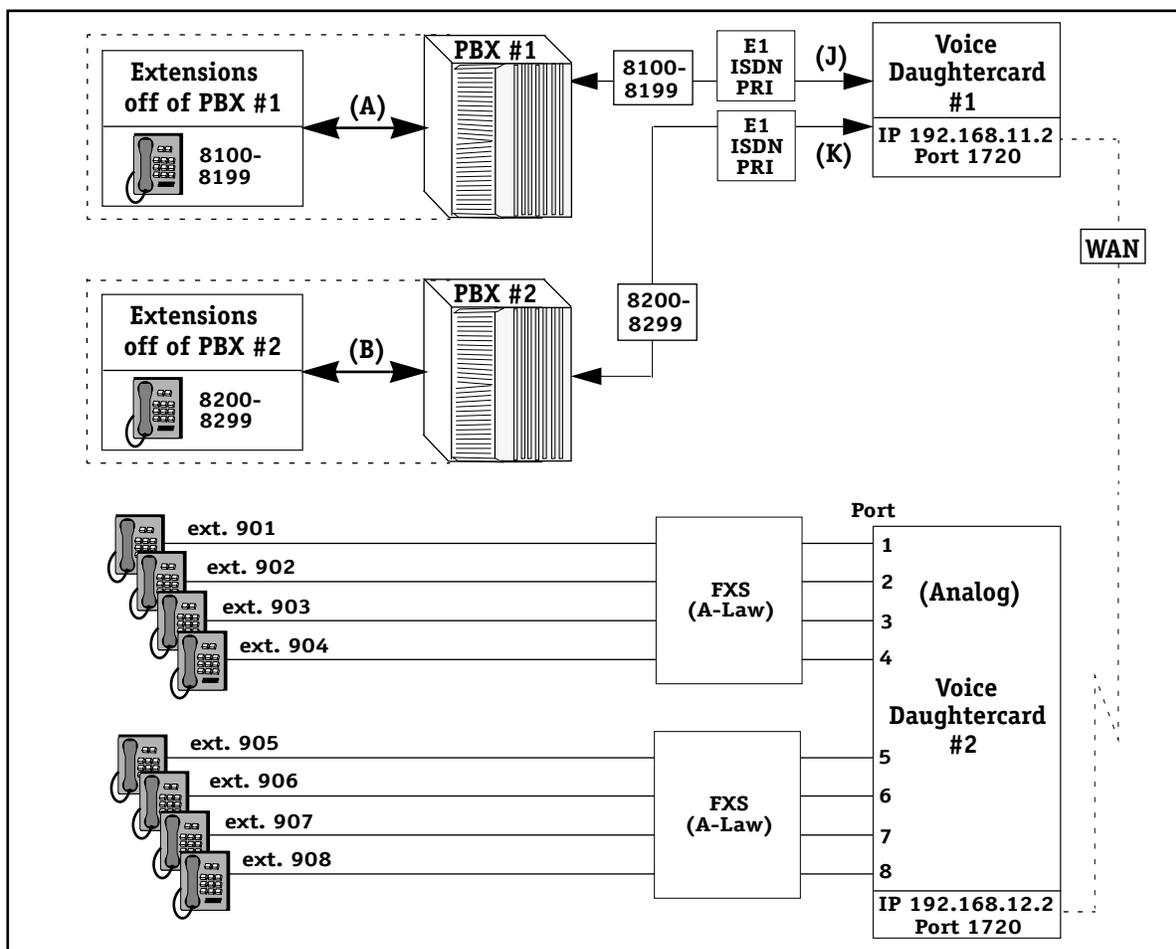
VoIP Networks with Interoperability — Example 26

Mixed European Digital and Analog Voice Daughtercards

This dialing scheme is used for European countries using E1 ISDN PRI (ETSI) with analog and digital VoIP gateways together on the same VoIP network. A-Law companding on the VSA is generally used in Europe (instead of the North American Mu-Law companding).

On the VSA, the Foreign Exchange Station (FXS) Loop Start lines are used to send calls to the VoIP network from the telephones via analog trunks; all digits are automatically stripped on incoming FXS calls.

Installation of an analog voice switching daughtercard (VSA with FXS module) is also required in addition to the standard VSD. See Chapter 2, “VoIP Daughtercards” for a description of the various daughtercards, and Chapter 4, “Setup and Installation” for details on installation.



Example 26 — Mixed European Digital and Analog Voice Daughtercards

LEGEND for Diagram Components	
PBX #1 Configuration	PBX #2 Configuration
— Expects to receive four digits starting with 81 on trunk (J), and then uses these digits to route calls to lines (A).	— Expects to receive four digits starting with 82 on trunk (K), and then uses these digits to route calls to lines (B).

LEGEND for Diagram Components	
— Routes four digits calls starting with 82 to trunk (J) , and then the VoIP network uses these digits to route calls to trunk (K) .	— Routes four digits calls starting with 81 to trunk (K) , and then the VoIP network uses these digits to route calls to trunk (J) .
— Routes three digit calls starting with 901-908 to trunk (J) , and then the VoIP network uses these digits to route calls to the corresponding port (line) on Voice Daughtercard #2.	— Routes three digit calls starting with 901-908 to trunk (K) , and then the VoIP network uses these digits to route calls to the corresponding port (line) on Voice Daughtercard #2.

Remarks

Supported VoIP features and main CLI commands used with this dialing scheme are as follows.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — E1 ISDN PRI (Euro PRI) (Y)	<i>voice port interface type</i> (page 5-34) <i>voice signaling companding</i> (page 5-77)
Digital Interface type — FXS (AA)	<i>voice port interface type</i> (page 5-34)

AVNDS Master List of Features by CLI Command

Refer to the table below for a list of VoIP features used in the AVNDS, along with the most common commands used for setting up each feature. Corresponding page numbers for the *VoIP CLI Commands* chapter are also included.

Features Supported	Primary CLI Commands Used
H.323 gateway to voice daughtercard (A)	<i>voice destination local channel</i> (page 5-231)
H.323 gateway to H.323 gatekeeper (RADVision) (B)	<i>voice network card b.323 gatekeeper control</i> (page 5-217) <i>voice network b.323 gatekeeper mode</i> (page 5-218) <i>voice network b.323 gatekeeper address</i> (page 5-219) <i>voice network b.323 gatekeeper associate</i> (page 5-223)
H.323 gateway to H.323 device (C)	<i>voice destination b.323 endpoint</i> (page 5-230)
Local channel — individual hunt groups (48 channels per group/T1) (D)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Local channel — four hunt groups (12 channels per group/T1) (E)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Local channel — two hunt groups (24 channels per group/T1) (F)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Local channel — one hunt group (48 channels across two T1s) (G)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Local channel — one hunt group (60 channels across two E1s) (H)	<i>voice numbering plan hunt method</i> (page 5-255) <i>voice numbering plan destination member</i> (page 5-257) <i>voice numbering plan phone group member</i> (page 5-258)
Site prefix — no site prefix (I)	<i>voice phone group site prefix</i> (page 5-238)
Site prefix — single or multiple digits (J)	<i>voice phone group site prefix</i> (page 5-238) <i>voice phone group site prefix digits</i> (page 5-239)
Voice phone group type — three digit local extensions (K)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — four digit local extensions (L)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — eleven digit local extensions (M)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — NANP extensions (N)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)

AVNDS Master List of Features by CLI Command

Features Supported	Primary CLI Commands Used
Voice phone group type — INTL extension (O)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — PSTN NANP (P)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Voice phone group type — PSTN INTL (Q)	<i>voice phone group type</i> (page 5-240) <i>voice phone group format</i> (page 5-243) <i>voice phone group add numbers</i> (page 5-247)
Strip digit length — no strip digits (R)	<i>voice phone group strip digit length</i> (page 5-244)
Strip digit length — 1 (S)	<i>voice phone group strip digit length</i> (page 5-244)
Strip digit length — 2 (T)	<i>voice phone group strip digit length</i> (page 5-244)
Strip digit length — 4 (U)	<i>voice phone group strip digit length</i> (page 5-244)
Strip digit length — 7 (V)	<i>voice phone group strip digit length</i> (page 5-244)
Digital Interface type — T1 (W)	<i>voice port interface type</i> (page 5-34)
Digital Interface type — E1 (QSIG) (X)	<i>voice port interface type</i> (page 5-34) <i>voice signaling companding</i> (page 5-77)
Digital Interface type — E1 ISDN PRI (Euro PRI) (Y)	<i>voice port interface type</i> (page 5-34) <i>voice signaling companding</i> (page 5-77)
Digital Interface type — Euro BRI (Z)	<i>voice port interface type</i> (page 5-34) <i>voice signaling companding</i> (page 5-77)
Digital Interface type — FXS (AA)	<i>voice port interface type</i> (page 5-34)
Digital Interface type — FXO (AB)	<i>voice port interface type</i> (page 5-34)

4 Setup and Installation

This chapter describes the setup and installation procedures for VoIP in the switch. General operations between the components are discussed in Chapter 1, “VoIP Overview.” These instructions apply to the more basic installations of VoIP on an OmniAccess 512 or Omni Switch/Router.

Components of VoIP

VoIP requires installation of the following Alcatel and third-party components.

Components	Description
VoIP Components	<p>•Voice Hardware and Software Modules (for hardware details see also Chapter 2, “VoIP Daughtercards”).</p> <p>Alcatel’s VoIP consists of a dynamically loadable switch software module (vsoa.img, vsx.img) intended to bring Voice over IP to switched networks by means of voice modules (i.e., voice switching daughtercards) installed in either the OmniAccess 512 or the Omni Switch/Router. At least one of the available daughtercards and associated switch hardware modules must be installed per switch (and/or other IP telephony device) to provide a VoIP gateway:</p> <p>Digital Voice Switching Daughtercard (VSD); supports T1, E1 (Euro PRI/E1QSig)</p> <p>Digital Voice Switching Daughtercard (VSB); supports Euro ISDN BRI</p> <p>Analog Voice Switching Daughtercard (VSA); supports Analog Fax (with FXS or FXO modules).</p> <p>To run VoIP on the OmniAccess 512, the following software modules must be installed:</p> <ul style="list-style-type: none"> oa5.img vsoa.img oa5_text_cfg.img oa5drv.img (for WAN connectivity; like wsx.img) t1e1drv.img (for WAN connectivity; uplink ports) qos.img (recommended but not required) <p>To run VoIP on the OSR, the following software modules must be installed:</p> <ul style="list-style-type: none"> mpx.img vsx.img text_cfg.img wsx.img (for WAN connectivity) t1e1drv.img (for WAN connectivity; T1/E1 uplink ports) qos.img (recommended but not required) esx.img (if Ethernet or Gigabit Ethernet modules are installed)
	<p>•AVNDS (Alcatel VoIP Network Dialing Scheme) (*)</p> <p>VoIP requires installation of a master configuration boot file (vsmboot.asc) which contains pre-configured operational settings for the entire VoIP network. This file must be merged with one or more available dialing schemes, i.e., vsmboot##.asc [01-24] files. (See Chapter 3, “Network Dialing Schemes,” for details.) Files may be copied and merged using a standard text editor.</p> <p>•Command Line Interface (CLI) Configuration Tool (*)</p> <p>VoIP includes a CLI text-based configuration tool which must be accessed via the AVNDS master (vsmboot.asc) file.</p> <p>(*) Compatible with PC or UNIX workstation running either TELNET or FTP (File Transfer Protocol).</p>

Components	Description
Compatible Alcatel and Third-Party Components	<p>At least one of the following IP telephony devices must be used in conjunction with Voice over IP in the switch; contact customer support for the most current list. <i>(Other generic PBXs and Key Systems have not been tested, but may be used at customer's discretion.)</i></p> <ul style="list-style-type: none">• Alcatel OmniPCX 4400• Lucent Definity G3 PBX; Partner ACS Key System• Nortel Option 11 PBX; ICS Key System• Channel bank (for test purposes) <hr/> <ul style="list-style-type: none">• H.323 Gatekeeper (optional) NT100 RADVision H.323 gatekeeper software recommended but other H.323 gatekeepers may be used. See vendor's operational manual for install instructions.

Assumptions and Recommendations

The following assumptions and recommendations should be considered before installing VoIP in the switch:

- Existing IP network is fully operational and ready for installation of VoIP. Some key elements to consider includes adequate bandwidth, reliable, end-to-end, round trip packet delay time, and quality of service (QoS), for example. See the Quality of Service (QoS) user guide for more information.

◆ Note ◆

Alcatel's H.323 VoIP Gateway automatically prioritizes all VoIP calls via QoS ToS (Type of Service) controls. For more information on ToS, see the switch manual.

- A VoIP dialing scheme has been developed to accommodate enterprise-specific call capabilities. Both IP network and telephone networks must be converged using an appropriate dialing scheme that has been customized to the enterprise. See Chapter 3, "Network Dialing Schemes," for guidelines in using and developing an AVNDS dialing scheme.
- All related configuration requirements for H.323 Gatekeeper, OmniPCX or other third party PBX or key system have been met. For test purposes, channel banks may be substituted for PBXs. Refer to the vendor supplied product manuals for more information.
- Whenever any subsequent changes are made to the VoIP configuration after installation is complete, the impact of those changes on other connected devices should first be considered and properly coordinated, e.g., if PBX is connected to VoIP switch, updates to VoIP should be coordinated with PBX administrator.

Configuration Restrictions

A maximum of two (VSMs) (Voice Switching Modules, i.e. VSA/VSB/VSD) are supported in a chassis. These VSMs may be placed in the same VLAN or different VLANs; however, no other ports may be assigned to these VLAN(s). As a result, all VSM traffic must be routed in order to reach VLANs on remote devices, and non-VSM ports and/or devices may not be added to the VSM VLANs.

VSM VLANs must have spanning tree disabled, and must not send routing updates (RIP, OSPF, or BGP) into the VLAN. The VSM uses the default gateway defined in the **vsmboot.asc** text configuration file to route traffic to remote destinations. The subnet mask on the default gateway must match the subnet mask defined for the VSM in the **vsmboot.asc** file.

The VSM MAC address should be statically added to the VSM port to avoid 5 to 6 seconds of silence at the start of a call, when not used with a gatekeeper. Optionally, the VSM port can be set to optimized mode, and only the first call placed will have the period of silence.

General Installation Procedures

These instructions apply to all voice daughtercards being installed in either the OmniAccess 512 and Omni Switch/Router, unless otherwise specified, but are geared toward setting up an OA 512 voice network using a T1 VSD card as an example configuration. At least two communicating devices, e.g. two VoIP switches, or a VoIP switch and an OmniPCX 4400, must be configured to provide VoIP. Components required to implement VoIP in the switch, as described above, must be set up and configured as follows.

1. Start HyperTerminal or some other terminal emulation program on a workstation connected to the switch via the console port.
2. Apply power to switch and watch for switch IP address to be displayed on HyperTerminal. When a message similar to the following appears, write down the displayed IP address as it will be required for downloading switch files in the next step:

“Configuring default VLAN hostname no-name at addr 127.0.0.0”

3. Using TELNET or FTP, download all required VoIP files to the switch flash as per the component table above. Contact customer service if these files cannot be located.
4. Locate the master **vsmboot.asc** file, and the file for the selected dialing scheme **vsmboot##.asc [01-24]** to use as per Chapter 3, “Network Dialing Schemes,” and then copy the files to a Network Administrator’s workstation connected to the VoIP switch. A dialing scheme must be selected as there is no default dialing scheme in place. Contact customer service if these files cannot be located.
5. Insert contents of selected dialing scheme file **vsmboot##.asc [01-24]** into master **vsmboot.asc** file above the “**voice daughtercard activate**” line, and then save the master **vsmboot.asc** file by the same name in a different directory so as not to overwrite the original master **vsmboot.asc** file. Do not change the name of the filenames, and keep them lowercase as they are case sensitive. Do not use any type of **view** commands in any VSM boot files at any time as they may cause the switch to time-out or hang. Refer to Appendix A, “VSM Boot Files,” to see complete boot file examples.

The last two lines of the **vsmboot.asc** file must always be the “**voice daughtercard activate**” line followed by a comment mark (!) only on the next line, otherwise the file will not load properly later. It is important that all entries be kept in the same order in which they appear in the master **vsmboot.asc** file, as the incorrect order will negatively impact boot time.

For a visual reference, see the example dialing scheme file following these instructions. Dialing Scheme Example 1 has been selected and used to demonstrate portions of a dial scheme (**vsmboot01.asc**) merged with the master **vsmboot.asc** file, which was then modified and saved to create a modified **vsmboot.asc** file.

Multiple dialing schemes can be merged into the master **vsmboot.asc** file, but only the master boot file (or a modified copy of it) on the flash can be used to configure and boot a VoIP network. Note that there is no default dialing scheme. A standard text editor may be used to merge and modify the files (Step 6).

Most CLI configuration data required by the switch to run VoIP are pre-installed as default settings on the voice daughtercard when the VoIP image file is downloaded to the switch; as a result, the master **vsmboot.asc** file contains only the minimum configuration data, namely, commands which must be modified to be enterprise-specific.

6. Modify the following entries in the merged **vsmboot.asc** file to be enterprise-specific, and then save the file again; also, be sure to make adjustments relative to standard switch settings (e.g., slot, port) in any CLI commands in which they are necessary. The default settings can be changed by entering the appropriate CLI command(s) in the **vsmboot.asc** file as per Chapter 5, “VoIP Commands.” For assistance when entering certain values such as slot, port and channel numbers in the command syntax, refer to the configuration table at the beginning of that chapter.

Depending on the dialing scheme selected, some command entries may need to be added altogether, such as those required for coding profiles. Other than the minimum configuration entries, there are no visual representations of the default settings in any **vsmboot.asc** file unless they are specifically entered in the file.

All configuration changes must be saved to this modified **vsmboot.asc** file or they will not take effect. Changes should be made in accordance with the CLI reference instructions in Chapter 5, “VoIP Commands.” Optional settings should only be changed if the default settings do not apply. Any modifications made through the CLI directly will be lost on reboot, as settings contained in the **vsmboot.asc** file override all other settings.

◆ **Note** ◆

Once the card has been configured and is operational, it can be reconfigured later without rebooting the switch using the dynamic reconfiguration feature via the **voice numbering plan activate** command. See Chapter 5, “VoIP Commands.”

The entries below are not necessarily listed in the order they appear in the master **vsmboot.asc** file; to reduce boot time for large files, lines which are commented out (!) may be removed. Depending on the number of CLI commands in the modified **vsmboot.asc** file, it may take from three to 15 minutes to load depending on the complexity of the configuration.

Switch Settings (Required)

- IP address (unique), IP [subnet] mask and the default gateway addresses for the voice daughtercard.

◆ **Notes** ◆

A separate routing group just for the VoIP gateway(s) must also be created on the switch, and Spanning Tree for that group must be turned OFF; see the appropriate switch manual for further details on using groups and Spanning Tree.

The IP mask and default gateway addresses in the **vsmboot.asc** file must match the IP subnet mask and the IP broadcast address used in the routing group for the VoIP gateway(s).

Alcatel VoIP (VSD, VSB, VSA) modules *cannot* be in a Virtual LAN (VLAN) with non-voice ports (i.e. data ports), IP phone ports, etc. All voice traffic *must* route in and out of the VoIP VLAN.

- voice port interface type (e.g., T1, E1; *required only for digital voice daughtercard)

AVNDS Dialing Scheme Settings (Required) — The following configuration data, obtained from the dialing scheme selected as per Step 4, will vary depending on the dialing scheme.

- voice destination (ports/channels to which VoIP calls are routed)
- voice phone group (telephone numbers that callers may dial)
- voice numbering plan (associated voice phone groups and destinations)
- multiple PSTN connections (*required only for sites using Public Switched Telephone Network)

Except when local channels are used, all AVNDS commands function with the H.323 endpoints, e.g., OmniPCX, Cisco Routers, Microsoft NetMeeting.

Convergence Settings (Optional)

- voice signaling channels (e.g., E&M, FXO and FXS signaling protocols; *required if not using E&M Immediate Start protocol. *Required only in Europe; FXS ring cadence coefficient and ring cadence ID.
- voice switching daughtercards (e.g., dialing timers)
- coding profiles (codec type, e.g., voice, fax; *required only if connecting to other type of H.323 device, including OmniPCX 4400 with LIOE (Link Optimizer Ethernet [voice] card, Cisco Router, and Microsoft NetMeeting)
- Voice (VoIP) network (*required only if using H.323 gatekeeper software)

7. Configure the jumpers on the voice daughtercard being installed as required and detailed in Chapter 2, “VoIP Daughtercards.” Proceed to Step 8 only after the jumpers (if necessary) have been properly configured in the order in which they are presented in Chapter 2, “VoIP Daughtercards.” Damage to voice daughtercards may result if jumpers are not properly configured.
8. Turn OFF power to switch, then insert and seat daughtercard configured as per Step 7 above; do not proceed with Step 8 unless jumpers on the card are properly configured. Refer to Chapter 2, “VoIP Daughtercards,” for configuration details.

For OA 512 with digital daughtercard; insert (DIMMs side up) into empty expansion slot #4.

For OA 512 with analog daughtercard; insert (FXO, FXS module side up) into empty expansion slot #4.

For OSR with digital daughtercard; insert (DIMMs side up) into VSX (HSX-H) motherboard in left most position (connector J17), and (connector J15) for a second card. Once voice daughtercard(s) seated, locate VSX into empty slot (2-9); MPX must be installed in slot (1).

For OSR with analog daughtercard; insert (FXO, FXS module side up) into VSX (HSX-H) motherboard in left most position (connector J17), and (connector J15) for a second card. Once voice daughtercard(s) seated, locate VSX into empty slot (2-9); MPX must be installed in slot (1).

9. Connect port on voice daughtercard to PBX or other device using straight-through or cross-over cable (straight-through recommended).

For digital voice daughtercard only; if communication link error occurs between switch and PBX or Key Set, turn OFF switch, remove daughtercard, and then flip blue T1 cross-over toggle switch on the top side of the board to swap the T1 transmit and receive connections for the designated port.

10. Connect VoIP switch to external components, e.g., OmniPCX 4400, and then set these devices so that no calls can be processed until setup and installation is complete.

For analog daughtercard (with FXO, FXS module) only; confirm the analog (VSA) daughtercard (North American or European version) being installed is the correct board for the location in which it will be used, and that the ring cadence is appropriately set for that location (if necessary, contact Customer Support for further assistance); also, confirm that all phones/fax machines are connected to the switch (VSA) are on hook. For information on the ring cadence CLI command, see Chapter 5, “VoIP Commands.”

11. Reapply power to switch to load all contents of the master **vsmboot.asc** file. Watch console for VoIP error conditions as not all errors display in the error file. For an OA 512, when the file has completely loaded, the following screen message appears:

“Configuration complete for voice daughtercard in slot 4, submodule 1.”

◆ Notes ◆

The screen message will be different for OSRs depending on the slot number entered in the modified **vsmboot.asc** file.

The Reboot command is not applicable to VoIP in the switch. The switch must be powered ON/OFF instead.

12. From the console screen, type **ls** at the prompt. Look for any “**vsmboot.asc.[01-99].err**” files. The list of files that would typically display for the OA 512 are shown below:

```
oa5.cmd
oa5.cnf
oa5.cfg
mpm.log
oa5isdn.img
oa5diag.img
t1e1drv.img
oa5drv.img
oa5.img
oa5_text_cfg.img
vsoa.img
vsmboot.asc
```

13. If the error file is listed in addition to the above files, correct the offending CLI command(s) in the modified master **vsmboot.asc** file, save the file and download it again to the flash.
14. Turn OFF power to the switch, and then repeat Steps 10 and 11, and if necessary, Steps 12 and 13.
15. Set VoIP switch, including other digital devices, e.g., OmniPCX 4400, connected to VoIP switch to route calls to VoIP network. Analog devices should already be set as per Step 9. Assuming the entire VoIP network and the network dialing scheme (AVNDS) were properly configured, the VoIP switch is now ready to receive and route calls.

Instructions for Additional VoIP Installations

VoIP should be setup and installed in another switch in the same manner as described above with the following main exceptions:

- Do not download the modified master **vsmboot.asc** file to the next VoIP switch installation. A copy of that modified boot file should first be created, saved to a *different* directory, and then modified to be specific to the next installation.
- All modifications to the master **vsmboot.asc** file must be done *before* downloading it to the

flash on the next VoIP switch installation.

Example VSM Boot File (`vsmboot.asc`)

The VSM boot file example on the following page matches Dialing Scheme Example 1 as depicted in Chapter 3, “Network Dialing Schemes.” To modify the `vsmboot.asc` file to match the dialing scheme shown in the diagram, the IP address mask and the IP address for the voice switching daughtercard must be changed; entry of the default IP address gateway is also required. The four digit extensions unique to the VoIP network must also be specified, in addition to slot, port and channel numbers, among other pertinent details as shown below. The modifications are indicated in the VSM boot file example which depicts the corresponding CLI commands.

```

!-----
!--
!-- Filename: vsmboot.asc
!-- Description:
!--     Enterprise Dialing Scheme between two OA 512s
!--     Dialing Scheme with four-digit extensions
!--
!-----
!
!-- This is a minimum voice text config. file. The voice daughtercard will
not start up properly if not installed on the switch before power is
applied.
!
voice echo on
!-----
*!voice daughtercard
!-----
!
IP Mask must } voice daughter card 4/1 ip mask 255.255.255.0
be listed first }
!
voice daughtercard 4/1 ip address 127.0.0.0
!
voice daughtercard 4/1 ip default gateway 127.0.0.1
*!
!-----
!voice port commands
!-----
!
voice port 4/1 interface type T1
*!
!-----
!voice coding profile (optional)
!-----
*!
!-----
!voice signaling channel (optional)
!-----
!
Insert AVNDS }
CLI commands } ! AVNDS
in this section }
!-----
!
!voice destination
!-----
Create Local }
Destination } voice destination "VSD_1" h.323 192.168.200.31 1720
               } voice destination "VSD_2" h.323 192.168.200.32 1720
               }
!
voice destination "To VSD_1 port 1" local channel "VSD_1"/1/1-24
voice destination "To VSD_1 port 2" local channel "VSD_1"/2/1-24
!

```

```

!
!
!-----
*!voice phone group
!-----
!-- This corresponds to the Trunk labeled "T1 (D)" on dialing scheme
example 1.
!
voice phone group "Ext. of PBX_1"
voice phone group "Ext. of PBX_1" site prefix off
voice phone group "Ext. of PBX_1" type local extension
voice phone group "Ext. of PBX_1" format "xxxx"
voice phone group "Ext. of PBX_1" add numbers "1000" to "1999"
voice phone group "Ext. of PBX_1" strip digit length 0
!
voice phone group "Ext. of PBX_2"
voice phone group "Ext. of PBX_2" site prefix off
voice phone group "Ext. of PBX_2" type local extension
voice phone group "Ext. of PBX_2" format "xxxx"
voice phone group "Ext. of PBX_2" add numbers "2000" to "2999"
voice phone group "Ext. of PBX_2" strip digit length 0
!
!-----
*!voice numbering plan
!-----
!-- This corresponds to the Trunk labeled "T1 (E)" on dialing scheme
example 1.
!
voice numbering plan "to PBX_1"
voice numbering plan "to PBX_1" hunt method round robin
voice numbering plan "to PBX_1" associate destination member "To
VSD_1 port 1"
voice numbering plan "to PBX_1" associate phone group member "Ext.
of PBX_1"
voice numbering plan "to PBX_1" description "Trunk to route calls from
VSD_1 to PBX1"
!
voice numbering plan "to PBX_2"
voice numbering plan "to PBX_2" hunt method round robin
voice numbering plan "to PBX_2" associate destination member "To
VSD_2 port 1"
voice numbering plan "to PBX_2" associate phone group member "Ext.
of PBX_2"
voice numbering plan "to PBX_2" description "Trunk to route calls from
VSD_2 to PBX2"
!
!-----
*!voice daughtercard activate
!-----
!
voice daughtercard 4/1 activate
!

```

Prefix
Dialing Type
Format
Nos. to Include
In Range of Digits
Strip Digits

Outgoing
Hunt Method
Associate
Destinations
Associate
Phone Groups

5 VoIP Commands

The following chapter contains information on using the VoIP command line interface (CLI) to configure VoIP switches. The commands are divided into seven major categories consisting of the following subcategories as listed below.

Command Category	Command Subcategory
Voice Switching Daughtercards	<ul style="list-style-type: none"> • Dialing Timers • Gateway Mode • Voice Ports
Channel Properties	<ul style="list-style-type: none"> • Voice Channel Mode • PLAR (Private Line Automatic Ringdown) • Voice Channel Initialization
Telephony Signaling/Interface	<ul style="list-style-type: none"> • General Signaling • Ear & Mouth (E&M) • Foreign Exchange Station (FXS) • Foreign Exchange Office (FXO) • Call Signaling Capabilities • Inbound/Outbound Caller ID • Call Progress Tones • Echo and Acoustic Echo Cancellers • Overrides for Call Signaling
Coding Profiles	<ul style="list-style-type: none"> • Codecs • Voice Mode Parameters • Voice Network Buffers • Voice Activity Detection • Tone Detection • Echo Canceller • Facsimile Modem • Facsimile T.38 Modem • Silence Detection • General Caller ID
Voice Network/Interface	<ul style="list-style-type: none"> • H.323 Gateway Discovery • H.323 Gateway Configuration • H.323 Gateway Operations
Network Dialing Scheme	<ul style="list-style-type: none"> • Destinations • Phone Groups/Parameters • Numbering Plans/Hunt Method
System-Wide VoIP	<ul style="list-style-type: none"> • Voice Switching Daughtercard Parameters • General Telephony and Telephony Channels • Voice Play Out • Voice Daughtercard (DSP Receive and Transmit) • Errors • Modem, Fax and ISDN statistics

To use this chapter, refer to the command task list below to find the page number for a specific task. The commands use a simple, line-at-a-time prompt and response scheme. The CLI interface presents a single prompt character at the beginning of each command line; however, this does not apply to the **vsmboot.asc** file in which there is no response. For details on the **vsmboot.asc** file and other similar type files, see Chapter 4, “Setup and Installation,” which contains an example of a partial VoIP boot file.

◆ Note ◆

Do *not* use any CLI **view** or **reset** commands from within the aforementioned VSM boot files at any time.

The CLI text-based commands used in VoIP are intended for use by Network Administrators and technical staff to configure Alcatel switches for VoIP. Commands are not case sensitive unless otherwise stated; however, if a name or string is used in a command it will be case sensitive. Commands which may apply to either E1, E1 ISDN PRI, or BRI Euro, are in general, collectively referenced in command names under E1.

Typically, command tasks which begin with “specify” have more than two parameters from which to choose, whereas command tasks beginning with “set” are generally an either/or type command, e.g. **on** or **off**.

When entering certain values such as slot, port and channel numbers in the command syntax, refer to the configuration table on the following page for valid entries.

VoIP Parameters, Coding Profiles and Other Settings

VoIP configuration parameters include system-wide configuration as well as per channel, per port, and per daughtercard configuration. System-wide configuration includes voice signaling interfaces, coding profiles, and the VoIP network interface (including H.323 gateway and gatekeeper configuration). Per channel configuration includes general channel and channel-level telephony signaling configuration.

Coding profiles, in particular, are important parameters that define the way a device such as a VoIP H.323 gateway card acts. Once a coding profile is available to a channel in the network, it remains in effect until or unless it is made unavailable to the channel. If any individual parameter of the profile is modified, it will take effect on all entities to which it had been defined until the next time the profile is requested from the voice switching daughtercard. *Coding profiles are not assigned to a physical entity because the coding profile to be used is determined at runtime.* If coding profile parameters are changed, the next time a modified coding profile is requested by the voice switching daughtercard, the new information will be obtained. Coding profiles are also used to associate specific entities (daughtercard with channels) on a voice switching daughtercard.

Other voice switching daughtercard parameters may also be applied in a manner similar to coding profiles, such as phone groups and numbering plans.

Phone groups and voice numbering plans are used as containers for VoIP gateway operations to group parameters so that all related parameters can be referenced (by other commands) using a single name, e.g., “topbx”. The phone group or numbering plans must be created before parameters can be added to them.

Many other parameters available for configuring VoIP networks are typically used in conjunction with Alcatel’s patent-pending Alcatel VoIP Network Dialing Schemes (AVNDS). See Chapter 3, “Network Dialing Schemes,” for more information.

GENERAL CONFIGURATION TABLE		
Parameter	Value	Range
Slot	Physical	<ul style="list-style-type: none"> •Slot Number OmniAccess 512 value is always 4. Omni Switch/Router range is 2 to 3, 2 to 5, or 2 to 9 depending on chassis size.
Card	Physical	<ul style="list-style-type: none"> •Card Number OA 512 value is always 1. O S/R values are always 1 to 2.
Port	Physical	<ul style="list-style-type: none"> •Port Number VSD (digital) range is 1 to 2, or 1 to 4. VSB (digital; Euro ISDN BRI) range is 1 to 2 (OA 512), or 1 to 4 (OSR). VSA (analog) range is 1 to 2 (FXO), 1 to 4 (FXO/FXO or FXS), 1 to 6 (FXS and FXO), or 1 to 8 (FXS/FXS modules); applies only to OA 512. Same for O S/R except range is 1 to 16 based upon module installed. See Chapter 2, "VoIP Daughtercards," for more information on port numbering for VoIP.
Channel	Logical	<ul style="list-style-type: none"> •Channel Number VSD T1 range is 1 to 24. VSD E1 range is 1 to 30. VSB (Euro ISDN BRI) is 1 to 2. VSA (analog) value is always 1.
StartChannel-EndChannel	Logical	<ul style="list-style-type: none"> •Range of Channels Channel-to-Channel, e.g., 11-16. A channel number is assigned to every channel in the specified range; in this case, Channels 11, 12, 13, 14, 15 and 16.

◆ Note ◆

In the current release you will not receive an error message if you enter an invalid slot number, port number, or channel number for **view** commands.

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Voice Switching Daughtercard Commands

The commands listed and described below are used to configure and activate voice switching daughtercards, and include the following daughtercard functions: dialing timers, gateway mode, and voice ports.

Dialing timers determine how the DSPs on the daughtercard detect digits, and the ports provide an interface from the switch to the PSTN via a PBX or key system. The H.323 gateway enables or disables communications between the switch, the PSTN, and the PBX or key system.

Voice Switching Daughtercard (Activate)

- assign IP address mask to voice switching daughtercard
- assign IP address to voice switching daughtercard
- assign default gateway to voice switching daughtercard
- activate voice switching daughtercard
- activate voice switching daughtercard *configuration*
- save current text-based configuration to flash (global “Save All” dump command)

H.323 Gateway Configuration and Runtime Parameters

- outgoing Fast Start mode for gateway (on/off)
- incoming Fast Start mode for gateway (on/off)
- automatic answer for gateway (on/off)

Dialing Timers

- first digit wait duration
- inter digit wait duration
- dial time wait duration
- termination digit (optional)

Analog

- cadence coefficient (North America/Europe)
- ring ID

Euro BRI ISDN

- clock source (external/free running)
- external clock port

Digital Port Configuration

Port Connection Type

voice port digital connection interface type (T1/E1/E1 ISDN PRI/BRI Euro)

Telephony Interface

voice port frame format

voice port circuit identifier (optional)

E1 voice port NFAS (enable/disable)

Line Build Out

voice port line haul (short haul/long haul)

T1 voice port line length

T1 voice port attenuation

E1 voice port cable type

Line Coding

voice port line coding

Facilities Data Link

T1 voice port facilities data link protocol

T1 voice port facilities data link port role (network/user)

Transmit Clock Source

voice port transmit clock source

Loop Back Mode

T1 voice port *loop back configuration*

Signaling Mode

voice port channel signaling mode

Trap Generation

E1 voice port trap generation (enable/disable)

ISDN

E1 ISDN port connection protocol (net/user/qmaster/qslave)

E1 ISDN connection switch type (net3/net5)

ISDN (VSB only)

E1 ISDN BRI Euro line type (point to point/point to multipoint)

ISDN Control and Bearer Channels (VSB only)

E1 ISDN control (Data or "D") channels

E1 ISDN bearer ("B") channels

voice daughter card ip mask

Command Usage

Assign IP address mask to voice switching daughtercard.

Syntax Options

```
voice daughter card <slot/card_number> ip mask <ip_address >
```

Definitions:

slot Specifies the chassis slot number where VSM is installed, (e.g., **2**).

card_number Specifies the voice daughtercard position number, (e.g., **1**).

◆ Syntax Note ◆

The OA 512 value is always 1, and the OS/R values are always 1 to 2.

ip_address Identifies the voice switching daughtercard by IP address mask, (e.g., **255.255.255.0**).

Default:

None

Command Examples:

```
voice daughter card 2/1 ip address mask 255.255.255.0
```

```
voice daughter card 2/2 ip address mask 255.255.255.0
```

Remarks

The IP [subnet] mask address must be the same as the IP subnet mask address used in the routing group for the VoIP gateway(s). See Chapter 4, “Setup and Installation.”

voice daughter card ip address

Command Usage

Assign IP address to voice switching daughtercard.

Syntax Options

```
voice daughter card <slot/card_number> ip address <ip_address >
```

Definitions:

slot Specifies the chassis slot number where VSM is installed, (e.g., **2**).

card_number Specifies the voice daughtercard position number, (e.g., **1**).

◆ Syntax Note ◆

The OA 512 value is always 1, and the OS/R values are always 1 to 2.

ip_address Identifies the voice switching daughtercard by IP address, (e.g., **127.0.0.0**).

Default:

None

Command Examples:

```
voice daughter card 2/1 ip address 127.0.0.0
```

```
voice daughter card 2/2 ip address 127.0.0.0
```

Remarks

This command is required and must be included in the master **vsmboot.asc** file as per Chapter 5, "Setup and Installation."

voice daughter card ip default gateway

Command Usage

Specify the IP address of the default gateway for the voice switching daughtercard.

Syntax Options

```
voice daughter card <slot/card_number> ip default gateway <ip_address >
```

Definitions:

slot Specifies the chassis slot number where VSM is installed, (e.g., **2**).

card_number Specifies the voice daughtercard position number, (e.g., **1**).

◆ Syntax Note ◆

The OA 512 value is always 1, and the OS/R values are always 1 to 2.

ip_address Identifies the IP address of the default gateway, (e.g., **127.0.0.1**).

Default:

None

Command Examples:

```
voice daughter card 2/1 ip default gateway 127.0.0.1
```

```
voice daughter card 2/2 ip default gateway 127.0.0.1
```

Remarks

The IP address of the default gateway must be the same as the IP broadcast address used in the routing group for the VoIP gateway(s). See Chapter 4, “Setup and Installation.”

voice daughter card activate

Command Usage

Activate previously configured voice switching daughtercard parameters.

Syntax Options

voice daughter card <slot/card > activate

Definitions:

slot Specifies slot number of switching module installed in chassis, (e.g., **2**).

card_number Specifies the voice daughtercard position number, (e.g., **1**).

◆ Syntax Note ◆

This command is always required, and must only be issued after all other relevant VoIP CLI commands have been configured. See Chapter 5, “Setup and Installation,” for details regarding its use in the **vsmboot.asc** file.

If using this command on the CLI, all channels should be placed out of service via the **voice channel state** command. (*Not available this release.*)

It is recommended that this command be used only during “off hours,” or after connected devices are first instructed to stop routing calls to the card.

Default:

None

Command Examples:

voice daughter card 2/1 activate
voice daughter card 2/2 activate

Remarks

This command is required and must be included in the master **vsmboot.asc** file as per Chapter 5, “Setup and Installation.”

This command activates all previously configured voice switching daughtercard parameters by transferring cached parameters to the daughtercard, and then placing the parameters into service from the card.

Using this command immediately puts all channels on the voice switching daughtercard out of service, shuts down the card, and automatically erases the previous configuration; this includes disconnecting any calls in progress, and ignoring any incoming traffic from either the phone or data networks.

Once the activation command is issued, the configuration should be saved to the flash directory on the switch via the global **dump** command. The **dump** command should only be used *after* a configured voice switching daughtercard has been activated, otherwise an invalid configuration might be saved instead which could prevent the switch from booting properly the next time. For more details on using this command, see Chapter 4, “Setup and Installation.”

The **dump** command can be used to create a text file in the flash directory of the switch. To view the contents of the generated text file, use the **view file** command. For more information, refer to the **dump** and **view file** command descriptions in the *Command Line Reference Guide*.

voice dump

Command Usage

Save current text-based configuration to flash (global “dump” save all command). The aggregate configuration is first captured and then saved to a single text file that can be viewed, edited, or reapplied to additional switches for implementation.

Syntax Options

```
dump {all | feature-type } [file name ]
```

Definitions:

all Specifies that information for all supported switch features (including VoIP) will be saved to the dump file.

feature-type Specifies that only a particular feature will be saved to the dump file, (e.g., **voice**, **vlan**).

name Specifies a user-defined name for the resulting dump file (18 characters maximum); (e.g., **snapshot1**, **vsmboot.asc**).

Default:

None

Command Examples:

```
dump all
```

```
dump voice
```

```
dump all file snapshot.1
```

```
dump voice file vsmboot.asc
```

Remarks

The **dump** command automatically creates the text file in the flash directory of the switch. To view the contents of the generated text file, you can use the **view file** command. For more information, refer to the **view file** command description in the *Command Line Reference Guide*.

Aside from basic system parameters, (e.g., system name), the **dump** command captures only *non-default* switch parameters for the specified switch features. For example, the default settings for VoIP will not be included in a dump file.

The text file can be edited using a standard text editor.

Screen Output

To view all voice daughtercard parameters, type **dump voice file** followed by valid vsm boot file name, e.g., **dump voice file vsmboot.asc**, and then press **<Enter>**.

A screen similar to the following displays.

```
*****
***          Snap All          ***
*****
!
voice echo on
!
voice daughtercard 4/1 ip mask 255.255.255.0
!
voice daughtercard 4/1 ip address 127.0.0.0
!
!voice port 4/1 interface type T1
!
voice coding profile cp1
!
voice coding profile cp1 type pcm mulaw
!
voice channel 4/1/1 mode telephony
!
voice channel 4/1/2 mode telephony
!
voice channel 4/1/3 mode telephony
!
voice channel 4/1/4 mode telephony
!
voice channel 4/1/5 mode telephony
!
voice channel 4/1/6 mode telephony
!
voice channel 4/1/7 mode telephony
!
voice channel 4/1/8 mode telephony
!
voice channel 4/1/9 mode telephony
!
voice channel 4/1/10 mode telephony
!
voice channel 4/1/11 mode telephony
!
voice channel 4/1/12 mode telephony
!
voice destination VSD_1 h.323 address 195.167.10.33 1720
!
voice destination VSD_2 h.323 address 195.167.10.34 1720
!
voice destination to VSD_1 port 1 local channel VSD_1/1/1-24
!
voice destination to VSD_2 port 1 local channel VSD_1/1/1-24
!
voice phone group Ext. of PBX__1
!
voice phone group Ext. of PBX__2
!
```

```
voice phone group Ext. of PBX__1 type local extensions
!
voice phone group Ext. of PBX__2 type local extensions
!
voice phone group Ext. of PBX__1 site prefix off
!
voice phone group Ext. of PBX__2 site prefix off
!
voice phone group Ext. of PBX__1 format "xxxx"
!
voice phone group Ext of PBX_2 format "xxxx"
!
voice phone group Ext. of PBX__1 strip digit length 0
!
voice phone group Ext. of PBX__2 strip digit length 0
!
voice numbering plan to PBX_1
!
voice numbering plan to PBX_2
!
voice numbering plan to PBX_1 hunt method round robin
!
voice numbering plan to PBX_1 hunt method round robin
!
voice numbering plan to PBX_1 associate destination member to VSD_1
!
voice numbering plan to PBX_2 associate destination member to VSD_2
!
voice numbering plan to PBX_1 associate phone group member Ext. of PBX_1
!
voice numbering plan to PBX_2 associate phone group member Ext. of PBX_2
!
voice numbering plan to PBX_1 description trunk to route calls from VSD1 to PBX1
!
voice numbering plan to PBX_2 description trunk to route calls from VSD2 to PBX2
!
voice numbering plan to PBX_1
!
voice daughter card 4/1 activate
!
```

voice daughter card h.323 out fast start

Command Usage

Set *outgoing* H.323 Fast Start mode for gateway (on/off).

Syntax Options

```
voice daughter card <slot/card_number> h.323 out[going] fast start {on | off}
```

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
going	Optional command syntax. Type either out or outgoing in the command line.
on	Turns ON H.323 outgoing fast start for specified voice network.
off	Turns OFF H.323 outgoing fast start for specified voice network.

Default:

The default setting is **on**.

Command Examples:

```
voice daughter card 2/1 h.323 outgoing fast start off
voice daughter card 2/2 h.323 out fast start on
```

Remarks

Fast Start commands take effect immediately and do not require use of the **voice daughter card activate** command.

This command selects H.323 Fast Start mode on the outgoing side of the link. Fast Start mode reduces the number of messages exchanged between endpoints.

H.323 Fast Start calls connect after a single round-trip message. The Fast Start information is attached to H.225 messages from general setup. H.225 setup messages contain information about voice channels proposed by the originator of the call. The terminating endpoint accepts one of the proposed channels, and informs the originator through the connect message. The connected endpoints then establish logical channels and switch to voice mode. Ringback is sent inband and the voice path exists when the remote endpoint picks up the phone. If the terminating endpoint picks up the phone before the voice channels are established, the originator receives the voice signal directly with no preceding ringback.

If Fast Start calls do not connect in the voice switching daughtercard, the switch automatically reverts the call to the general H.323 setup; this prevents Fast Start calls from being dropped.

H.225, which provides the call setup and control signaling needed to connect h.323 endpoints, is part of the H.323 signaling protocol stack. Q.931 is a similar protocol used over ISDN lines to set up, maintain and terminate calls between H.323 endpoints or agents.

voice daughter card h.323 in fast start

Command Usage

Set *incoming* H.323 Fast Start mode for gateway (on/off).

Syntax Options

```
voice daughter card <slot/card_number> h.323 in[coming] fast start {on | off}
```

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
coming	Optional command syntax. Type either in or incoming in the command line.
on	Turns ON H.323 incoming fast start for specified voice network.
off	Turns OFF H.323 incoming fast start for specified voice network.

Default:

The default setting is **on**.

Command Examples:

```
voice daughter card 2/1 h.323 incoming fast start off  
voice daughter card 2/1 h.323 in fast start on
```

Remarks

For a brief description of Fast Start mode, see the **voice network h.323 outgoing Fast Start** command.

Fast Start commands take effect immediately and do not require use of the **voice daughter card activate** command.

voice daughter card h.323 auto answer

Command Usage

Set automatic answering for incoming calls on gateway (on/off).

Syntax Options

```
voice daughter card <slot/card_number> h.323 auto[matic] answer {on | off}
```

Definitions:

<i>slot</i>	Specifies slot number of switching module installed in chassis, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
matic	Optional command syntax. Type either auto or automatic in the command line.
on	Turns ON automatic answer for incoming calls on gateway.
off	Turns OFF automatic answer for incoming calls on gateway.

Default:

The default setting is **off**.

Command Examples:

```
voice daughter card 2/1 h.323 automatic answer off
voice daughter card 2/2 h.323 auto answer on
```

Remarks

This command is used to select H.323 automatic call answering mode on the incoming side of the link; if turned ON, the call is connected automatically for instant voice.

voice daughter card first digit wait duration

Command Usage

Specify maximum time for voice daughtercard dialing timers to wait between off-hook state and first dialed tone (digit) to be detected.

Syntax Options

voice daughter card <slot/card_number> first digit wait duration <timer_value >

Definitions:

slot Specifies slot number of switching module installed in chassis, (e.g., **2**).

card_number Specifies the voice daughtercard position number, (e.g., **1**).

timer_value Specifies maximum time for dialing timers to wait *between off-hook/first digit dialed* to be detected in milliseconds from 1 to 4,294,967,295 ms, (e.g., **10000**); (1 ms = 1/1000th of a second), (1,000 ms = 1 second), (10,000 ms = 10 seconds), etc. Refer to conversion table below to quickly determine the proper setting.

◆ Syntax Note ◆

Do not use commas when entering a dialing timer value (for example, **10,000** (10 seconds) will return a syntax error message).

Default:

The default *timer_value* is **10000**.

Command Examples:

voice daughter card 2/1 first digit wait duration 10000
voice daughter card 2/2 first digit wait duration 60000

Remarks

When this timer expires, a “no digits received” error condition occurs and the numbering plan dial attempt fails.

Use this table to quickly determine duration values for dialing timers.

Voice Switching Daughtercard Digit Durations (ms) Conversion Table			
10,000 ms = 10 seconds	15,000 ms = 15 seconds	20,000 ms = 20 seconds	30,000 ms = 30 seconds
60,000 ms = 60 sec. (1 min.)	600,000 ms = 10 min.	900,000 ms = 15 min.	1,200,000 ms = 20 min.
1,800,000 ms = 30 min.	3,600,000 ms = 60 min. (1 hour)	21,600,000 ms = 6 hours	36,000,000 ms = 10 hours
43,200,000 ms = 12 hours	86,400,000 ms = 24 hours (1 day)	604,800,000 ms = 7 days (1 week)	864,000,000 ms = 10 days
1,209,600,000 ms = 14 days (2 weeks)	1,814,400,000 ms = 21 days (3 weeks)	2,419,200,000 ms = 28 days (4 weeks)	2,505,600,000 ms = 29 days'
2,592,000,000 ms = = 30 days (1 month)	2,678,400,000 ms = = 31 days	4,233,600,000 ms = = 49 days	4,294,967,295 ms = 49 days, 17 hours, 2 min., 47 sec. and 295 ms

voice daughter card inter digit wait duration

Command Usage

Specify maximum time for voice daughtercard dialing timers to wait between tones (digits) being dialed.

Syntax Options

```
voice daughter card <slot/card_number> inter digit wait duration <timer_value >
```

Definitions:

<i>slot</i>	Specifies slot number of switching module installed in chassis, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
<i>timer_value</i>	Specifies maximum time for dialing timers to wait <i>between digits being dialed</i> in milliseconds from 1 to 4,294,967,295 ms, (e.g., 5000); (100 ms = 1/10 of a second), (1,000 ms = 1 second), (10,000 ms = 10 seconds), (60,000 ms = 1 minute), (300,000 = 5 minutes), etc. Refer to conversion table (see voice daughter card first digit wait duration command) to quickly determine the proper setting.

◆ Syntax Note ◆

Do not use commas when entering a dialing timer value (for example, **5,000** (5 seconds) will return a syntax error message).

Default:

The default *timer_value* is **5000**.

Command Examples:

```
voice daughter card 2/1 inter digit wait duration 5000
voice daughter card 2/2 inter digit wait duration 60000
```

Remarks

When this timer expires, unless a termination digit is dialed, it assumes the caller is finished dialing digits. The numbering plan in use then attempts a match.

voice daughter card dial time wait duration

Command Usage

Specify maximum time for dialing timers to wait for all tones (digits) to be dialed.

Syntax Options

voice daughter card <slot/card_number> dial time wait duration <timer_value >

Definitions:

<i>slot</i>	Specifies slot number of switching module installed in chassis, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
<i>timer_value</i>	Specifies maximum time for dialing timers to wait <i>for all digits to be dialed</i> , from 1 to 4,294,967,295 ms, (e.g., 30000); (100 ms = 1/10 of a second), (1,000 ms = 1 second), (10,000 ms = 10 seconds), (60,000 ms = 1 minute), (300,000 = 5 minutes), etc. second), (1,000 ms = 1 second), (10,000 ms = 10 seconds), (60,000 ms = 1 minute), (300,000 = 5 minutes), etc. Refer to conversion table (see voice daughtercard first digit wait duration command) to quickly determine the proper setting.

◆ Syntax Note ◆

Do not use commas when entering a dialing timer value (for example, **30,000** (30 seconds) will return a syntax error message).

Default:

The default *timer_value* is **120000**.

Command Examples:

voice daughter card 2/1 dial time wait duration 30000
voice daughter card 2/2 dial time wait duration 60000

Remarks

When this timer expires, unless a termination digit is dialed, it assumes the caller is finished dialing digits. The numbering plan in use then attempts a match.

voice daughter card termination digit

Command Usage

Specify DTMF digit used by voice daughtercard dialing timers to terminate dial process.

Syntax Options

voice daughter card <slot/card_number> [**no**] **termination digit** <character>

Definitions:

<i>slot</i>	Specifies slot number of switching module installed in chassis, (e.g., 2).
<i>card_number</i>	Specifies voice daughtercard position number, (e.g., 1).
no	Restores the digit to the default value # .
<i>value</i>	Specifies one of 16 characters: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, #, * or no termination digit (no commas allowed) that can be used by dialing timers to terminate dial process, (e.g., #). Dial termination digit cannot be part of valid phone number.

◆ Syntax Note ◆

If a termination digit is used to terminate the dial process, only one digit (or single character) can be used for the value.

Do not use **no** if entering a value.

Default:

The default *character* value is **#**.

Command Examples:

```
voice daughter card 2/1 no termination digit
voice daughter card 2/2 termination digit 0
voice daughter card 2/3 termination digit #
```

Remarks

The **voice daughter card termination digit** command is used to determine when the dial process is complete. All other digits are ignored after the termination digit. The numbering plan in use attempts a match before the terminating digit is received. All digits dialed until the termination digit is received are considered valid, and the termination digit is discarded. Use of the termination digit is optional.

This command effects the behavior of all phone groups on the daughtercard.

voice daughter card cadence coefficient

Command Usage

Set cadence coefficient, or continental ring tone (North America/Europe) for all ports on an analog voice switching daughtercard (VSA).

Syntax Options

```
voice daughter card <slot/card_number> vsa cadence coefficient {north america | europe}
```

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
north america	Sets the cadence coefficient to North America.
europe	Sets the cadence coefficient to Europe.

◆ Syntax Note ◆

This command must be set in relation to the **voice daughter card ring ID** command.

Default:

The default setting is **north america**.

Command Examples:

```
voice daughter card 2/1 vsa cadence coefficient north america  
voice daughter card 2/2 vsa cadence coefficient europe
```

Remarks

When this command is used, the P33 and P34 jumpers on the VSA daughtercard must be set accordingly for ring voltage amplitude (P33) and ring frequency (P34). See Chapter 2, “VoIP Daughtercards,” for details.

voice daughter card ring id

Command Usage

Set ring ID for the VSA cadence coefficient specified for all ports on an analog voice switching daughtercard (VSA).

Syntax Options

voice daughter card <slot/card_number> vsa ring id {0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | default}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
0, 1, 2, 3, 4, 5, 6, 7	Specifies variance in continental ring tone for either North America or Europe (e.g., 2) as per selected coefficient. The value 0 is the same as default .
default	Sets the tone variance to the default setting, i.e, 0.

◆ Syntax Note ◆

This command must be set in relation to the **voice daughter card vsa cadence coefficient** command.

Default:

The default setting is **0**.

Command Examples:

```
voice daughter card 2/1 vsa ring id default
voice daughter card 2/2 vsa ring id 0
voice daughter card 2/3 vsa ring id 7
```

voice daughter card vsb clock source

Command Usage

Set clock source for the BRI interface for each port on the VSB daughtercard.

Syntax Options

```
voice daughter card <slot/card_number> vsb clock source { external | free running }
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
external	Sets BRI interface to use an external clock source.
free running	Sets BRI interface to use the free running (internal) clock.

◆ Syntax Notes ◆

To use this command, the port connection type must be set to **bri euro** via the **voice port interface type** command.

This command must be set in relation to the **voice daughtercard vsb external clock port** command.

When this command is entered, whether at the command line prompt or in the **vsmboot.asc** file, the **voice daughter card vsb clock source** command line must precede the **voice daughter card vsb clock port** command line.

Default:

The default setting is **external**.

Command Example:

```
voice daughter card 2/1 vsb clock source external  
voice daughter card 2/2 vsb clock source free running
```

voice daughter card vsb external clock port

Command Usage

Specifies which port on the VSB daughtercard to use for the master clock signal.

Syntax Options

voice daughter card <slot/card_number> vsb external clock port <value >

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
<i>value</i>	Specifies physical port number on voice daughtercard, (e.g., 1).

◆ Syntax Notes ◆

To use this command, the port connection type must be set to **bri euro** via the **voice port interface type** command.

To use this command, the vsb clock source must be set to **external** via the **voice daughter card vsb clock source** command.

When this command is used, the **voice daughter card vsb clock source** command line in the **vsmboot.asc** file must precede the **voice daughter card vsb clock port** command line.

Default:

The default is **1**.

Command Example:

```
voice daughter card 2/1 vsb external clock port 1
voice daughter card 2/2 vsb external clock port 2
```

voice port interface type

Command Usage

Specify voice daughtercard port *digital* connection interface type (does not include analog interface). This setting determines the number of channels per port.

Syntax Options

```
voice port <slot/port > interface type { t1 | e1 | e1 isdn pri | bri euro }
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
t1	Specifies T1 as the voice daughtercard port connection interface type.
e1	Specifies E1 (QSIG) as the voice daughtercard port connection interface type.
e1 isdn pri	Specifies PRI E1 (ISDN) as the voice daughtercard port connection interface type.
bri euro	Specifies BRI Euro (Euro ISDN) as the voice daughtercard port digital connection interface type. This setting applies only to the VSB.

◆ Syntax Notes ◆

If PRI E1 is specified, this command sets the ISDN DS-1 type to PRI E1 (also referred to as ISDN).

If T1 is specified, Mu-law companding must be selected via the **voice signaling companding** command. Likewise, if E1 (either ISDN PRI or BRI) is specified, A-law companding must be selected.

When a digital voice port interface type is specified, a corresponding protocol type must also be specified via the **voice signaling protocol** command.

Default:

For *VSD only*, the default setting is **t1**. For *VSB only*, the default setting is **bri euro**.

Command Examples:

```
voice port 2/1 interface type t1
voice port 2/2 interface type e1
voice port 2/3 interface type e1 isdn pri
voice port 2/4 interface type bri euro
```

Remarks

The **voice port interface type** command is used to determine the number of channels allowed per physical port on the voice daughtercard; the interfaces as described below are only supported on the digital versions of this card. (The analog voice switching daughtercard (VSA) does not support any of these interfaces).

The T1 interface uses a maximum of 24 DS-0 (64 Kpbs) channels, has a capacity of 1.544 Mbps, and follows Mu-law companding, which is used in North America (United States and Canada), and Japan.

The E1 (QSIG/ISDN signaling) interface uses a maximum of 32 DS-0 (64 Kbps) channels, has a capacity of 2.048 Mbps, and follows A-law companding which is used in Europe. This interface is a European CEPT (Conference of European Postal and Telecommunications Administrations) carrier. Thirty channels can be used for voice. E1 uses timeslots rather than robbed-bit signaling. Timeslot 0 is for framing (synchronization) and timeslot 16 is for signaling (either Common Channel Signaling (CCS), i.e., (QSIG/ISDN), or Channel Associated Signaling (CAS)/R2, handshaking on multi-frequency signaling digit). Timeslot 1-15 and 16-31 are for voice channels.

The E1 ISDN PRI interface uses a maximum of 32 DS-0 (64 Kbps) channels, and has a capacity of 2.048 Mbps. The interface follows A-law companding and is a European CEPT carrier. Two channels are D (data) channels. Channel 16 is reserved as a control channel. Channel 0 is reserved for framing. Remaining channels are B (bearer) channels.

The BRI E1 (Euro ISDN) interface uses a maximum of 3 DS-0 (64 Kbps) channels, and has a capacity of 2.048 Mbps. The interface follows A-law companding and is a European CEPT carrier. Two voice channels are B (B1 and B2 bearer) channels supported at 64 Kbps; another channel supports data at 16 Kbps. Channel 0 is reserved as a control channel. This interface is only supported on the BRI voice switching digital daughtercard (VSB).

◆ **Note** ◆

When used separately, the terms E1 and ETSI both entail European PRI and BRI interfaces. E1 ETSI used together as one term refers specifically to Euro PRI.

voice port frame format

Command Usage

Specifies the frame format of the voice port.

Syntax Options

voice port <*slot/port*> **frame format** {**none** | **superframe** | **extended superframe** | **e1** | **e1 crc** | **e1 mf**}

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
none	Specifies unframed voice port frame format.
superframe	Specifies superframe voice port frame format (also known as AT&T D4 format DS-1).
extended superframe	Specifies extended superframe (ESF) voice port frame format (DS-1 standard used for Wide Area Networks).
e1	Specifies CCITT (ITU Geneva) Recommendation G.704 (ITU-T Recommendation for synchronous frame structures used at primary and secondary levels; double frame (FAS / Pulse Code Modulation (PCM) 31) voice port frame format. CCITT stands for Consultive Committee on International Telegraphy and Telephony.
e1 crc	Specifies Cyclic Redundancy Check for transmission; CCITT Recommendation G.704; CRC4 multiframe (FAS/PCM 31).

◆ Syntax Notes ◆

A frame format **none** means the frame is unframed.

If the voice daughtercard connection port type is set to T1, then only superframe, extended superframe and [no] are allowed. If the voice daughtercard connection port type is set to E1, E1 ISDN PRI, or BRI Euro, then only E1, E1 CRC, E1 MF, E1 CRC MF, and [no] are allowed. The port connection type can be set via the **voice port interface type** command.

A no frame format superframe, or no frame format extended superframe, or no frame format E1, or no frame format E1 CRC, or no frame format E1 MF, or no frame format E1 CRC MF means the frame is unframed; in which case, superframe, extended superframe, E1, E1 CRC, E1 MF, E1 CRC MF are ignored.

Default:

The default frame format setting is **none**.

Command Examples:

```
voice port 2/1 frame format none
voice port 2/2 frame format superframe
voice port 2/3 frame format extended superframe
```

Remarks

The **voice port frame format** command is used to indicate the type of DSL line implementing the circuit. The circuit affects the number of bits per second that the circuit can reasonably carry, as well as the interpretation of the usage and error statistics.

The time slot divisions, which are the basis for T1 and E1 circuit connections, e.g., multiplexed Digital Service (DS-1), are determined as follows for frames, superframes, extended superframes and multiframes.

T1 Framing

A T1 frame consists of 24, 8-bit time slots and a 1 bit-synchronization and control bit. Twelve (12) T1 frames can be grouped into a superframe (SF/D4), or 24 T1 frames can be grouped into an extended superframe. In each superframe, the 6th and 12th frame may contain “robbed bit” (A,B) signaling, which means the least significant bit is robbed from each time in the 6th and 12th frame and used for signaling. In extended superframes, this robbed bit signaling (A, B, C, D) occurs in the 6th, 12th, 18th, and 24th frames.

E1 Framing

The E1 frame consists of 32, 8-bit time slots (two of these slots are used for synchronization and multiframe signaling) for 256 bits per frame at 2.048 megabits per second. Sixteen (16) E1 frames are grouped into multiframe. An E1 multiframe can use Channel Associated Signaling (CAS) contained in time slot 16. Timeslot 16 in multiframe 0 is used for multiframe synchronization and control. Timeslot 16 of multiframes 1 through 15 are used to carry A, B, C, and D signaling bits.

voice port circuit identifier

Command Usage

Define voice port circuit identifier.

Syntax Options

```
voice port <slot/port > circuit identifier {text_string }
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>text_string</i>	Identifies vendor transmission circuit for troubleshooting, (e.g., 38.ivbd.005719.001.pt); can be a maximum of 30 characters. The following characters are permitted in the text string: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & / \ < > () [] { }

◆ Syntax Note ◆

Circuit IDs tend to use this baseline format: "xx.xxx.xxxxxx.xxx"

At least one ASCII character must be used in the text string, and quotes must be located at each end of the circuit identifier.

Default:

None

Command Examples:

```
voice port 2/1 circuit identifier "38.ivbd.oo5719.001.pt"
```

```
voice port 2/2 circuit identifier "39.hqac.001727.000"
```

```
voice port 2/3 circuit identifier "37.hsgc.305001.508"
```

Remarks

The **voice port circuit identifier** command is used to identify transmission circuits (trunks) for troubleshooting by a telephone company. The circuit identifier is obtained from the telephone company who may need or require this identifying information in order to fix line transmission problems. Identification of the voice port circuit identifier is optional.

voice port nfas framing

Command Usage

Set E1 voice port NFAS (Non-Facility Associated Framing); (enable/disable).

Syntax Options

```
voice port <slot/port > nfas framing {enable | disable}
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
enable	Turns ON NFAS framing on E1 voice port.
disable	Turns OFF NFAS framing on E1 voice port.

Default:

The default setting is **disable**.

Command Examples:

```
voice port 2/1 nfas framing disable
voice port 2/1 nfas framing enable
```

Remarks

The **voice port nfas framing** command determines whether NFAS framing, e.g, out-of-band signaling, will be used with E1 or PRI E1.

When NFAS framing is *enabled*, the framing is based on bit 2 of Time Slot 0 NOT-NFAS.

NFAS framing can only be enabled if the voice daughtercard connection port type is set to E1, E1 ISDN PRI, or BRI Euro. The port connection type can be set via the **voice port interface type** command.

◆ Note ◆

This command is not supported on VSA modules.

voice port line build out

Command Usage

Set voice port line haul (short haul/long haul).

Syntax Options

```
voice port <slot/port > line build out { short [haul] | long [haul]}
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
short	Specifies short haul line build out.
haul	Optional command syntax. Type either short or short haul in the command line.
long	Specifies long haul line build out.
haul	Optional command syntax. Type either long or long haul in the command line.

Default:

The default setting is **short haul**.

Command Examples:

```
voice port 2/1 line build out short haul  
voice port 2/2 line build out long haul  
voice port 2/3 line build out short  
voice port 2/4 line build out long
```

Remarks

Indicates line build out of this port. Only T1/E1 ports with Line Interface Unit (LIU) equipped can support Long Haul (LH); otherwise, only Short Haul (SH) is supported.

To use this command, the voice daughtercard connection port type must be set to T1, E1 or EI ISDN PRI, or BRI Euro via the **voice port interface type** command.

voice port line length

Command Usage

Specify T1 voice port line length.

Syntax Options

```
voice port <slot/port > [line build out] line length <value >
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
line build out	Optional command syntax.
<i>value</i>	Specifies T1 voice port line length in ranges from 0 to 200 meters, (e.g., 30).

Default:

The default line length *value* is **30**.

Command Examples:

```
voice port 2/1 line build out line length 30
voice port 2/2 line length 30
```

Remarks

To use this command, the voice daughtercard connection type must be set to T1 via the **voice port interface type** command.

To use this command, the voice daughtercard connection type must be set to **short haul** via the **voice port line build out** command.

voice port attenuation

Command Usage

Specify T1 voice port attenuation.

Syntax Options

```
voice port <slot/port > [line build out] attenuation {0 | -7.5 | -15.0 | -22.5}
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
line build out	Optional command syntax.
0	Specifies 0 decibels. The decibels value indicates attenuation (i.e., the allowed decrease in power signal).
-7.5	Specifies -7.5 decibels. The decibels value indicates attenuation (i.e., the allowed decrease in power signal).
-15.5	Specifies -15.5 decibels. The decibels value indicates attenuation (i.e., the allowed decrease in power signal).
-22.5	Specifies -22.5 decibels. The decibels value indicates attenuation (i.e., the allowed decrease in power signal).

Default:

The default attenuation setting is **0** decibels.

Command Examples:

```
voice port 2/1 line build out attenuation 0
voice port 2/2 line build out attenuation -7.5
voice port 2/3 line build out attenuation -15.0
voice port 2/4 line build out attenuation - 22.5
voice port 3/1 attenuation 0
voice port 3/2 attenuation -7.5
voice port 3/3 attenuation -15.0
voice port 3/4 attenuation -22.5
```

Remarks

To use this command, the voice daughtercard connection type must be set to T1 via the **voice port interface type** command.

To use this command, the line build out must be set to long haul via the **voice port line build out** command.

voice port cable type

Command Usage

Specify E1, E1 ISDN PRI, or BRI Euro voice port cable type.

Syntax Options

```
voice port <slot/port > [line build out] cable type {75 | 120}
```

Definitions:

slot Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).

port Specifies physical port number on voice daughtercard, (e.g., **1**).

line build out Optional command syntax.

75 Specifies 75 Ohms for type of cable connected to port.

120 Specifies 120 Ohms for type of cable connected to port.

Default:

The default cable type setting is **120** Ohms.

Command Examples:

```
voice port 2/1 line build out cable type 120
```

```
voice port 2/2 line build out cable type 75
```

```
voice port 2/1 cable type 120
```

```
voice port 2/2 cable type 75
```

Remarks

To use this command, the voice daughtercard connection type must be set to E1, E1 ISDN PRI, or BRI Euro via the **voice port interface type** command.

voice port line coding

Command Usage

Specify line coding of voice port.

Syntax Options

```
voice port <slot/port > [line build out] line coding {ami | b8zs | hdb3}
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
line build out	Optional command syntax.
ami	Specifies Alternative Mark Inversion line coding for the voice port.
b8zs	Specifies Bipolar 8 Zero substitution line coding for the voice port.
hdb3	Specifies High density Bipolar with 3 zero substitution line coding for the voice port.

Default:

The default line coding type is **b8zs for T1 or AMI for E1**.

Command Examples:

```
voice port 2/1 line build out line coding ami
voice port 2/2 line build out line coding b8zs
voice port 2/3 line build out line coding hdb3
voice port 2/4 line coding ami
voice port 3/1 line coding b8zs
voice port 3/2 line coding hdb3
```

Remarks

AMI line coding is supported when the voice daughtercard connection port interface type is set to T1, E1, E1 ISDN PRI, or BRI Euro via the **voice port interface type** command.

The term dsx1AMI refers to a mode wherein no zero code suppression is present is used on the link because line encoding does not solve the problem directly. In this application, the higher layer must provide data which meets or exceeds the requirements, such as inverting High Level Data Link Control (HDLC) data. E1 links, with or without a Cyclic Redundancy Check (CRC), use dsx1AMI line coding.

B8Zs line coding is supported when the voice daughtercard connection port type is T1. The term dsx1B8ZS refers to the use of a specified pattern of normal bits and bipolar violations which are used to replace a sequence of 8 zero bits.

HDB3 line coding is supported when the voice daughtercard connection port type is E1, ISDN PRI E1, or BRI Euro.

voice port facilities data link protocol

Command Usage

Specify T1 voice port facilities data link protocol.

Syntax Options

```
voice port <slot/port > facilities data link protocol {none | ansi t1.403 | at&t 54016 | t1.403 at&t}
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
none	Indicates device does not use the facilities data link.
ansi t1.403	Indicates device uses the dsx1ANSI-T1-403 facilities data link exchange recommended by ANSI.
at&t 54016	Indicates device uses the dsx1ATT-54016 ESF (Extended Super Frame) facilities data link exchange.
t1.403 at&t	Indicates device uses the ANSI t1.403 for ESF (Extended Super Frame) facilities data link exchange.

◆ Syntax Notes ◆

A “no facilities data link protocol” means that the frame is unframed.

A “no facilities data link protocol ANSI T1.403,” or “no facilities data link protocol AT&T 54016,” or “no facilities data link protocol T.1403 AT&T,” means that the device does not use the facilities data link protocol. The “ANSI T1.403 and AT&T 54106 and T1.403 AT&T are ignored.

Default:

The default setting is **none**.

Command Examples:

```
voice port 2/1 facilities data link protocol none
voice port 2/2 facilities data link protocol ansi t1.403
voice port 2/3 facilities data link protocol at&t 54016
voice port 2/4 facilities data link protocol t1.403 at&t
```

Remarks

To use this command, the voice daughtercard connection interface type must be set to T1 via the **voice port interface type** command.

voice port facilities data link port role

Command Usage

Set T1 voice port facilities data link port role (network/user).

Syntax Options

```
voice port <slot/port > facilities data link port role {network | user}
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
network	Indicates facilities data link port is controlled by the network, i.e., the VSD.
user	Indicates facilities data link port is controlled by the user, e.g., the telephone company.

◆ Syntax Note ◆

If the port role is network and the **fdlMode** is set to AT&T 54016 via the **voice port facilities data link protocol** command, then this port periodically sends AT&T performance requests to customer interface.

Default:

The default setting is **user**.

Command Examples:

```
voice port 2/1 facilities data link port role user  
voice port 2/2 facilities data link port role network
```

Remarks

To use this command, the voice daughtercard connection interface type must be set to T1 via the **voice port interface type** command.

voice port transmit clock source**Command Usage**

Specify voice port transmit clock source.

Syntax Options

```
voice port <slot/port > transmit clock source {loop timing | local timing}
```

Definitions:

slot Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).

port Specifies physical port number on voice daughtercard, (e.g., **1**).

loop timing Indicates that the recovered receive clock is being used as the transmit clock.

local timing Indicates that a local clock source is being used as the transmit clock.

Default:

The default setting is **local timing**.

Command Examples:

```
voice port 2/1 transmit clock source local timing
```

```
voice port 2/2 transmit clock source loop timing
```

voice port loop back mode

Command Usage

Specify T1 voice port loop back mode configuration.

Syntax Options

voice port <slot/port > loop back mode {none | payload | line | inward}

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
none	Indicates T1 interface sends looped or normal data for loopback (i.e., not in the loopback state; a device incapable of performing a loopback on the interface always returns “none” as its value. <i>Also known as Dsx1NoLoop.</i>
payload	Indicates T1 interface sends request for a payload loopback (i.e., received signal at this interface is looped through the device; typically, the received signal is looped back for retransmission after it has passed through the device’s framing function). <i>Also known as Dsx1PayloadLoop.</i>
line	Indicates T1 interface sends request for a line loopback (i.e., received signal at this interface does not go through the device (minimum penetration) but is looped back out). <i>Also known as Dsx1LineLoop.</i>
inward	Indicates undefined T1 interface loopback request. <i>Also known as Dsx1OtherLoop.</i>

Default:

The default setting is **payload**.

Command Examples:

```
voice port 2/1 loop back mode none
voice port 2/2 loop back mode payload
voice port 2/3 loop back mode line
voice port 2/4 loop back mode inward
```

Remarks

This variable represents the loop back configuration of the T1 (DS-1) interface, and indicates what type of code is being sent across the T1 interface by the device. A bad value is returned in response to a requested loop back state that the interface providing read/write access does not support.

voice port signaling mode

Command Usage

Specify voice port channel signaling mode.

Syntax Options

```
voice port <slot/port > signaling mode {none | cas | ccs}
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
none	Indicates that no bits are reserved for signaling on this channel.
cas	Indicates that T1 Channel Associated Signaling (CAS) is in use; (applies only to VSD T1).
ccs	Indicates that Common Channel Signaling (CCS) is in use on channel 16 of an E1 link (applies to VSD E1, VSD E1 ISDN PRI), and to channel 3 (VSB BRI Euro voice daughtercard).

Default:

Refer to definitions above.

Command Examples:

```
voice port 2/1 signaling mode none
voice port 2/2 signaling mode CAS
voice port 2/3 signaling mode CCS
```

Remarks

A “no signal mode” means that no bits are reserved for signaling on this channel.

A “no signal mode CAS” or “no signal mode CCS” means that no bits are reserved for signaling on this channel. The CAS and CCS are ignored.

voice port trap generation

Command Usage

Set voice port trap generation (enable/disable). Indicates if line status change trap is sent to the network management system (NMS). (*Not available this release*).

Syntax Options

```
voice port <slot/port > trap generation {enable | disable}
```

Definitions:

slot Specifies slot number of voice switching daughtercard installed in switching module, (e.g., **2**).

port Specifies physical port number on voice daughtercard, (e.g., **1**).

enable Turns ON trap generation on voice port.

disable Turns OFF trap generation on voice port.

Default:

The default setting is **disable**.

Command Examples:

```
voice port 2/1 trap generation disable
```

```
voice port 2/2 trap generation enable
```

Remarks

To use this command, the voice daughtercard connection type must be set to T1, E1, E1 ISDN PRI, or BRI Euro via the **voice port interface type** command.

voice port isdn protocol

Command Usage

Specify ISDN connection protocol for ISDN (BRI Euro) ports.

Syntax Options

```
voice port <slot/port> isdn protocol {net | user | qmaster | qslave}
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
net	Sets the ISDN connection protocol to network mode for PRI E1 (European Telecommunications Standards Institute (ETSI) standard protocol). <i>For VSB only:</i> net configures the port as NT (Network Terminator) /LT (Line Terminator).
user	Sets the ISDN connection protocol to user mode for PRI E1 (ETSI standard protocol). <i>For VSB only:</i> user configures the port as TE (Terminal Endpoint).
qmaster	Sets the QSIG ISDN connection protocol to QSIG standard protocol for PRI E1; QSIG <i>master</i> (ETSI std. protocol).
qslave	Sets the ISDN connection protocol to QSIG standard protocol for PRI E1; QSIG <i>slave</i> (ETSI std. protocol).

Default:

The default setting is **qmaster**.

Command Examples:

```
voice port 2/1 isdn protocol qmaster
voice port 2/2 isdn protocol net
voice port 2/3 isdn protocol user
voice port 2/4 isdn protocol qslave
```

Remarks

The **voice port isdn protocol** command is used to configure the channel protocol mode for the ISDN ports at the voice daughtercard level. Specification of the protocol is optional.

To use this command, the voice daughtercard digital connection port type must be set to BRI Euro, or E1, or E1 ISDN PRI via the **voice port interface type** command.

When this command is used, the **J29** and **J30** jumpers on the VSB daughtercard must be set accordingly for **NT (LT)/TE**, or damage to either the VSB and/or any connected device may occur. See also Chapter 2, "VoIP Daughtercards," for details.

◆ Note ◆

When used separately, the terms E1 and ETSI both entail European PRI and BRI interfaces. E1 ETSI used together as one term refers specifically to Euro PRI.

For VSB jumpers **J29** and **J30**, only the following five NT (LT)/TE port configurations are allowed:

- All four ports can be configured as TE (ports are TE, TE, TE, TE).
- All four ports can be configured as NT (Ports are NT, NT, NT, NT).
- Ports 1, 2, and 3 configured as TE and port 4 configured as NT (ports are TE, TE, TE, NT).
- Ports 1 and 3 configured as TE, and port 2 and 4 configured as NT (ports are TE, NT, TE, NT).
- Port 1 configured as TE, and port 2, 3, and 4 configured as NT (ports are TE, NT, NT, NT).

◆ **Warning** ◆

Failure to properly configure the VSB to use one of the five NT (LT)/TE port configurations above will cause unexpected results, and can cause physical damage to the hardware.

voice port isdn switch type

Command Usage

Specify ISDN connection switch type for ISDN ports.

Syntax Options

voice port <slot/port > isdn switch type {net3 | net5}

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
net3	Sets ISDN connection switch type to Euro ISDN (BRI E1 2 B + D channels).
net5	Sets ISDN connection switch type to Euro ISDN (PRI E1 30 B + D channels).

◆ Syntax Notes ◆

If the voice daughtercard connection port type is set to PRI E1, the **voice port isdn connection switch type** command uses the default value of **net5**.

The port connection type must be set via the **voice port interface type** command to indicate the number of channels available on the port, e.g., 2 channels (BRI Euro), 30 channels (E1 PRI ISDN).

Default:

The default setting is **net5 (for VSD), and net3 (for VSB)**.

Command Examples:

```
voice port 2/1 isdn switch type net5
voice port 2/2 isdn switch type net3
```

Remarks

The **voice port isdn connection switch type** command is used to configure the type of switch connection for the ISDN ports at the voice daughtercard level. Specification of the E1 ISDN protocol is optional unless E1 ISDN PRI or BRI Euro is used.

In the “*user mode*,” set through the **voice port isdn protocol** command, the ISDN connection switch type uses the switch type to which the ISDN link is connected.

In the “*network mode*,” set through the **voice port isdn protocol** command, the ISDN connection switch type commands selects the switch type to be emulated.

voice port bri line type

Command Usage

Specify ISDN EURO BRI line type (point to point/point to multipoint). (Point to point *not available this release*.)

Syntax Options

```
voice port <slot/port> isdn bri line type {point to point | point to multipoint}
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
point to point	Sets physical connection wherein a single network termination point supports a single terminal equipment (TE) device. (<i>Not available this release</i> .)
point to multipoint	Sets physical connection wherein a single network termination point supports multipoint terminal equipment (TE) devices.

◆ Syntax Notes ◆

To use this command, the port connection type must be set to **bri euro** via the **voice port interface type** command.

If the voice daughtercard connection port type is set to **bri euro**, this command will automatically be generated with the default value of **point to point**.

To use this command, the isdn connection protocol must be set to **net** via the **voice port isdn protocol** command.

Default:

The default setting is **point to multipoint**

Command Examples:

```
voice port 2/1 bri line type point to multipoint
voice port 2/2 bri line type point to multipoint
```

Remarks

The **voice port isdn connection switch type** command is used to configure the type of switch connection for the ISDN ports at the voice daughtercard level. Specification of the E1 ISDN protocol is optional unless E1 ISDN PRI or BRI Euro is used.

voice channel isdn d channel

Command Usage

Specify control (data or “D”) channels for E1 ISDN PRI or BRI Euro. (*Not available this release.*)

Syntax Options

voice channel <slot/port/channel > isdn d channel

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies the control (D) channel number (i.e., 16 for E1 ISDN PRI and 0 for BRI Euro).

Syntax Notes ♦

The port connection type must also be set via the **voice port interface type** command to indicate the number of channels available on the port, e.g., 3 channels (BRI Euro), 32 channels (E1 PRI ISDN).

Only **0** and **16** are allowed for data channel numbers; control channel **16** is reserved for signaling on E1 ISDN PRI, and **0** is reserved for the BRI Euro control channel. For E1 ISDN PRI, channel **0** is used for framing (there is no framing on BRI Euro).

Data and corresponding bearer channels must apply to the same voice switching daughtercard.

Default:

For *E1 ISDN PRI*, the default *channel* value is **16**. For *BRI Euro*, the default *channel* value is **0**.

Command Example:

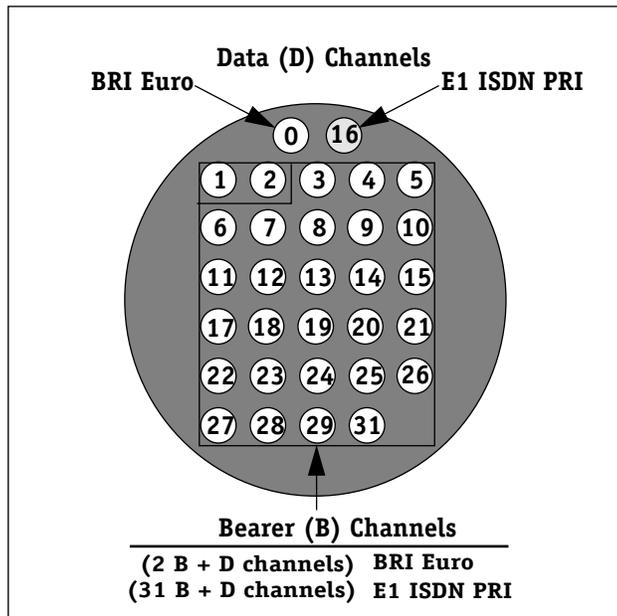
```
voice channel 2/1 16 isdn d channel
voice channel 2/2 16 isdn d channel
voice channel 2/3 16 isdn d channel
voice channel 2/4 16 isdn d channel
voice channel 2/1 0 isdn d channel
voice channel 2/2 0 isdn d channel
voice channel 2/3 0 isdn d channel
voice channel 2/4 0 isdn d channel
```

Remarks

The **voice channel isdn d channel** command is used to configure the type of switch connection for the ISDN ports at the voice daughtercard level. Specification of the ISDN protocol is optional if not using E1 ISDN PRI or BRI Euro.

Use this table and illustration below to determine valid data/bearer channel, slot and port values (applies to OS/R with two voice switching daughtercards).

Interface Type	Data Channels	Bearer Channels	Slot Number	Port Number
E1 ISDN PRI	16	1-30	2-9	1-4
BRI Euro	0	1-2	2-9	1-4



voice channel isdn b channel

Command Usage

Specify E1 ISDN bearer (“B”) channels. (*Not available this release.*)

Syntax Options

```
voice channel <slot/port/channel > isdn b channel <port/dchannel dsl_id >
```

Definitions:

<i>slot</i>	Specifies slot number of voice switching daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies bearer (B) channel number (e.g., 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 17, 18, 19, 20, 21, 22, 23, 24, 25, 26, 27, 28, 29, 31) for E1 ISDN PRI; BRI Euro is limited to two bearer channels.
<i>dsl_id</i>	Specified identification number for related D channel.

◆ Syntax Notes ◆

The port connection type must also be set via the **voice port interface type** command to indicate the number of channels available on the port, e.g., 3 channels (BRI Euro), 32 channels (E1 PRI ISDN).

Only **0** and **16** are allowed for data channel numbers; control channel **16** is reserved for signaling on E1 ISDN PRI, and **0** is reserved for the BRI Euro control channel. For E1 ISDN PRI, channel **0** is used for framing (there is no framing on BRI Euro).

Bearer channels and corresponding Data channels must apply to to the same voice switching daughtercard.

Default:

For *E1 ISDN PRI*, the default *channel* values are **1-15, 17-31**. For *BRI Euro*, the default *channel* values are **1-2**.

Command Example:

```
voice channel 2/1 1 isdn b channel
voice channel 2/2 2 isdn b channel
voice channel 3/1 1 isdn b channel
voice channel 3/2 15 isdn b channel
voice channel 3/3 17 isdn b channel
voice channel 3/4 31 isdn b channel
```

Remarks

The **voice port isdn connection switch type** command is used to configure the type of switch connection for the ISDN ports at the voice daughtercard level. Specification of the ISDN protocol is optional if not using E1 ISDN PRI or BRI Euro.

Use the table and illustration above to determine valid bearer/data channel, slot and port values (applies to OS/R with two voice switching daughtercards).

Channel Properties

The commands listed and described below are used to configure the channel properties for individual voice channels as follows: voice channel mode, PLAR (Private Line Automatic Ring-down), outbound caller ID, and voice channel initialization.

Voice Channel Configuration

voice channel mode

voice channel PLAR dial-in phone number

Channel Operational State

voice channel initialization (in-service/out-of-service)

voice channel mode

Command Usage

Specify voice channel mode—telephony, passthrough or PLAR (Private Line Automatic Ring-down) used to complete calls on designated channels. Establishes semi-fixed bandwidth connections between points in the network.

Syntax Options

voice channel <slot/port/startChannel-endChannel > mode {telephony | pass through | plar}

Definitions:

<i>slot</i>	Specifies slot number of voice daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 2).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

telephony	Allows use of Numbering Plan to complete calls to specified hunted destination.
pass through	Allows two DS-0 channels to be cross-connected to complete calls (<i>not available this release</i>).
plar	Allows calls to be routed based upon telephone number configured to dialing plan.

Default:

The default setting is **telephony**.

Command Examples:

```
voice channel 2/2/1-12 mode telephony
voice channel 2/2/13-24 mode pass through
voice channel 2/2/1-12 mode plar
```

Remarks

In the *telephony mode*, digits come from voice path or signaling. Upon in-seize, the voice switching daughtercard collects inbound digits, then uses the Numbering Plan to complete calls to the specified hunted destination.

In the *passthrough mode*, no digit processing is performed by the DSPs on the voice switching daughtercard, and no routing is performed. Upon in-seize, the channel is immediately connected to another channel, via Numbering Plan processing.

In the *PLAR mode*, no inbound digit processing takes place. Upon in-seize, the PLAR telephone number is immediately routed based upon the dialing plan. The telephone number must be configured to automatically dial using the dial plan.

voice channel dial in private line automatic ringdown

Command Usage

Specify voice channel Private Line Automatic Ringdown (PLAR) dial-in phone number. This command is used to configure the number to be dialed in Switched CAS mode when a specified channel goes off-hook.

Syntax Options

voice channel <slot/port/startChannel-endChannel > dial in private line automatic ringdown <“plar phone number” >

Definitions:

- slot* Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **2**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

- plar phone number* Telephone number configured to dialing plan (e.g., **8188803500**).

◆ **Syntax Notes** ◆

If no number is entered for the PLAR dial-in phone number, operation will be as normal.

This command must be issued before activating the voice daughtercard.

Default:

None

Command Examples:

voice channel 2/2/1-12 dial in private automatic line ringdown 8188803500
voice channel 2/3/13-24 dial in private automatic line ringdown 8188803501

Remarks

To use this command, the channel type (mode) must be set to PLAR (Private Line Automatic Ringdown) via the **voice channel mode** command.

This command is normally used for calls placed from courtesy phones.

voice channel state

Command Usage

Set voice channel initialization (in-service/out-of-service). Signifies the initial state (admin-status) of the channel, or attempt to modify the state of the channel.

Syntax Options

voice channel <slot/port/startChannel-endChannel > state {in service | out of service}

Definitions:

<i>slot</i>	Specifies slot number of voice daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

in service	Puts channel in service as soon as possible after configuration.
out of service	Keeps channel out of service and unused. Channel does not accept hunting or in/out seizures.

Default:

The default setting is **in service**.

Command Examples:

voice channel 2/1/1-12 state in service
voice channel 2/2/13-24 state out of service

Remarks

If the configuration of this channel is incomplete, the command is ignored, the channel is NOT placed into service, and error conditions result.

Telephony Signaling Attributes

The commands listed and described below are used to configure Telephony Signaling attributes including general signaling, Ear & Mouth (E&M), Foreign Exchange Station (FXS), and Foreign Exchange Office (FXO) signaling attributes. This entails call signaling capabilities, outbound caller ID, call progress tones, echo and acoustic echo cancellers, and overrides for call signaling.

Telephony Signaling Channel View

Telephony Signaling Protocol

Signaling Attributes

Dial Out Signaling Tones

time to wait before first tone is sent

duration for a single tone

duration to pause between tones

out dialing port type

Channel Timing

maximum call time length

time to wait for call to be answered

time to wait to force caller to disconnect

time to wait to tear down fax call

Signal Power

companding type (Mu-Law/A-Law)

gain inserted at *receiver*

gain inserted at *transmitter*

amplitude of comfort (idle) noise

E&M Common Signaling

E&M signaling time for *transition to off-hook* (debounce)
E&M signaling time for *transition to on-hook* (debounce)
E&M signaling time to wait before *declaring on-hook* (seize detect)
E&M signaling time to wait before *declaring off-hook* (clear detect)
E&M signaling time to wait before confirming on-hook
E&M signaling time to wait for on-hook after a clear
E&M signaling time to wait between termination and *origination*
E&M signaling time to wait between termination and *receiving*
E&M signaling dial tone generation on incoming calls (on/off)
minimum E&M signaling connection time
time to wait after E&M signaling hang up before disabling

E&M Wink Start Signaling

minimum E&M wink delay on incoming calls
maximum E&M wink delay on incoming calls
duration of E&M wink on incoming calls
time to ignore tones after E&M wink
time to wait for E&M wink on outgoing calls
minimum E&M wink duration
maximum E&M wink duration

E&M Immediate Start Signaling

E&M immediate start time to remain off-hook when congested
E&M immediate start time to wait before beginning digit collection

E&M Delay Start Signaling

minimum E&M delay start response to off-hook state
maximum E&M delay start response to off-hook state
time to ignore incoming digits after E&M delay start
E&M delay start signal detection
minimum E&M delay start detection time on “M” lead
maximum E&M delay start detection time on “M” lead
maximum time to wait for E&M delay start detection

Foreign Exchange Station (FXS) Loop Start (LS) Signaling

FXS LS debounce interval to *on-hook* transition
FXS LS debounce interval to *off-hook* transition
FXS LS time to wait before declaring off-hook
FXS LS minimum time to wait before declaring on-hook by *originator*
FXS LS minimum time to wait before declaring on-hook by *answerer*
FXS LS time to wait after supervisory disconnect before declaring on-hook
FXS LS duration of supervisory disconnect
FXS LS to *generate outbound* caller ID (on/off)

Foreign Exchange Office (FXO) Loop Start (LS) signaling

FXO LS incoming ring signal debounce interval
FXO LS debounce interval to on-hook transition
FXO LS *supervisory disconnect* detection signal (enable/disable)
FXO LS *duration of supervisory disconnect* detection signal
FXO LS time before originating calls while receiving calls
FXO LS time between ring *cycles* to detect ringing
FXO LS time between ring *pulses* to detect ringing
FXO LS to *detect inbound* caller ID (on/off)
FXO LS number of rings allowed before answering
FXO LS debounce for loop current detector
FXO LS debounce for battery reversal detector

Foreign Exchange Station (FXS) Ground Start (GS) Signaling

FXS GS time to wait before declaring off-hook
FXS GS debounce interval for on-hook transition
minimum FXS GS time to wait before declaring on-hook by *originator*
minimum FXS GS time to wait before declaring on-hook by *answerer*
FXS GS time to wait after ring ground before grounding tip
maximum FXS GS time to wait for loop to close after grounding tip
minimum FXS GS start time between open loop and idle state
FXS GS to *generate outbound* caller ID
FXS GS debounce interval for off-hook
FXS GS debounce interval for ring ground detector

Foreign Exchange Station (FXO) Ground Start (GS) Signaling

FXO GS debounce interval for loop open detection
maximum FXO GS time between ring ground and tip ground
FXO GS debounce interval for tip ground detector
FXO GS debounce for incoming ring signal
FXO GS time between consecutive ring *cycles*
FXO GS time between consecutive ring *pulses*
FXO GS to *detect inbound* caller ID (on/off)
FXO GS number of rings allowed before answering
FXO GS debounce interval for loop current detector
FXO GS debounce interval for battery reversal detector

Outbound Caller ID

outbound caller ID name (private/unavailable) to transmit
outbound caller ID number (published/non-published) to transmit

Tones

outbound tone table (ringing/silence)
call progress tone detection (on/off/relative)
call progress tone detection configuration (default/alternate)
V.18 tone detection *threshold hang time*
V.18 tone detection *threshold level*
V.18 *single tone* detection threshold level
V.18 *single tone* detection threshold time

Echo Canceller

echo canceller non-linear sensitivity

Acoustic Echo Canceller

acoustic echo canceller mode (on/off)
acoustic echo canceller non-linear processor (on/off)
acoustic echo canceller output (on/off)
acoustic echo canceller handset (hs) speaker gain
acoustic echo canceller hands free (hf) speaker gain

Override Call Signaling Capabilities

override call signaling for in-band call progress tones (on/off)

override call signaling for full call progress tones (on/off)

override call signaling for ring back (on/off)

override call signaling for in-band codec switching (on/off)

override call signaling for packet switch (PSU) codec switching (on/off)

override call signaling for network overlap dialing (on/off)

override call signaling for information element (IE) transport (on/off)

override call signaling for QSIG information (IE) transport (on/off)

override call signaling for voice, fax, modem, data setup (on/off)

view voice signaling channel

Command Usage

Display Telephony Signaling channel(s).

Syntax Options

```
view voice signaling channel <slot/port/startChannel-endChannel >
```

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

Default:

None

Command Examples:

```
view voice signaling channel 2/1/1-12
view voice signaling channel 2/2/13-12
view voice signaling channel 2/3/31-30
```

Screen Output

To view parameters for a voice signaling channel, type **view voice signaling channel** and a valid voice signaling channel, e.g., **view voice signaling channel 4/1/1**, and then press **<Enter>**.

A screen similar to the following displays.

```
*****
Viewing Signaling Channel
*****
!
voice signaling channel 4/1/1 protocol emi
!
voice signaling channel 4/1/1 companding mulaw
!
voice signaling channel 4/1/1 em dial tone on
!
voice signaling channel 4/1/1 emi glare report 5500
!
voice signaling channel emi digit wait 250
!
```

voice signaling protocol

Command Usage

Specify the protocol to use for Telephony Signaling.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel > **protocol** {**fxs ls** | **fxo ls** | **fxs gs** | **fxo gs** | **emi** | **emd** | **emw** | **isdn**}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard (e.g., 1).
fxs ls	Specifies (CAS) foreign exchange station loop start signaling as the signaling protocol.
fxo ls	Specifies (CAS) foreign exchange office loop start as the signaling protocol.
fxs gs	Specifies (CAS) foreign exchange station ground start as the signaling protocol.
fxo gs	Specifies (CAS) foreign exchange office ground start as the signaling protocol.
emi	Specifies (CAS) E&M (ear & mouth) immediate start as the signaling protocol.
emd	Specifies (CAS) E&M (ear & mouth) delay start as the signaling protocol.
emw	Specifies (CAS) E&M (ear & mouth) wink start as the signaling protocol.
isdn	Specifies (CCS) Integrated Services Digital Network (ISDN) as the signaling protocol.

Default:

The default setting is **fxs ls**.

Command Examples:

```
voice signaling channel 2/1/1-12 protocol fxs ls
voice signaling channel 2/2/13-24 protocol fxo gs
voice signaling channel 2/3/1-30 protocol emw
voice signaling channel 2/4/1-30 protocol isdn
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

To use this command for ISDN, the voice daughtercard connection type must be set to ISDN PRI E1 via the **voice port interface type** command. If use of the ISDN protocol for Telephony Signaling is specified, then all channels on the port must be set to ISDN (E1 ISDN PRI or BRI Euro); the setting must also match the daughtercard connection type set via the **voice port interface type** command.

voice signaling out wait

Command Usage

Specify time to wait before first tone (digit) is sent (dialed out) after going off-hook.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **out wait** <value >

Definitions:

slot Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).

port Specifies physical port number on voice daughtercard (e.g., **1**).

startChannel The first number in the range of voice channels (e.g., **1**).

endChannel The last number in the range of voice channels (e.g., **30**).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value When tone dialing is used, specifies in milliseconds from 0 ms to 20,000, the time to delay after going off-hook before sending the first outbound dial digit (e.g., **20000**).

◆ Syntax Note ◆

Do not use commas when entering the time to delay after going off-hook before sending the first outbound dial digit (for example, **20,000** will return a syntax error message).

Default:

The default value is **400**.

Command Examples:

voice signaling channel 2/1/1-12 out wait 0

voice signaling channel 2/2/13-24 out wait 400

voice signaling channel 2/3/1-30 out wait 20000

voice signaling out tone digit duration

Command Usage

Specify duration for each single tone (digit) dialed out.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> out tone digit duration <value>

Definitions:

slot Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).

port Specifies physical port number on voice daughtercard (e.g., **1**).

startChannel The first number in the range of voice channels (e.g., **1**).

endChannel The last number in the range of voice channels (e.g., **30**).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value When tone dialing is used, specifies in milliseconds from 0 ms to 2000, the duration of each tone, (e.g., **2000**).

◆ **Syntax Note** ◆

Do not use commas when entering the duration of a tone (for example, **2,000** will return a syntax error message).

Default:

The default value is **200**.

Command Examples:

voice signaling channel 2/1/1-12 out tone digit duration 0

voice signaling channel 2/2/13-24 out tone digit duration 200

voice signaling channel 2/3/1-30 out tone digit duration 2000

voice signaling out tone interdigit duration

Command Usage

Specify duration to pause between tones (digits) dialed out.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **out tone interdigit duration** <value >

Definitions:

slot Specifies slot number of voice daughtercard installed in switching module, (e.g., **2**).

port Specifies physical port number on voice daughtercard, (e.g., **1**).

startChannel The first number in the range of voice channels (e.g., **1**).

endChannel The last number in the range of voice channels (e.g., **30**).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value When tone dialing is used, specifies in milliseconds from 0 ms to 2000, the duration of the interdigit gap between tones, (e.g., **200**).

◆ Syntax Note ◆

Do not use commas when entering the duration of the interdigit gap between tones (for example, **2,000** will return a syntax error message).

Default:

The default value is **200**.

Command Examples:

voice signaling channel 2/1/1-12 out tone interdigit duration 0

voice signaling channel 2/2/13-24 out tone interdigit duration 200

voice signaling channel 2/3/31-30 out tone interdigit duration 2000

voice signaling out dialing type

Command Usage

Specify out dialing characteristics (tone or pulse) of the channels duration on the port.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> out dialing type {tone | pulse}

Definitions:

<i>slot</i>	Specifies slot number of voice daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

tone Specifies use of Dual Tone Multifrequency (DTMF) dialing.

pulse Specifies use of rotary pulse dialing.

Default:

The default value is **tone**.

Command Examples:

voice signaling channel 2/1/1-12 out dialing type tone

voice signaling channel 2/2/13-24 out dialing type pulse

Remarks

DTMF (Dual Tone Multi Frequency) tones are comprised of eight tones divided into high and low frequency groups for signaling dialed numbers. Each DTMF tone has one high and one low tone each corresponding to a key on a push button dialing pad. Older and less commonly used rotary or circular dials cause breaks in the call circuit flow to signal each number dialed.

voice signaling call duration limit

Command Usage

Specify maximum call time length (channel timing). The call is automatically terminated when the allotted call time length is exceeded.

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> [**no**] **call duration limit** <*value*>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no	Specifies no call time length (i.e., infinite).
<i>value</i>	Specifies maximum call time length in seconds, from 0 ms to 65,534.

◆ Syntax Notes ◆

Do not use commas when entering the maximum call duration limit (for example, **65,534** will return a syntax error message).

Do not use **no** if entering a value.

A “no call duration limit” means the call can go on forever.

A “no call duration limit 100” or any other number of seconds, means the call can go on forever.

Default:

The default setting is **no call duration limit**.

Command Examples:

```
voice signaling channel 2/1/1-12 no call duration limit
voice signaling channel 2/2/13-24 call duration limit 0
voice signaling channel 2/3/1-30 no call duration limit 100
voice signaling channel 2/4/1-30 call duration limit 65534
```

voice signaling answer wait limit

Command Usage

Specify time to wait for call to be answered (channel timing). The call is automatically terminated if the call destination does not answer within the allotted time period for answering.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> [**no**] **answer wait limit** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Specifies no answer wait limit (i.e., infinite).

value Specifies time to wait before disconnecting a call when there is no answer at the destination, in seconds, from 0 ms to 65,534.

◆ Syntax Notes ◆

Do not use commas when entering the no answer wait limit (for example, **65,534** will return a syntax error message).

Do not use **no** if entering a value.

A “no answer wait limit” means wait forever for the call to be answered.

A “no answer wait limit 100” or any other number of seconds, means wait forever for calls to be answered.

Default:

The default setting is **no answer wait limit**.

Command Examples:

```
voice signaling channel 2/1/1-12 no answer wait limit
voice signaling channel 2/2/13-24 answer wait limit 0
voice signaling channel 2/3/1-20 no answer wait limit 100
```

voice signaling hang up wait limit

Command Usage

Specify time to wait to force caller to disconnect (channel timing). The call is automatically terminated if the call originator does not hang up within the allotted time period after the destination has hung up.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> [**no**] hang up wait limit <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no	Specifies no hang up wait limit (i.e., infinite).
<i>value</i>	Specifies time to wait to before disconnecting a call once the call originator has hung up the phone, in seconds, from 0 ms to 65,534.

◆ Syntax Notes ◆

Do not use commas when entering the “no hang up wait limit” (for example, **65,534** will return a syntax error message).

Do not use **no** if entering a value.

A “no hang up wait limit” means the call remains connected until the call originator hangs up the phone.

A “no hang up wait limit 100” or any other number of seconds, is considered an invalid setting.

Default:

The default setting is **no hang up wait limit**.

Command Examples:

voice signaling channel 2/1/1-12 no hang up wait limit
voice signaling channel 2/2/13-24 hang up wait limit 0

voice signaling fax holdover

Command Usage

Specify time to wait to tear down fax call (channel timing), i.e., set the fax call holdover delay. The delay occurs after an on-hook state is detected during fax mode operation, and before a call clear signal is generated on the line.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fax holdover** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the fax call holdover delay in milliseconds from 0 ms to 65,534, (e.g., 2000).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fax call holdover value (for example, **2,000** will return a syntax error message).

Default:

The default value is **2000**.

Command Examples:

```
voice signaling channel 2/1/1-12 fax holdover 2000
voice signaling channel 2/2/13-24 fax holdover 0
voice signaling channel 2/3/1-30 fax holdover 65534
```

voice signaling companding

Command Usage

Define companding type (Mu-Law/A-Law) for signaling power at the digital signal processing pulse code modulation (PCM) interface.

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> **companding** {*alaw*|*mulaw*}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

alaw Specifies PCM A-Law companding.

mulaw Specifies PCM Mu-Law companding.

Default:

For *VSD only*, the default setting is **mulaw**. For *VSB only*, the default setting is **alaw**.

Command Examples:

voice signaling channel 2/1/1-12 companding mulaw
voice signaling channel 2/2/13-24 companding alaw

Remarks

A-Law companding should be used when the voice port interface type is set to E1, or E1 ISDN PRI, or Euro BRI.

Mu-Law companding cannot be used on the VSB daughtercard.

voice signaling receive gain

Command Usage

Specify gain in signaling power inserted at *receiver*.

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> **receive gain** <*gain_value*>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>gain_value</i>	Specifies the numerical gain value. Values may range from -14 through 14 (e.g., -13, -2, 0, 4, 13 , etc). A negative gain value lowers the gain and a positive gain value increases the gain.
-------------------	--

Default:

The default gain value is **0**.

Command Examples:

voice signaling channel 2/1/1-12 receive gain 0
voice signaling channel 2/3/1-30 receive gain 14

Remarks

Transmit (TX) and receive (RX) gain is normally used when a device has volume problems. Gain adjusts (increases or decreases) the signal level. Transmit and receive signal gains occur per call on each end of the call. Signal gains are calculated per channel. The signal gains are applied at the channel level. Each channel on a voice switching daughtercard can have a different gain applied. Total gain is calculated between TX/RX points, e.g., if the receiving VSD is set to a gain of -3 and the transmitting VSD is set to a gain of 1, the gain would be -2.

The gain inserted at the receiver comes from the voice switching daughtercard which interprets receive gain as PCM interface-to-packet (H.323) network. This means that the gain is applied to the PCM packet when it is received by the card, and after the packet has been converted to H.323. Analog voice switching daughtercards (VSAs) convert signals to/from PCM before transmit or receive gains are applied.

voice signaling transmit gain

Command Usage

Specify gain in signaling power inserted at *transmitter*.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **transmit gain** <gain_value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

gain_value Specifies the numerical gain value in decibels. Values may range from -14 through 14 (e.g., **-13, -2, 0, 4, 13**, etc).

Default:

The default gain value is **0**.

Command Examples:

```
voice signaling channel 2/1/1-12 transmit gain 0
voice signaling channel 2/2/13-24 transmit gain -14
```

Remarks

Transmit (TX) and receive (RX) gain is normally used when a device has volume problems. Gain adjusts (increases or decreases) the signal level. Transmit and receive signal gains occur per call on each end of the call. Signal gains are calculated per channel. The signal gains are applied at the channel level. Each channel on a voice switching daughtercard can have a different gain applied. Total gain is calculated between TX/RX points, e.g., if one VSD is set to a gain of -3 and another to a gain of 1, the gain would be -2.

The gain inserted at the transmitter comes from the voice switching daughtercard which interprets transmit gain as (H.323) packet network-to-PCM interface. This means that the gain is applied to the PCM packet when it is transmitted by the card, and after the packet has been converted from H.323. Analog voice switching daughtercards (VSAs) convert signals to/from PCM before transmit or receive gains are applied.

voice signaling idle noise

Command Usage

Specify signaling power amplitude for comfort (idle) noise.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> idle noise <value>

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

- value* Specifies the idle noise level (comfort noise) in 0.01 decibels from -7000 to 7000, (e.g., **1000**); a value of -5000 means -50 decibels. A value of 0 means no comfort noise.

◆ **Syntax Note** ◆

Do not use commas when entering the idle noise value (for example, **1,000** will return a syntax error message).

Default:

The default value is **0**.

Command Examples:

- voice signaling channel 2/1/1-12 idle noise 0**
- voice signaling channel 2/2/13-24 idle noise 1000**
- voice signaling channel 2/3/1-30 idle noise 7000**
- voice signaling channel 2/4/1-30 idle noise -7000**

voice signaling em on hook debounce

Command Usage

Specify E&M signaling time for *transition* (debounce) *to on-hook*.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **em on hook debounce** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the E&M debounce (delay interval) transition to on-hook state in milliseconds from 5 to 1,000, (e.g., 50).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the E&M transition to on-hook state or debounce transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **50**.

Command Examples:

```
voice signaling channel 2/1/1-12 em on hook debounce 50
voice signaling channel 2/2/13-24 em on hook debounce 5
voice signaling channel 2/3/1-30 em on hook debounce 1000
```

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em off hook debounce

Command Usage

Specify Ear & Mouth signaling time for *transition* (debounce) to *off-hook*.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> em off hook debounce <value >

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the E&M debounce (delay interval) transition to off-hook (dial tone) state in milliseconds from 5 to 1,000, (e.g., **50**).

◆ Syntax Note ◆

Do not use commas when entering the E&M off-hook debounce transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **50**.

Command Examples:

- voice signaling channel 2/1/1-12 em off hook debounce 50**
- voice signaling channel 2/2/13-24 em off hook debounce 5**
- voice signaling channel 2/3/1-30 em off hook debounce 1000**

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em seize detect

Command Usage

Specify E&M signaling time to wait before *declaring on-hook* (seize detect), i.e., amount of time M-lead must be off-hook before an incoming call is declared.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **em seize detect** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the <i>E-lead</i> seize detect (delay interval) in milliseconds from 5 to 5,000, (e.g., 50) before an incoming call is declared. <i>M-lead</i> must be off-hook before an incoming call can be declared.
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the E&M seize detect value on the E-lead (for example, **5,000** will return a syntax error message).

Default:

The default value is **150**.

Command Examples:

```
voice signaling channel 2/1/1-12 em seize detect 150
voice signaling channel 2/2/13-24 em seize detect 5
voice signaling channel 2/3/1-30 em seize detect 5000
```

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em clear detect

Command Usage

Specify E&M Signaling time to wait before *declaring off-hook* (clear detect), i.e., time off-hook before call clearing is declared.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> em clear detect <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the <i>E-lead</i> clear detect (delay interval) in milliseconds from 5 to 5,000, (e.g., 50) before call clearing is declared. <i>M-lead</i> needs to be on-hook before call clearing can be declared.
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the E&M clear detect value on the M-lead (for example, **5,000** will return a syntax error message).

Default:

The default value is **400**.

Command Examples:

voice signaling channel 2/2/13-24 em clear detect 400
voice signaling channel 2/3/1-30 em clear detect 5000

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em clear confirm detect

Command Usage

Specify E&M signaling time to wait before confirming on-hook.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> em clear confirm detect <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the *E-lead* clear confirm detect (delay interval) in milliseconds from 5 to 5,000, (e.g., **50**) before call clear confirm is declared.

◆ Syntax Note ◆

Do not use commas when entering the E&M clear confirm detect transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **5000**.

Command Examples:

```
voice signaling channel 2/1/1-12 em clear confirm detect 5000
voice signaling channel 2/2/13-24 em clear confirm detect 5
voice signaling channel 2/3/ 1-30 em clear confirm detect 1000
```

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em clear confirm wait max

Command Usage

Specify Ear & Mouth signaling time to wait for on-hook after a clear.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **em clear confirm wait max**[imum] <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **max** or **maximum** in the command line.

value Specifies the maximum duration (delay interval) to wait for an on-hook response on the *M-lead* after going on-hook on the *E-lead*, (e.g., **25000**).

◆ Syntax Note ◆

Do not use commas when entering the E&M on-hook after a clear detect value (for example, **60,000** will return a syntax error message).

Default:

The default value is **60000**.

Command Examples:

```
voice signaling channel 2/1/1-12 em clear confirm wait maximum 60000
voice signaling channel 2/2/13-24 em clear confirm wait max 25000
voice signaling channel 2/3/1-30 em clear confirm wait max 45000
```

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em guard all

Command Usage

Specify Ear & Mouth signaling time to wait between termination and *origination*.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **em guard all** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	After an aborted call, specifies the time period (delay interval) in milliseconds, (e.g., 400) from 0 to 60,000, when neither incoming nor outgoing calls are accepted or initiated.
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the E&M time span (in which no incoming or outgoing calls are accepted or initiated) for call termination and origination value (for example, **10,000** will return a syntax error message).

Default:

The default value is **400**.

Command Examples:

```
voice signaling channel 2/1/1-12 em guard all 400
voice signaling channel 2/2/13-24 em guard all 20000
voice signaling channel 2/3/1-30 em guard all 60000
```

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em guard out

Command Usage

Specify Ear & Mouth signaling time to wait between termination and *receiving*.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **em guard out** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the time period (delay interval) in milliseconds, (e.g., **400**) from 0 to 60,000, when only incoming calls are accepted or initiated. Outgoing calls are aborted.

◆ Syntax Note ◆

Do not use commas when entering the E&M extended time span (in which only incoming calls are accepted or initiated) for call termination and receiving value (for example, **10,000** will return a syntax error message).

Default:

The default value is **400**.

Command Examples:

```
voice signaling channel 2/1/1-12 em guard out 400
voice signaling channel 2/2/13-24 em guard out 20000
voice signaling channel 2/3/1-30 em guard out 60000
```

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em dial tone

Command Usage

Set Ear & Mouth signaling dial tone generation on incoming calls (on/off).

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> **em dial tone** {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON E&M dial tone generation for incoming calls.
off	Turns OFF E&M dial tone generation for incoming calls.

Default:

The default setting is **none**.

Command Examples:

```
voice signaling channel 2/1/1-12 em dial tone on
voice signaling channel 2/2/13-24 em dial tone off
voice signaling channel 2/3/1-30 em dial tone on
```

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em min connection time

Command Usage

Specify minimum Ear & Mouth signaling connection time.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **em min[imum] connect time** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **min** or **minimum** in the command line.

value Specifies the minimum time period, in milliseconds from 0 to 20,000, (e.g., **2000**), that a connection is maintained.

◆ Syntax Note ◆

Do not use commas when entering the minimum E&M connection value (for example, **2,000** will return a syntax error message).

Default:

The default value is **2000**.

Command Examples:

voice signaling channel 2/3/1-30 em minimum connection time 1000

Remarks

If the remote (or called) end disconnects during the specified minimum E&M connection time, the disconnect is acknowledged by a dial tone.

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling em hang up wait

Command Usage

Specify time to wait after Ear & Mouth signaling hang up before disabling.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **em hang up wait** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the time period, in milliseconds, from 5 to 60,000, (e.g., 2000) a port waits to hang up before signaling that the call is in a disabled state.
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the value for E&M time to wait after signaling call hang up before disabling (for example, **2,000** will return a syntax error message).

Default:

The default value is **2000**.

Command Examples:

```
voice signaling channel 2/1/1-12 hang up wait 2000
voice signaling channel 2/2/13-24 em hang up wait 10000
voice signaling channel 2/2/1-30 em hang up wait 20000
```

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Wink Start, Immediate Start, or Delay Start Signaling.

voice signaling emw in wink wait min

Command Usage

Specify *minimum* Ear & Mouth wink delay on incoming calls.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel > **emw in wink wait min**[imum]
<value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **min** or **minimum** in the command line.

value Specifies the minimum delay in milliseconds, from 5 to 60,000, (e.g., **150**), before beginning the wink on the *E-lead* after detecting a line seizure on the *M-lead* for incoming calls.

◆ Syntax Note ◆

Do not use commas when entering the minimum E&M wink delay value on incoming calls (for example, **24,000** will return a syntax error message).

Default:

The default value is **2000**.

Command Examples:

```
voice signaling channel 2/1/1-12 emw in wink wait minimum 150
voice signaling channel 2/2/13-24 emw in wink wait min 30000
voice signaling channel 2/3/1-30 emw in wink wait min 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw in wink wait max

Command Usage

Specify *maximum* Ear & Mouth wink delay on incoming calls.

Syntax Options

```
voice signaling channel <slot/port/startChannel-endChannel > emw in wink wait max[imum]
<value >
```

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **max** or **maximum** in the command line.

value Specifies the maximum delay in milliseconds, from 5 to 60,000, (e.g., **3000**), before beginning the wink on the *E-lead* after detecting a line seizure on the *M-lead* for incoming calls.

◆ Syntax Note ◆

Do not use commas when entering the maximum E&M wink delay value on incoming calls (for example, **3,000** will return a syntax error message).

Default:

The default value is **3000**.

Command Examples:

```
voice signaling channel 2/1/1-12 emw in wink wait maximum 3000
voice signaling channel 2/2/13-24 emw in wink wait max 30000
voice signaling channel 2/3/1-30 emw in wink wait max 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw in wink duration

Command Usage

Specify duration of Ear & Mouth wink delay on incoming calls.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> emw in wink duration <value >

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

- value* Specifies the duration of the wink signal on the *E-lead* for incoming calls in milliseconds from 5 to 60,000, (e.g., **200**).

◆ Syntax Note ◆

Do not use commas when entering the E&M wink duration value for incoming calls (for example, **200** will return a syntax error message).

Default:

The default value is **200**.

Command Examples:

- voice signaling channel 2/1 1-12 emw in wink duration 200**
- voice signaling channel 2/2 13-24 emw in wink duration 30000**
- voice signaling channel 2/3 1-30 emw in wink duration 60000**

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw in wink digit ignore

Command Usage

Specify time to ignore tone (digits) after Ear & Mouth wink.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emw in wink digit ignore** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the period of time, in milliseconds from 5 to 1,000, (e.g., 30), after the wink to ignore tones (digits) for incoming calls.
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the value for E&M time to ignore digits for incoming after wink (for example, **1,000** will return a syntax error message).

Default:

The default value is **30** milliseconds.

Command Examples:

```
voice signaling channel 2/1 1-12 emw in wink digit ignore 30
voice signaling channel 2/2 13-24 emw in wink digit ignore 5
voice signaling channel 2/3 1-30 emw in wink digit ignore 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw out wink wait max

Command Usage

Specify time to wait for E&M wink on outgoing calls.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emw out wink wait max**[imum] <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **max** or **maximum** in the command line.

value Specifies the maximum time to wait for a wink response on the *M-lead* after going off-hook on the *E-lead*, in milliseconds from 5 to 60,000, (e.g., **8000**).

◆ Syntax Note ◆

Do not use commas when entering the E&M maximum time to wait value for a wink response on outgoing calls (for example, **8,000** will return a syntax error message).

Default:

The default value is **8000** milliseconds.

Command Examples:

voice signaling channel 2/1/1-12 emw out wink wait maximum 8000
voice signaling channel 2/1/13-24 emw out wink wait max 30000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw out wink duration min

Command Usage

Specify *minimum* E&M wink duration.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emw out wink duration min**[imum] <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **min** or **minimum** in the command line.

value Specifies the minimum duration of the wink response to the *M-lead* for detection by the *M-lead*, in milliseconds from 5 to 60,000, (e.g., **100**).

◆ Syntax Note ◆

Do not use commas when entering the value for the minimum duration of the wink response to the M-lead for detection (for example, **1,000** will return a syntax error message).

Default:

The default value is **100** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 emw out wink duration min 100
voice signaling channel 2/2/13-24 emw out wink duration min 30000
voice signaling channel 2/3/1-30 emw out wink duration min 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emw out wink duration max

Command Usage

Specify *maximum* E&M wink duration.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emw out wink duration max**[imum] <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **max** or **maximum** in the command line.

value Specifies the maximum duration of the wink response on the *M-lead* before glare condition (trunk ends are seized simultaneously) declared on outgoing call, in milliseconds from 5 to 60,000, (e.g., **800**).

◆ Syntax Note ◆

Do not use commas when entering the value for the maximum duration of the wink response to the M-lead for detection (for example, **1,000** will return a syntax error message).

Default:

The default value is **800** milliseconds.

Command Example:

voice signaling channel 2/3/1-30 emw out wink duration max 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

Use this command to reduce instances of trunk deadlock caused by glare (trunks which are seized on both ends of a call at the same time).

voice signaling emi glare report

Command Usage

Specify E&M immediate start time to remain off-hook when congested.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emi glare report** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies time period, in milliseconds from 0 to 60,000, (e.g., 5000), wherein if glare condition is detected, the line stays off-hook and generates a congestion tone.
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the E&M immediate start-time value in which a glare condition is reported and a congestion tone is generated while off-hook (for example, **5,000** will return a syntax error message).

Default:

The default value is **5000** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 emi glare report 5000
voice signaling channel 2/2/13-24 emi glare report 30000
voice signaling channel 2/3/1-30 emi glare report 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emi digit wait

Command Usage

Specify E&M immediate start time to wait before beginning digit collection.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> emi digit wait <value >

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

- value* Specifies time period, in milliseconds from 0 to 60,000, (e.g., **200**), to wait for the voice daughtercard to be ready before collecting digits. The line stays off-hook and generates a dial tone if glare condition detected in the interim.

◆ **Syntax Note** ◆

Do not use commas when entering the E&M immediate start time value, i.e., time to wait for before digit collection is enabled and an off-hook congestion tone with glare condition is reported (for example, **5,000** will return a syntax error message).

Default:

The default value is **200** milliseconds.

Command Examples:

- voice signaling channel 2/1/1-12 emi digit wait 200**
- voice signaling channel 2/2/13-24 emi digit wait 30000**
- voice signaling channel 2/3/1-30 emi digit wait 60000**

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd in delay min

Command Usage

Specify minimum E&M delay start response to off-hook (dial tone) state.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emd in delay min**[imum] <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum	Optional command syntax. Type either min or minimum in the command line.
<i>value</i>	Specifies the minimum duration of the delay signal response to the seize detect of incoming calls on the <i>M-lead</i> , in milliseconds from 0 to 60,000, (e.g., 200).

◆ Syntax Note ◆

Do not use commas when entering the minimum value for E&M delay start response to off-hook (dial tone) state (for example, **5,000** will return a syntax error message).

Default:

The default value is **200** milliseconds.

Command Examples:

```
voice signaling channel 2/1-15 emd in delay minimum 200
voice signaling channel 2/2/13-24 emd in delay min 30000
voice signaling channel 2/3/1-30 emd in delay min 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd in delay max

Command Usage

Specify maximum E&M delay start response to off-hook (dial tone) state.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emd in delay max**[imum] <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum	Optional command syntax. Type either max or maximum in the command line.
<i>value</i>	Specifies the maximum duration of the delay signal response to the seize detect of incoming calls on the <i>M-lead</i> , in milliseconds from 5 to 60,000, (e.g., 2500).

◆ Syntax Note ◆

Do not use commas when entering the maximum value for E&M delay start response to off-hook (dial tone) state (for example, **5,000** will return a syntax error message).

Default:

The default value is **2500** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 emw in delay maximum 2500
voice signaling channel 2/2/13-24 emw in delay max 30000
voice signaling channel 2/3/1-30 emw in delay max 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd in digit ignore

Command Usage

Specify time to ignore incoming digits after E&M delay start.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emd in digit ignore** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the period of time after which E&M delay signal is completed before digits on incoming calls are accepted, in milliseconds from 5 to 60,000, (e.g., 2500).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the value for time to ignore incoming digits after E&M delay start (for example, **5,000** will return a syntax error message).

Default:

The default value is **2500** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 emd in digit ignore 2500
voice signaling channel 2/2/13-24 emd in digit ignore 30000
voice signaling channel 2/3/1-30 emd in digit ignore 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd out integrity check

Command Usage

Specify E&M delay start signal detection.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> emd out integrity check {on | off}

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

- on** Turns ON E&M delay integrity check on outgoing calls.
- off** Turns OFF E&M delay integrity check on outgoing calls.

Default:

The default setting is **off**.

Command Examples:

voice signaling channel 2/1/1-12 emd out integrity check off
voice signaling channel 2/2/13-24 emd out integrity check on

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

When the Integrity Check mode is ON, the delay signal response is required from the PBX for outgoing calls.

voice signaling emd out delay duration min

Command Usage

Specify *minimum* E&M delay start detection time on “M” lead.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emd out delay duration min**[imum] <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **min** or **minimum** in the command line.

value Specifies the minimum duration of the delay signal response on the *M-lead* for detection on outgoing calls, in milliseconds from 5 to 60,000, (e.g., **100**).

◆ Syntax Note ◆

Do not use commas when entering the value for the minimum E&M delay start detection on the M-lead signal (for example, **1,000** will return a syntax error message).

Default:

The default value is **100** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 emd out delay duration minimum 100
voice signaling channel 2/2/13-24 emd out delay duration min 30000
voice signaling channel 2/3/1-30 emd out delay duration min 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd out detail duration max

Command Usage

Specify *maximum* E&M delay start detection time on “M” lead.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emd out delay duration max**[imum] <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **max** or **maximum** in the command line.

value Specifies the maximum duration of the delay signal response on the *M-lead* before a glare condition on outgoing calls can be declared, in milliseconds from 5 to 60,000, (e.g., **100**).

◆ Syntax Note ◆

Do not use commas when entering the value for the maximum E&M delay signal response on the M-lead before a glare condition can be declared (for example, **1,000** will return a syntax error message).

Default:

The default value is **8000** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 emd out delay duration maximum 8000
voice signaling channel 2/2/13-24 emd out delay duration max 3000
voice signaling channel 2/3/1-30 emd out delay duration max 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling emd out delay check

Command Usage

Specify *maximum* time to wait for E&M delay start detection.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **emd out delay check** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the period of time after going off-hook on the <i>E-lead</i> before checking the <i>M-lead</i> for the delay signal response, in milliseconds from 5 to 60,000, (e.g., 170). If the response is not detected in the interim, the call setup process resumes immediately.
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the value for the maximum time to wait for E&M delay start detection (for example, **5,000** will return a syntax error message).

Default:

The default value is **170** milliseconds.

Command Examples:

voice signaling channel 2/1/1-12 emd out delay check 170
voice signaling channel 2/2/13-24 emd out delay check 30000

Remarks

In order to use this command, the **voice signaling protocol** command must be set to the corresponding protocol type of E&M Delay Start Signaling.

In order to use this command, the integrity check for outgoing calls must be turned OFF via the **voice signaling emd out integrity check** command.

voice signaling fxs ls on hook debounce

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) debounce interval to *on-hook* transition.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxs ls on hook debounce <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the debounce (delay interval) transition to on-hook state in milliseconds from 1 to 1,000, (e.g., 20).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxs ls on-hook debounce transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs ls on hook debounce 200
voice signaling channel 2/2/13-24 fxs ls on hook debounce 500
voice signaling channel 2/3/1-30 fxs ls on hook debounce 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

A minimum debounce value of **250** must be set for analog voice switching (VSA) daughter-cards.

voice signaling fxs ls off hook debounce

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) debounce interval to *off-hook* transition.

Syntax Options

```
voice signaling channel <slot/port/startChannel-endChannel> fxs ls off hook debounce <value >
```

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the debounce (delay interval) transition to off-hook (dial tone) state in milliseconds from 1 to 1,000, (e.g., **20**).

◆ Syntax Note ◆

Do not use commas when entering the fxs ls off-hook debounce transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs ls off hook debounce 20
voice signaling channel 2/2/13-24 fxs ls off hook debounce 500
voice signaling channel 2/3/1-30 fxs ls off hook debounce 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

A minimum debounce value of **250** must be set for analog voice switching (VSA) daughter-cards.

voice signaling fxs ls seize detect

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) time to wait before declaring off-hook.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs ls seize detect** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the time to wait before off-hook condition is declared, in milliseconds from 1 to 5,000, (e.g., 150).
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the fxs ls time to wait value before declaring off-hook condition (for example, **1,000** will return a syntax error message).

Default:

The default value is **150** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs ls seize detect 150
voice signaling channel 2/2/13-24 fxs ls seize detect 2500
voice signaling channel 2/3/1-30 fxs ls seize detect 5000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls originate clear detect

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) minimum time to wait before declaring on-hook by *originator* (calling party).

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxs ls originate clear detect <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the minimum time to wait if call originator hangs up before on-hook condition declared, in milliseconds from 1 to 60,000, (e.g., 300).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxs ls call originate clear detect value (for example, **1,000** will return a syntax error message).

Default:

The default value is **300** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs ls originate clear detect 300
voice signaling channel 2/1/13-24 fxs ls originate clear detect 30000
voice signaling channel 2/1/1-30 fxs ls originate clear detect 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls answer clear detect

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) minimum time to wait before declaring on-hook by *answerer* (called party) after the calling party has ended the call.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxs ls answer clear detect <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the minimum time to wait if answering party hangs up before on-hook condition declared, in milliseconds from 1 to 60,000, (e.g., 300).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxs ls answering party clear detect value (for example, **1,000** will return a syntax error message).

Default:

The default value is **300** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs ls answer clear detect 300
voice signaling channel 2/2/13-24 fxs ls answer clear detect 30000
voice signaling channel 2/3/1-30 fxs ls answer clear detect 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls supervisory disconnect wait

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) time to wait after supervisory disconnect before declaring on-hook.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxs ls supervisory disconnect wait <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the fxs ls supervisory disconnect (CPC signal) wait value, in milliseconds from 1 to 60,000, (e.g., 200) is generated before on-hook condition declared.
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxs ls supervisory disconnect wait value (for example, **1,000** will return a syntax error message).

Default:

The default value is **200** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs ls supervisory disconnect wait 200
voice signaling channel 2/2/16-24 fxs ls supervisory disconnect wait 30000
voice signaling channel 2/3/1-30 fxs ls supervisory disconnect wait 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls supervisory disconnect duration

Command Usage

Specify Foreign Exchange Station Loop Start (FXS LS) duration of supervisory disconnect.

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> **fxs ls supervisory disconnect duration** <*value*>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the fxs ls supervisory disconnect (CPC signal) duration value, in milliseconds from 1 to 60,000, (e.g., **850**) generated before on-hook condition declared.

◆ Syntax Note ◆

Do not use commas when entering the fxs ls supervisory disconnect wait value (for example, **10,000** will return a syntax error message).

Default:

The default value is **850** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs ls supervisory disconnect duration 850
voice signaling channel 2/2/13-24 fxs ls supervisory disconnect duration 30000
voice signaling channel 2/3/1-30 fxs ls supervisory disconnect duration 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls caller id

Command Usage

Set Foreign Exchange Station Loop Start (FXS LS) to *generate outbound* caller ID (on/off).

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs ls caller id** {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON fxs ls caller ID for specified telephony signaling channel.
off	Turns OFF fxs ls caller ID for specified telephony signaling channel.

◆ Syntax Note ◆

The **voice coding profile caller id** command must be enabled to use this command.

Default:

The default setting is **off**.

Command Examples:

voice signaling channel 2/1/1-12 fxs ls caller id off
voice signaling channel 2/2/13-24 fxs ls caller id on

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs ls ringing debounce

Command Usage

Specify Foreign Exchange Office Loop Start (FXO LS) incoming ring signal debounce interval.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel > fxo ls ringing debounce <value >

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

- value* Specifies the fxo ls incoming ring signal debounce (delay interval) in milliseconds from 1 to 1,000, (e.g., **50**).

◆ Syntax Note ◆

Do not use commas when entering the fxo ls incoming ring signal debounce value (for example, **1,000** will return a syntax error message).

Default:

The default value is **50** milliseconds.

Command Examples:

- voice signaling channel 2/1/1-12 fxo ls ringing debounce 50**
- voice signaling channel 2/2/13-24 fxo ls ringing debounce 500**
- voice signaling channel 2/3/1-30 fxo ls ringing debounce 1000**

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

A minimum debounce value of **500** must be set for analog voice switching (VSA) daughter-cards.

voice signaling fxo ls supervisory disconnect detection

Command Usage

Set Foreign Exchange Office Loop Start (FXO LS) *supervisory disconnect* detection signal (enable/disable).

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxo ls supervisory disconnect detection {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON fxo ls detection of supervisory disconnect for specified telephony signaling channel.
off	Turns OFF fxo ls detection of supervisory disconnect for specified telephony signaling channel.

◆ Syntax Note ◆

To use this command properly, the duration of the fxo ls supervisory disconnect signal must be specified via the **voice signaling fxo ls supervisory disconnect** command.

Default:

The default setting is **on**.

Command Examples:

voice signaling channel 2/1/1-12 fxo ls supervisory disconnect detection on
voice signaling channel 2/2/13-24 fxo ls supervisory disconnect detection off

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls supervisory disconnect

Command Usage

Specify Foreign Exchange Office (FXO) Loop Start *duration of supervisory disconnect* detection signal.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxo ls supervisory disconnect** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the fxo ls incoming ring signal debounce (delay interval) in milliseconds from 1 to 60,000, (e.g., **600**). If the loop current drops below the specified value for a period of time, it is not considered a “supervisory disconnect” of the signal.

◆ Syntax Notes ◆

Do not use commas when entering the duration of the fxo ls supervisory disconnect (CPC signal) value (for example, **10,000** will return a syntax error message).

To use this command, detection of the fxo ls supervisory disconnect signal must be turned ON via the **voice signaling fxo ls supervisory disconnect detection** command.

Default:

The default value is **600** milliseconds.

Command Examples:

voice signaling channel 2/1/1-12 fxo ls supervisory disconnect 600
voice signaling channel 2/3/1-30 fxo ls supervisory disconnect 60000

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls guard out

Command Usage

Specify Foreign Exchange Office (FXO) Loop Start allowed time period after a call when a channel cannot originate, but may receive a call.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxo ls guard out <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the fxs ls guard out allowed time period, in milliseconds from 1 to 60,000 (e.g., 2000) after a call, when a channel cannot originate but may receive a call.
--------------	---

◆ Syntax Notes ◆

Do not use commas when entering the duration of the fxo ls guard out value (for example, **10,000** will return a syntax error message).

Default:

The default value is **2000** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo ls guard out 50
voice signaling channel 2/2/13-24 fxo ls guard out 5
voice signaling channel 2/2/1-30 fxo ls guard out 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls ringing inter cycle

Command Usage

Specify Foreign Exchange Office (FXO) Loop Start time between ring *cycles* to detect ringing.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxo ls ringing inter cycle** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the time period between ring cycles (ringing pulse plus time between ring pulses) to detect ringing, in milliseconds from 1 to 60,000, (e.g., 2000)
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the fxo ls between ring cycles ring detection value (for example, **10,000** will return a syntax error message).

Default:

The default value is **2000** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo ls ringing inter cycle 2000
voice signaling channel 2/2/13-24 fxo ls ringing inter cycle 30000
voice signaling channel 2/3/1-30 fxo ls ringing inter cycle 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls ringing inter pulse

Command Usage

Specify Foreign Exchange Office (FXO) Loop Start time between ring *pulses* to detect ringing.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxo ls ringing inter pulse <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the time period between ring pulses to detect ringing, in milliseconds from 1 to 60,000, (e.g., 550).
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the fxo ls between ring pulses ring detection value (for example, **10,000** will return a syntax error message).

Default:

The default value is **550** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo ls ringing inter pulse 550
voice signaling channel 2/2/13-24 fxo ls ringing inter pulse 30000
voice signaling channel 2/3/1-30 fxo ls ringing inter pulse 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls caller id

Command Usage

Set Foreign Exchange Office Loop Start (FXO LS) to *detect inbound* caller ID (on/off) before the second ring.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxo ls caller id {on | off}

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

- on** Turns ON fxo ls caller ID for specified telephony signaling channel.
- off** Turns OFF fxo ls caller ID for specified telephony signaling channel.

◆ **Syntax Note** ◆

The **voice coding profile caller id** command must be enabled to use this command.

Default:

The default setting is **on**.

Command Examples:

- voice signaling channel 2/1/1-12 fxo ls caller id on**
- voice signaling channel 2/2/13-24 fxo ls caller id off**

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls answer after

Command Usage

Specify Foreign Exchange Office Loop Start (FXO LS) number of rings allowed before answering an incoming call (inseize).

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxo ls answer after <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the number of rings permitted to elapse before answering incoming calls, per ring from 1 to 65,535, (e.g., 2) rings. The value must be at least or caller ID will not function properly.
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxs gs call originate clear detect value (for example, **10,000** will return a syntax error message).

Default:

The default value is **2** rings.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo ls answer after 2
voice signaling channel 2/2/13-24 fxo ls answer after 30000
voice signaling channel 2/3/1-30 fxo ls answer after 65535
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls loop current debounce

Command Usage

Specify Foreign Exchange Office Loop Start (FXO LS) debounce for loop current detector.

Syntax Options

voice signaling channel *<slot/port/startChannel-endChannel>* **fxo ls loop current debounce** *<value>*

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the time to use as a debouncer (delay interval) for debouncing the loop current detector, in milliseconds from 0 to 60,000, (e.g., **20**).

◆ Syntax Note ◆

Do not use commas when entering the fxs ls debounce loop current detector value (for example, **10,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo ls loop current debounce 20
voice signaling channel 2/2/13-24 fxo ls loop current debounce 30000
voice signaling channel 2/3/1-30 fxo ls loop current debounce 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo ls battery reversal debounce

Command Usage

Specify Foreign Exchange Office Loop Start (FXO LS) debounce for battery reversal detector.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxo ls battery reversal debounce** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the time to use as a debouncer (delay interval) for debouncing the battery reversal detector, in milliseconds from 0 to 60,000, (e.g., 20).
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the fxo ls debounce value for the battery reversal detector (for example, **10,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo ls battery reversal debounce 20
voice signaling channel 2/2/13-24 fxo ls battery reversal debounce 30000
voice signaling channel 2/3/1-30 fxo ls battery reversal debounce 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs seize detect

Command Usage

Specify Foreign Exchange Station Ground Start (FXO GS) time to wait before declaring off-hook.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs gs seize detect** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the time to wait before declaring off-hook (seize detect) condition, in milliseconds from 0 to 5,000, (e.g., 150).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxs gs time to wait before declaring seize detect value (for example, **2,500** will return a syntax error message).

Default:

The default value is **150** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs gs seize detect 150
voice signaling channel 2/2/13-24 fxs gs seize detect 2500
voice signaling channel 2/3/1-30 fxs gs seize detect 5000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs on hook debounce

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) debounce interval for on-hook transition.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs gs on hook debounce** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the debounce (delay interval) transition time to on-hook (seize detect) condition, in milliseconds from 0 to 1,000, (e.g., 20).
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the fxs gs debounce interval on-hook transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs gs on hook debounce 20
voice signaling channel 2/2/13-24 fxs gs on hook debounce 500
voice signaling channel 2/3/1-24 fxs gs on hook debounce 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs originate clear detect

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) minimum time to wait before declaring on-hook by the call *originator* to the call *answerer*.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxs gs originate clear detect <value>

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

- value* Specifies the minimum time to wait if call originator hangs up before on-hook condition declared, in milliseconds from 1 to 60,000, (e.g., **200**).

◆ **Syntax Note** ◆

Do not use commas when entering the fxs gs call originate clear detect value (for example, **10,000** will return a syntax error message).

Default:

The default value is **200** milliseconds.

Command Examples:

- voice signaling channel 2/1/1-12 fxs gs originate clear detect 200**
- voice signaling channel 2/2/13-24 fxs gs originate clear detect 30000**
- voice signaling channel 2/3/1-30 fxs gs originate clear detect 60000**

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs answer clear detect

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) minimum time to wait before declaring on-hook by the call *answerer* to the call *originator*.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxs gs answer clear detect <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the minimum time to wait if answering party hangs up before on-hook condition declared, in milliseconds from 1 to 60,000, (e.g., 300).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxs gs answering party clear detect value (for example, **1,000** will return a syntax error message).

Default:

The default value is **100** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs gs answer clear detect 100
voice signaling channel 2/2/13-24 fxs gs answer clear detect 30000
voice signaling channel 2/3/1-30 fxs gs answer clear detect 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs min ring ground

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) time to wait after ring before grounding tip.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs gs min**[imum] **ring ground** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum	Optional command syntax. Type either min or minimum in the command line.
<i>value</i>	Specifies the minimum time to wait after ring ground detection before line responds by grounding tip, in milliseconds from 0 to 65535, (e.g., 100).

◆ Syntax Note ◆

Do not use commas when entering the minimum fxs gs time to wait value after ring ground before grounding tip (for example, **10,000** will return a syntax error message).

Default:

The default value is **100** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs gs minimum ring ground 100
voice signaling channel 2/2/13-24 fxs gs min ring ground 30000
voice signaling channel 2/2/1-30 fxs gs min ring ground 65535
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs max wait loop

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) maximum time to wait for loop to close after grounding tip.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs gs max[imum] wait loop** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum	Optional command syntax. Type either max or maximum in the command line.
<i>value</i>	Specifies the maximum time to wait after ring ground detection for loop to close after grounding tip but before disconnecting line, in milliseconds from 0 to 65535, (e.g., 100).

◆ Syntax Note ◆

Do not use commas when entering the maximum fxs gs time to wait value for loop to close after grounding tip (for example, **10,000** will return a syntax error message).

Default:

The default value is **100** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs gs maximum wait loop 100
voice signaling channel 2/1/13-24 fxs gs max wait loop 30000
voice signaling channel 2/1/1-30 fxs gs max wait loop 65535
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs min loop open

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) minimum time between open loop and idle state.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs gs min**[imum] **loop open** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum Optional command syntax. Type either **min** or **minimum** in the command line.

value Specifies the maximum time to wait after ring ground detection for loop to open after grounding tip but before returning line to idle state, in milliseconds from 0 to 65535, (e.g., **100**).

◆ Syntax Note ◆

Do not use commas when entering the maximum fxs gs time to wait value for loop to open after grounding tip (for example, **10,000** will return a syntax error message).

Default:

The default value is **100** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs gs minimum loop open 100
voice signaling channel 2/2/13-24 fxs gs min loop open 30000
voice signaling channel 2/2/1-30 fxs gs min loop open 65535
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs caller id

Command Usage

Set Foreign Exchange Station Ground Start (FXS GS) to *generate outbound* caller ID (on/off).

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxs gs caller id {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on Turns ON FXS GS caller ID for specified telephony signaling channel.

off Turns OFF FXS GS caller ID for specified telephony signaling channel.

◆ Syntax Note ◆

The **voice coding profile caller id** command must be enabled to use this command.

Default:

The default setting is **off**.

Command Examples:

voice signaling channel 2/1/1-12 fxs gs caller id off
voice signaling channel 2/2/13-24 fxs gs caller id on

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs off hook debounce

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) debounce interval for off-hook.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs gs off hook debounce** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the debounce (delay interval) transition time to off-hook condition, in milliseconds from 0 to 1,000, (e.g., 20).
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the fxs gs debounce interval on-hook transition value (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs gs off hook debounce 20
voice signaling channel 2/2/13-24 fxs gs off hook debounce 500
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs ring ground debounce

Command Usage

Specify Foreign Exchange Station Ground Start (FXS GS) debounce interval for ring ground detector.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs gs ring ground debounce** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies how long to use the debounce (delay interval) for debouncing the ring ground detector, in milliseconds from 0 to 1,000, (e.g., 20).
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the fxs gs debounce interval value for ring ground detection (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs gs ring ground debounce 20
voice signaling channel 2/2/13-24 fxs gs ring ground debounce 500
voice signaling channel 2/3/1-30 fxs gs ring ground debounce 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxs gs ring id

Command Usage

Set Foreign Exchange Station Ground Start (FXS GS) ring ID, or continental ring tone variance for North America or Europe.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxs gs ring id** {0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | default}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

0, 1, 2, 3, 4, 5, 6, 7 Specifies variance in continental ring tone for either North America or Europe (e.g., **2**) as per selected coefficient. The value **0** is the same as **default**.

default Sets the tone variance to the default setting, i.e., 0.

◆ Syntax Note ◆

This command must be set in relation to the **voice signaling fxs gs ring cadence coefficient** command.

Default:

The default value is **0**.

Command Examples:

```
voice signaling channel 2/1/1-12 fxs gs ring ID default
voice signaling channel 2/2/13-24 fxs gs ring ID 0
voice signaling channel 2/3/1-30 fxs gs ring ID 7
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

The ring cadence that is set for the first channel on the first port is automatically set for all remaining FXS/FXO ports.

All ports must have the same ring cadence.

On the OmniAccess 512, the ring cadence can only be specified for port 1.

On the Omni Switch/Router, the ring cadence can only be specified for port 1 (or 9 for a dual VSA-FXS/FXS in an HSX-H).

voice signaling fxo gs connection loop open debounce

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) debounce interval for loop open detection.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxo gs connection loop open debounce** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the debounce (delay interval) for loop open detection for existing connection, in milliseconds from 1 to 60,000, (e.g., 150).
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the fxs gs debounce interval value for loop open detection (for example, **1,000** will return a syntax error message).

Default:

The default value is **150** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo gs connection loop open debounce 150
voice signaling channel 2/2/13-24 fxo gs connection loop open debounce 30000
voice signaling channel 2/3/1-30 fxo gs connection loop open debounce 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs max tip ground wait

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) maximum time between ring ground and tip ground.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxo gs max[imum] tip ground wait** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

imum	Optional command syntax. Type either max or maximum in the command line.
<i>value</i>	Specifies the maximum time the line waits after ring ground asserted for tip ground received, in milliseconds from 1 to 60,000, (e.g., 150).

◆ Syntax Note ◆

Do not use commas when entering the maximum fxs gs value for time between ring ground and tip ground (for example, **1,000** will return a syntax error message).

Default:

The default value is **30** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo gs maximum tip ground wait 30
voice signaling channel 2/2/13-24 fxo gs max tip ground wait 30000
voice signaling channel 2/3/1-24 fxo gs max tip ground wait 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs tip ground debounce

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) debounce interval for tip ground detector.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxo gs tip ground debounce <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the debounce (delay interval) for debouncing the tip ground detector, in milliseconds from 1 to 1,000, (e.g., 20).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxs gs debounce interval value for loop open detection (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo gs tip ground debounce 20
voice signaling channel 2/2/13-24 fxo gs tip ground debounce 500
voice signaling channel 2/3/1-30 fxo gs tip ground debounce 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs ringing debounce

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) debounce for incoming ring signal.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxo gs ringing debounce** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the debounce (delay interval) for incoming ring signal, in milliseconds from 1 to 1,000, (e.g., 50).
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxs gs debounce interval value for incoming ring signal (for example, **1,000** will return a syntax error message).

Default:

The default value is **50** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo gs ringing debounce 50
voice signaling channel 2/2/13-24 fxo gs ringing debounce 5
voice signaling channel 2/3/1-30 fxo gs ringing debounce 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs ringing inter cycle

Command Usage

Specify Foreign Exchange Office Ground Start (FXS GS) time between consecutive ring *cycles*.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxo gs ringing inter cycle <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the time between fxs gs start time between consecutive ring cycles, in milliseconds from 1 to 1,000, (e.g., **5000**), to detect ringing.

◆ Syntax Note ◆

Do not use commas when entering the fxs gs start time between consecutive ring cycles to detect ringing (for example, **5,000** will return a syntax error message).

Default:

The default value is **5000** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo gs ringing inter cycle 5000
voice signaling channel 2/2/13-24 fxo gs ringing inter cycle 30000
voice signaling channel 2/3/1-30 fxo gs ringing inter cycle 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs ringing inter pulse

Command Usage

Specify Foreign Exchange Office Ground Start (FXO GS) time between consecutive ring pulses.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxo gs ringing inter pulse <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the time between fxo gs start time between consecutive ring pulses in the same ring cycle, in milliseconds from 1 to 60,000, (e.g., 550), to detect ringing.
--------------	--

◆ Syntax Note ◆

Do not use commas when entering the fxo gs start time between consecutive ring pulses to detect ringing (for example, **5,000** will return a syntax error message).

Default:

The default value is **550** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo gs ringing inter pulse 550
voice signaling channel 2/2/13-24 fxo gs ringing inter pulse 30000
voice signaling channel 2/3/1-30 fxo gs ringing inter pulse 60000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs caller id detection

Command Usage

Set Foreign Exchange Office (FXO) Ground Start (GS) to *detect inbound* caller ID (on/off).

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxo gs caller id detection** {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on Turns ON fxs gs caller ID detection for specified telephony signaling channel.

off Turns OFF fxs gs caller ID detection for specified telephony signaling channel.

◆ Syntax Note ◆

The **voice coding profile caller id** command must be enabled to use this command.

Default:

The default setting is **off**.

Command Examples:

voice signaling channel 2/1/1-12 fxo gs caller id detection off

voice signaling channel 2/2/13-24 fxo gs caller id detection on

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs answer after

Command Usage

Specify Foreign Exchange Office (FXO) Ground Start (GS) number of rings allowed before answering.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxo gs answer after** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the number of rings permitted to elapse before answering incoming calls, per ring from 1 to 65,535, (e.g., **2**) rings.

◆ Syntax Note ◆

Do not use commas when entering the fxs gs call answer after rings value (for example, **10,000** will return a syntax error message).

If caller ID is ON, then the number of rings should be greater than or equal to 2 rings.

Default:

The default value is **50** rings.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo gs answer after 50
voice signaling channel 2/2/13-24 fxo gs answer after 5
voice signaling channel 2/3/1-30 fxo gs answer after 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs loop current debounce

Command Usage

Specify Foreign Exchange Office (FXO) Ground Start (GS) debounce interval for loop current detector.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **fxo gs loop current debounce** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the debounce (delay interval) for debouncing the loop current detector, in milliseconds from 1 to 1,000, (e.g., **20**).

◆ Syntax Note ◆

Do not use commas when entering the fxs gs debounce interval value for loop current detection (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 fxo gs loop current debounce 20
voice signaling channel 2/2/13-24 fxo gs loop current debounce 500
voice signaling channel 2/3/1-30 fxo gs loop current debounce 1000
```

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling fxo gs battery reversal debounce

Command Usage

Specify Foreign Exchange Office (FXO) Ground Start (GS) debounce interval for battery reversal detector.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> fxo gs battery reversal debounce <value>

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

- value* Specifies the time to use as a debouncer (delay interval) for debouncing the battery reversal detector, in milliseconds from 1 to 1,000, (e.g., **20**).

◆ **Syntax Note** ◆

Do not use commas when entering the fxs gs debounce value for the battery reversal detector (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

- voice signaling channel 2/1/1-12 fxo gs battery reversal debounce 20**
- voice signaling channel 2/2/13-24 fxo gs battery reversal debounce 500**
- voice signaling channel 2/3/1-30 fxo gs battery reversal debounce 1000**

Remarks

The **voice signaling protocol** command must be set to the corresponding protocol type before any commands for E&M (Wink Start, Immediate Start, and Delay Start Signaling), and FXS/FXO (Loop and Ground Start) commands will take effect.

voice signaling caller id name

Command Usage

Set outbound caller ID name (private/unavailable) to transmit.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **caller id name** { "callerIdName" | private | unavailable }

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>callerIdName</i>	Identifies the caller by name, (e.g., " gleneck_daniel "); maximum length of 15 characters. The following characters are permitted in the caller ID name: a-z, A-Z, _ , and no spaces are allowed. Name must be surrounded by quotes.
private	Sets outbound originating caller ID name that transmits to "private" (may be abbreviated to "p").
unavailable	Sets outbound originating caller ID name that transmits to "unavailable" (may be abbreviate to "o").

◆ Syntax Note ◆

The **voice coding profile caller id** command must be enabled to use this command.

Default:

The default setting is **private**.

Command Examples:

```
voice signaling channel 2/1/1-12 caller id name "vernell_b"
voice signaling channel 2/1/1-12 caller id name private
voice signaling channel 2/2/13-24 caller id name unavailable
voice signaling channel 4/2/1-1 caller id name "p"
voice signaling channel 4/3/1-1 caller id name "o"
```

Remarks

Caller ID time is automatically determined (read) from the system time on the switch. If the caller ID name or number is changed, the voice switching daughtercard is initialized with the time automatically.

voice signaling caller id number

Command Usage

Set outbound caller ID number (published/non-published) number to transmit.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **caller id number** { “callerIdNumber” | private | unavailable}

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- startChannel* The first number in the range of voice channels (e.g., **1**).
- endChannel* The last number in the range of voice channels (e.g., **30**).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

callerIdNumber Identifies the caller by number, (e.g., **8188803500**); maximum length of ten characters (only 0-9 allowed); no _ (underscores) or spaces allowed.

private Sets outbound originating caller ID number that transmits to “private”.

unavailable Sets outbound originating caller ID number that transmits to “unavailable”.

◆ **Syntax Note** ◆

The **voice coding profile caller id** command must be enabled to use this command.

Default:

The default setting is **private**.

Command Examples:

- voice signaling channel 2/1/1-12 caller id number “8188803500”**
- voice signaling channel 2/1/1-12 caller id number “8188803500” private**
- voice signaling channel 2/2/13-24 caller id number 8188803501” unavailable**

Remarks

Caller ID time is automatically determined (read) from the system time on the switch. If the caller ID name or number is changed, the voice switching daughtercard is initialized with the time automatically.

voice signaling tone table

Command Usage

Set outbound tone table (ringing/silence) for telephony channel identifier (TCID).

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> **tone table** {**ringing** | **silence**}

Definitions:

<i>slot</i>	Specifies slot number of voice daughtercard installed in switching module, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

ringing Sets tone table for TCID to ringing (for “normal” ringback sound).

silence Sets tone table for TCID to silence (for “silent” ringback).

Default:

The default setting is **ringing**.

Command Examples:

voice signaling channel 2/1/1-12 tone table ringing
voice signaling channel 2/2/13-24 tone table silence

voice signaling call progress tone

Command Usage

Set call signaling for detection of call progress tones (on/off/relative).

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> call progress tone {on | off | relative}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON call progress tone detection for specified telephony signaling channel.
off	Turns OFF call progress tones detection for specified telephony signaling channel.
relative	Turns call progress tone detection ON or OFF according to the “call progress tone detection” parameter in the currently loaded coding profile.

Default:

The default setting is **off**.

Command Examples:

voice signaling channel 2/1/1-12 call progress tone off
voice signaling channel 2/2/13-24 call progress tone on

voice signaling call progress tone detection configuration

Command Usage

Set call signaling for call progress tone detection configuration (default/alternate).

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> call progress tone detection configuration {default | alternate}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

default	Use default call progress tone detection configuration for specified telephony signaling channel.
alternate	Use alternate call progress tone detection configuration for specified telephony signaling channel.

Default:

The default setting is **default**.

Command Examples:

voice signaling channel 2/1/1-12 call progress tone detection configuration default
voice signaling channel 2/2/13-24 call progress tone detection configuration alternate

Remarks

This command is used to specify which configuration to use for call progress tone detection. Each configuration contains filter configuration information (threshold and filter coefficients), and a table containing cadence information of all the call progress tones that need to be detected. Each configuration, whether default or alternate has a filter configuration for dial tone, ring back, including three supported cadences, bust, and congestion.

If the alternative tone detection configuration is selected, the tone detection process is limited to a busy tone, and other detected tones are ignored.

voice signaling v.18 tone detection threshold hang time

Command Usage

Set V.18 Annex A signal duration threshold for tone detection.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **v.18 tone detection threshold hang time** <value >

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the signal duration threshold for V.18 tone detection, in milliseconds from 5 to 32767, (e.g., 20).
--------------	---

◆ Syntax Note ◆

Do not use commas when entering the V.18 tone detection threshold hang time value (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Examples:

```
voice signaling channel 2/1/1-12 v.18 tone detection threshold hang time 50
voice signaling channel 2/2/13-24 v.18 tone detection threshold hang time 10000
voice signaling channel 2/3/1-30 v.18 tone detection threshold hang time 32767
```

Remarks

V.18 threshold commands are used to specify minimum detectable signal values, and are especially geared toward improving telecommunications for the hearing or speech-impaired; for example, various thresholds are available to control duration (hang time) and strength (level) or even incremental (fractional) signaling.

voice signaling v.18 tone detection threshold level

Command Usage

Set V.18 Annex A signal strength threshold level for tone detection.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **v.18 tone detection threshold level** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the signal duration threshold for V.18 tone detection, in dBm0 (decibels below 0 milliwatts) for output with no input power, from **-50** to **-15**, (e.g., **-50**).

Default:

The default value is **-40** dBm0.

Command Examples:

voice signaling channel 2/1/1-12 v.18 tone detection threshold level -40
voice signaling channel 2/2/13-24 v.18 tone detection threshold level -50

Remarks

V.18 threshold commands are used to specify minimum detectable signal values, and are especially geared toward improving telecommunications for the hearing or speech-impaired; for example, various thresholds are available to control duration (hang time) and strength (level) or even incremental (fractional) signaling.

voice signaling v.18 tone detection threshold fraction

Command Usage

Set V.18 Annex A signal strength threshold fraction for tone detection.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **v.18 tone detection threshold fraction** <value>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the signal fraction threshold for V.18 tone detection, in dBm0 (decibels below 0 milliwatt; output with no input power), 1 to 32767, (e.g., **10**).

◆ Syntax Note ◆

Do not use commas when entering the V.18 tone detection threshold fraction value (for example, **1,000** will return a syntax error message).

Default:

The default value is **10** dBm0.

Command Examples:

voice signaling channel 2/1/1-12 v.18 tone detection threshold fraction 10
voice signaling channel 2/2/13-24 v.18 tone detection threshold fraction 10000
voice signaling channel 2/3/1-30 v.18 tone detection threshold fraction 32767

Remarks

V.18 threshold commands are used to specify minimum detectable signal values, and are especially geared toward improving telecommunications for the hearing or speech-impaired; for example, various thresholds are available to control duration (hang time) and strength (level) or even incremental (fractional) signaling.

voice signaling single frequency tone detection threshold level

Command Usage

Set signal strength threshold level for single (frequency) tone detection.

Syntax Options

voice signaling channel *<slot/port/startChannel-endChannel>* **single frequency tone detection threshold level** *<value>*

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the signal frequency threshold level for tone detection, in dBm (decibels below 1 milliwatt; output signal power referenced to 1 milliwatt input signal power), from -50 to -15 , (e.g., -40).
--------------	--

Default:

The default value is **-40** dBm.

Command Examples:

voice signaling channel 2/1/1-12 single frequency tone detection threshold level -40
voice signaling channel 2/2/13-24 single frequency tone detection threshold level -50

voice signaling single frequency tone detection threshold time

Command Usage

Set signal strength threshold duration for single (frequency) tone detection.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **single frequency tone detection threshold time** <value>

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ **Syntax Note** ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the signal threshold time for single frequency tone detection, in milliseconds from 0 to 65535, (e.g., 50).
--------------	---

◆ **Syntax Note** ◆

Do not use commas when entering the single frequency tone detection threshold time (for example, **20,000** will return a syntax error message).

Default:

The default value is **50** milliseconds.

Command Examples:

voice signaling channel 2/1/1-12 single frequency tone detection threshold time 50
voice signaling channel 2/2/13-24 single frequency tone detection threshold time 20000
voice signaling channel 2/3/1-30 single frequency tone detection threshold time 65535

voice signaling echo canceller non-linear sensitivity

Command Usage

Set echo canceller processor non-linear signal sensitivity.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **echo canceller non-linear sensitivity** <value>

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).
◆ Syntax Note ◆	
	Be sure to separate the start and end range numbers with a hyphen (e.g., 1-30).
<i>value</i>	Specifies the echo canceller non-linear sensitivity, in milliseconds from 0 to 32767, (e.g., 327).

◆ Syntax Note ◆

Do not use commas when entering the echo canceller non-linear sensitivity value (for example, **20,000** will return a syntax error message).

Default:

The default value is **327** milliseconds.

Command Example:

```
voice signaling channel 2/1/1-12 echo canceller non-linear sensitivity 327
voice signaling channel 2/2/13-24 echo canceller non-linear sensitivity 20000
voice signaling channel 2/3/1-30 echo canceller non-linear sensitivity 65535
```

Remarks

Echo cancellers handle signal transmission echoes by isolating and filtering signals. Non-linear sensitivity echo cancellers are used to adjust the output and input amplitudes of a signal, and function as comfort noise generators.

voice signaling acoustic echo canceller mode

Command Usage

Set acoustic echo canceller processor mode (on/off).

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> **acoustic echo canceller mode** {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON acoustic echo canceller at channel startup for specified telephony signaling channel.
off	Turns OFF acoustic echo canceller at channel startup for specified telephony signaling channel.

Default:

The default setting is **off**.

Command Example:

voice signaling channel 2/1/1-12 acoustic echo canceller mode off
voice signaling channel 2/2/13-24 acoustic echo canceller mode on

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling acoustic echo canceller non-linear processor

Command Usage

Set acoustic echo canceller non-linear processor mode (on/off).

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> **acoustic echo canceller non-linear processor** {**on** | **off**}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

on	Turns ON non-linear acoustic echo canceller for specified telephony signaling channel.
off	Turns OFF non-linear acoustic echo canceller for specified telephony signaling channel.

◆ Syntax Note ◆

To use this command, the acoustic echo canceller must be enabled via the **voice signaling acoustic echo canceller non-linear processor** command.

Default:

The default setting is **off**.

Command Example:

```
voice signaling channel 2/1/1-12 acoustic echo canceller non-linear processor off
voice signaling channel 2/2/13-24 acoustic echo canceller non-linear processor on
```

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling acoustic echo canceller output

Command Usage

Set acoustic echo canceller processor output (handset/hands free).

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> **acoustic echo canceller output**
{**handset** | **hands free**}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

handset	Set acoustic echo canceller output to handset for specified telephony signaling channel.
hands free	Set acoustic echo canceller output to hands free for specified telephony signaling channel.

◆ Syntax Note ◆

To use this command, the acoustic echo canceller must be enabled via the **voice signaling acoustic echo canceller non-linear processor** command.

Default:

The default setting is **hands free**.

Command Example:

```
voice signaling channel 2/1/1-12 acoustic echo canceller output handset
voice signaling channel 2/2/13-24 acoustic echo canceller output hands free
```

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling acoustic echo canceller handset speaker gain

Command Usage

Set acoustic voice echo canceller processor handset speaker gain.

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> **acoustic echo canceller handset speaker gain** <*value*>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

value Specifies the acoustic echo canceller handset speaker gain, in milliseconds from 0 to 31, (e.g., **10**).

◆ Syntax Note ◆

To use this command, the acoustic echo canceller must be enabled via the **voice signaling acoustic echo canceller non-linear processor** command, and the **acoustic echo canceller output** command must be set to handset.

Default:

The default value is **10** milliseconds.

Command Example:

```
voice signaling channel 2/1/1-12 acoustic echo canceller handset speaker gain 10
voice signaling channel 2/2/13-24 acoustic echo canceller handset speaker gain 31
```

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling acoustic echo canceller hands free speaker gain

Command Usage

Set acoustic echo canceller hands free speaker gain.

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> **acoustic echo canceller hands free speaker gain** <*value*>

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

<i>value</i>	Specifies the acoustic echo canceller hands free speaker gain, in milliseconds from 0 to 31, (e.g., 10).
--------------	--

◆ Syntax Note ◆

To use this command, the acoustic echo canceller must be enabled via the **voice signaling acoustic echo canceller non-linear processor** command, and the **acoustic echo canceller output** command must be set to hands free.

Default:

The default value is **10** milliseconds.

Command Example:

```
voice signaling channel 2/1/1-12 acoustic echo canceller hands free speaker gain 10
voice signaling channel 2/2/13-24 acoustic echo canceller hands free speaker gain 31
```

Remarks

Acoustic echo cancellers handle signal transmission echoes, on calls originating from or being sent to IP telephones, by isolating and filtering the signals. Acoustic echo cancellers function as comfort noise generators.

voice signaling override in band call progress tones

Command Usage

Override call signaling for detection of call progress tones (on/off/relative). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> [no] override in band call progress tones {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Optional command syntax.

◆ Syntax Note ◆

If **no** is used then ON and OFF cannot be specified.

on Turns ON passing of in band call progress tones as voice data (after net-connect and before tele-connect state) for specified telephony signaling channel.

off Turns OFF passing of in band call progress tone as voice data (after net-connect and before tele-connect state) for specified telephony signaling channel.

Default:

The default setting is **no override in band call progress tones**.

Command Examples:

voice signaling channel 2/1/1-12 no override in band call progress tones
voice signaling channel 2/2/13-24 override in band call progress tones on

voice signaling override full call progress tones

Command Usage

Override call signaling for call progress tone detection configuration (default/alternate). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel>[no] override full call progress tones [on | off]

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Optional command syntax.

◆ Syntax Note ◆

If **no** is used then ON and OFF cannot be specified.

on Use full call progress tones for specified telephony signaling channel.

off Use full call progress tones for specified telephony signaling channel.

Default:

The default setting is **no override full call progress tones**.

Command Examples:

voice signaling channel no override full call progress tones
voice signaling channel override full call progress tones on

Remarks

This command is used to specify which configuration to use for call progress tone detection. Each configuration contains filter information (threshold and filter coefficients), and a table containing cadence information of all the call progress tones that need to be detected. The information (call progress tones) is transferred as information type packets to a coding profile. Full call progress tones means that all available call progress tones can be used in band. Each configuration, whether default or alternate has a filter configuration for dial tone, ring back, including three supported cadences, bust, and congestion.

If the alternative tone detection configuration is selected, the tone detection process is limited to a busy tone, and other detected tones are ignored.

voice signaling override ring back

Command Usage

Override call signaling for ring back (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> [**no**] **override ring back** {**on** | **off**}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Optional command syntax.

◆ Syntax Note ◆

If **no** is used then ON and OFF cannot be specified.

on Turns ON ring back for specified telephony signaling channel.

off Turns OFF ring back for specified telephony signaling channel.

◆ Syntax Note ◆

To use this command, call signaling must be turned OFF via the **voice signaling override full call progress tones** command.

Default:

The default setting is **no override ring back**.

Command Examples:

```
voice signaling channel 2/1/1-12 no override ring back
voice signaling channel 2/2/13-24 override ring back on
voice signaling channel 2/3/1-30 override ring back off
```

Remarks

Ring back is the only call progress indication supported in the signaling band.

voice signaling override in band codec switching

Command Usage

Override call signaling for in-band codec switching (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> [**no**] **override in band codec switching** {**on** | **off**}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Optional command syntax.

◆ Syntax Note ◆

If **no** is used then ON and OFF cannot be specified.

on Turns ON in band codec switching for specified telephony signaling channel.

off Turns OFF in band codec switching for specified telephony signaling channel.

◆ Syntax Note ◆

To use this command, call signaling must be set to voice via the call signaling **voice, fax, modem, data setup** command.

Default:

The default setting is **no override in band codec switching**.

Command Examples:

```
voice signaling channel 2/1/1-12 no override in band codec switching
voice signaling channel 2/2/13-24 override in band codec switching on
voice signaling channel 2/3/1-30 override in band codec switching off
```

Remarks

In voice mode, this command enables switching from one in band codec to another by detecting changes in the payload packet type.

voice signaling override psu codec switching

Command Usage

Override call signaling for packet switch unit (PSU) codec switching (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> [no] override psu codec switching {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Optional command syntax.

◆ Syntax Note ◆

If **no** is used then ON and OFF cannot be specified.

on Turns ON PSU codec switching (after net-connect and before tele-connect state) for specified telephony signaling channel.

off Turns OFF PSU codec switching (after net-connect and before tele-connect state) for specified telephony signaling channel.

Default:

The default setting is **no override psu codec switching**.

Command Examples:

voice signaling channel 2/1/1-12 no override psu codec switching
voice signaling channel 2/2/13-24 override psu codec switching on
voice signaling channel 2/2/1-30 override psu codec switching off

voice signaling override network overlap dialing

Command Usage

Override call signaling for network overlap dialing (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> [no] override network overlap dialing {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Optional command syntax.

◆ Syntax Note ◆

If **no** is used then ON and OFF cannot be specified.

on Turns ON network overlap dialing for specified telephony signaling channel.

off Turns OFF network overlap dialing for specified telephony signaling channel.

Default:

The default setting is **no override network overlap dialing**.

Command Examples:

voice signaling channel 2/1/1-12 no override network overlap dialing
voice signaling channel 2/2/13-24 override network overlap dialing on
voice signaling channel 2/3/1-30 override network overlap dialing off

Remarks

The call moves to call progress when dialing is completed, and net-connect occurs whether or not all digits have been collected.

voice signaling override information element transport

Command Usage

Override call signaling for information element (IE) transport (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> [**no**] **override information element transport** {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Optional command syntax.

◆ Syntax Note ◆

If **no** is used then ON and OFF cannot be specified.

on Turns ON information element transport for specified telephony signaling channel.

off Turns OFF information element transport for specified telephony signaling channel.

◆ Syntax Notes ◆

If this command is set to ON, then call signaling for QSIG information must be set to OFF via the **override call signaling for qsig ie transport** command.

Default:

The default setting is **no override information element transport**.

Command Examples:

```
voice signaling channel 2/1/1-12 no override information element transport
voice signaling channel 2/2/13-24 override information element transport on
voice signaling channel 2/3/1-30 override information element transport off
```

Remarks

This command controls transport of general user-to-user Information Element (IE) packets containing data fields of information.

voice signaling override qsig information element transport

Command Usage

Override call signaling for QSIG information element (IE) transport (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling channel <slot/port/startChannel-endChannel> [no] override qsig information element transport {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Optional command syntax.

◆ Syntax Note ◆

If **no** is used then ON and OFF cannot be specified.

on Turns ON QSIG information element transport for specified telephony signaling channel.

off Turns OFF QSIG information element transport for specified telephony signaling channel.

◆ Syntax Notes ◆

If this command is set to ON, then call signaling for the Information Element must be set to OFF via the **override call signaling for ie transport** command.

Default:

The default setting is **no override qsig information element transport**.

Command Examples:

voice signaling channel 2/1/1-12 no override qsig information element transport
voice signaling channel 2/2/13-24 override qsig information element transport on
voice signaling channel 2/3/1-30 override qsig information element transport off

voice signaling override setup

Command Usage

Override call signaling for voice, fax, modem, data setup (on/off). This command should only be used under the supervision of trained personnel.

Syntax Options

voice signaling channel <*slot/port/startChannel-endChannel*> [**no**] **override** {**voice setup** | **fax setup** | **modem setup** | **data setup**} {**on** | **off**}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

no Optional command syntax.

◆ Syntax Note ◆

If **no** is used then ON and OFF cannot be specified.

on Turns ON voice, fax (hard-coded), modem or data call setup for specified telephony signaling channel.

off Turns OFF voice, fax (hard-coded), modem or data call setup for specified telephony signaling channel.

Default:

The default setting is **no override data setup**.

Command Examples:

```
voice signaling channel 2/1/1-12 no override voice setup
voice signaling channel 2/2/13-24 override voice setup on
voice signaling channel 2/3/1-30 override modem setup off
voice signaling channel 2/1/1-12 no override data setup
voice signaling channel 2/2/13-24 override fax setup on
voice signaling channel 2/3/1-30 override fax setup off
```

Coding Profiles

The commands listed and described below are used to relate Coding Profiles to channels and configure the following associated components and functions: codecs, caller ID, voice mode parameters, voice network buffers, voice activity detection, tone detection, echo canceller, facsimile modem, facsimile T.38 mode, and silence detection.

Create Coding Profile

Delete Coding Profile

View Coding Profile

Reset all Coding Profiles to Factory Defaults

Relate to Channels

voice channel coding profile

preferred coding profile (voice, fax, modem, data) for calls on specified voice channel (optional)

General Parameters

codec type for coding profile

Voice Mode Parameters

coding profile voice packet interval size and field information size

Voice Network Buffer

coding profile buffer mode (adaptive/static)

coding profile nominal delay buffer

coding profile maximum delay buffer

Voice Activity Detector (VAD)

coding profile voice activity detector (on/off)

coding profile VAD threshold mode (adaptive/relative)

coding profile VAD audio threshold level (adaptive/relative; adaptive if threshold mode relative)

Tone Detection

coding profile voice DTMF relay (on/off)
coding profile fax modem switchover (enable/disable)
coding profile call progress tone detection (on/off)
coding profile V.18 Annex A call progress tone detection (on/off)
coding profile single frequency tone detection (on/off)

Echo Canceller

coding profile voice echo canceller (on/off)
coding profile voice echo canceller non-linear processor mode (on/off)
coding profile voice echo canceller comfort noise mode (static/mode)
coding profile voice echo canceller noise level (dBm)
coding profile voice echo canceller tail delay length
coding profile voice echo canceller refresh configuration state (refresh/freeze)
coding profile voice echo canceller coefficient refresh state (on/off)

Facsimile Modem

coding profile maximum allowed fax/modem data rate
coding profile fax/modem transmit level gain
coding profile fax/modem carrier detect threshold
coding profile inactivity detection time to automatically tear down fax

Facsimile Modem (T.38 Mode)

coding profile T.38 high speed fax rate
coding profile T.38 low speed packet redundancy
coding profile T.38 high speed packet redundancy
coding profile T.38 data handling method (local/over the network)

Silence Detection

coding profile voice/fax silence detection time
coding profile voice/fax silence signal level

G.711 (A-law/Mu-law)

G.711 coding profile modem coding resampling

Caller ID

caller ID coding profile (on/off) (*Command must set to apply all other caller ID settings*).

voice coding profile

Command Usage

Create coding profile with specified name.

Syntax Options

voice coding profile < “*codingProfName*” >

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

◆ Syntax Notes ◆

See Remarks below and details of default coding This command should only be used under the supervision of trained personnel.

Defaults:

For g.729ab, the default profile name is **cp1**.
For g.711 Mu-Law, the default profile name is **cp2**.
For fax t.38, the default profile name is **cp3**.
For g.711 A-Law, the default profile name is **cp4**.
For g.723.1 63, the default profile name is **cp5**.

Command Example:

```
voice coding profile salemprof1
voice coding profile calabprof2
voice coding profile cp1
voice coding profile cp2
voice coding profile cp3
voice coding profile cp4
voice coding profile cp5
```

Remarks

Only one coding profile per codec is allowed. At least one coding profile must be created before activating the voice switching daughtercard. The maximum number of coding profiles allowed per voice switching daughtercard is 128.

Before creating a coding profile, note that there are already five default coding profiles in the master **vsmboot.asc** file used with the voice daughtercard, namely: cp1, cp2, cp3, cp4 and cp5. If only the default coding profiles are used, then there is no need to create a coding profile. All five default coding profiles are automatically made available for every single voice channel upon power up of the voice switching daughtercard. The default coding profiles each contain preconfigured CLI commands pertaining to the selected codec type. See **view voice coding profile** command for more details and screen output for the command.

There is no default coding profile for modem codecs.

The cp5 coding profile can be used with the OmniPCX by changing the VPI to 30. See the **voice coding profile voice packet interval** command for details.

voice no coding profile**Command Usage**

Delete coding profile with specified name.

Syntax Options

voice no coding profile <“*codingProfName*” >

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

Default:

None

Command Example:

voice no coding profile salemprof1

voice no coding profile calabprof2

voice no coding profile cp1

voice no coding profile cp2

voice no coding profile cp3

voice no coding profile cp4

voice no coding profile cp5

view voice coding profile

Command Usage

Display coding profile for voice channels.

Syntax Options

view voice coding profile ["*codingProfName*"]

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

Default:

None

Command Example:

```
view voice coding profile salemprof1  
view voice coding profile calabprof2  
view voice coding profile  
view voice coding profile cp5
```

Screen Output

To view the default voice coding profiles, type **view voice coding profile**, and then press **<Enter>**. A screen similar to the following displays (shown on next page).

To view a coding profile by another name, type **view coding profile** and a valid coding profile name, e.g., **view voice coding profile calabprof2**, and then press **<Enter>**.

For details on editing the contents of a voice coding profile, default or otherwise, see Chapter 5, "Setup and Installation."

default voice coding profile

Voice Coding Profile cp1

!

voice coding profile "cp1"
 voice coding profile "cp1" codec type g.729ab
 voice coding profile "cp1" voice packet interval 30
 voice coding profile "cp1" voice activity detector on
 voice coding profile "cp1" voice network delay buffer nominal delay 60
 voice coding profile "cp1" voice network delay buffer max delay 120
 voice coding profile "cp1" voice echo canceller on
 voice coding profile "cp1" voice echo canceller non linear on
 voice coding profile "cp1" voice echo canceller tail 16
 voice coding profile "cp1" voice network delay buffer mode adaptive
 voice coding profile "cp1" voice dtmf relay on
 voice coding profile "cp1" fax rate 14400
 voice coding profile "cp1" call progress tone detection on
 voice coding profile "cp1" v.18 tone detection off
 voice coding profile "cp1" single frequency tone detection on
 voice coding profile "cp1" voice activity detection threshold mode adaptive
 voice coding profile "cp1" voice echo canceller comfort noise mode static
 voice coding profile "cp1" voice comfort noise level -40
 voice coding profile "cp1" echo canceller refresh configuration refresh
 voice coding profile "cp1" echo canceller refresh state on
 voice coding profile "cp1" caller id off
 voice coding profile "cp1" switchover off

!

Voice Coding Profile cp2

!

voice coding profile "cp2"
 voice coding profile "cp2" codec type g.711 mulaw
 voice coding profile "cp2" voice packet interval 20
 voice coding profile "cp2" voice activity detector on
 voice coding profile "cp2" voice network delay buffer nominal delay 80
 voice coding profile "cp2" voice network delay buffer max delay 160
 voice coding profile "cp2" voice echo canceller on
 voice coding profile "cp2" voice echo canceller non linear on
 voice coding profile "cp2" voice echo canceller tail 16
 voice coding profile "cp2" voice network delay buffer mode adaptive
 voice coding profile "cp2" voice dtmf relay on
 voice coding profile "cp2" fax rate 14400
 voice coding profile "cp2" call progress tone detection on
 voice coding profile "cp2" v.18 tone detection off
 voice coding profile "cp2" single frequency tone detection on
 voice coding profile "cp2" voice activity detection threshold mode adaptive
 voice coding profile "cp2" voice echo canceller comfort noise mode static
 voice coding profile "cp2" voice comfort noise level -40
 voice coding profile "cp2" echo canceller refresh configuration refresh
 voice coding profile "cp2" echo canceller refresh state on
 voice coding profile "cp2" caller id off
 voice coding profile "cp2" switchover off

!

Voice Coding Profile cp3

!

```
voice coding profile "cp3"  
voice coding profile "cp3" codec type fax t.38  
voice coding profile "cp3" fax rate 14400  
voice coding profile "cp3" fax transmit level -7  
voice coding profile "cp3" fax carrier detect threshold high  
voice coding profile "cp3" fax timeout 10  
voice coding profile "cp3" fax t.38 high speed packet rate 20  
voice coding profile "cp3" fax t.38 low speed redundancy 4  
voice coding profile "cp3" fax t.38 high speed redundancy 2  
voice coding profile "cp3" fax t.38 training check field method network  
voice coding profile "cp3" silence detect level -45
```

!

Voice Coding Profile cp4

!

```
voice coding profile "cp4"  
voice coding profile "cp4" codec type g.711 alaw  
voice coding profile "cp4" voice packet interval 20  
voice coding profile "cp4" voice activity detector on  
voice coding profile "cp4" voice network delay buffer nominal delay 80  
voice coding profile "cp4" voice network delay buffer max delay 160  
voice coding profile "cp4" voice echo canceller on  
voice coding profile "cp4" voice echo canceller non linear on  
voice coding profile "cp4" voice echo canceller tail 16  
voice coding profile "cp4" voice network delay buffer mode adaptive  
voice coding profile "cp4" voice dtmf relay on  
voice coding profile "cp4" fax rate 14400  
voice coding profile "cp4" call progress tone detection on  
voice coding profile "cp4" v.18 tone detection off  
voice coding profile "cp4" single frequency tone detection on  
voice coding profile "cp4" voice activity detection threshold mode adaptive  
voice coding profile "cp4" voice echo canceller comfort noise mode static  
voice coding profile "cp4" voice comfort noise level -40  
voice coding profile "cp4" echo canceller refresh configuration refresh  
voice coding profile "cp4" echo canceller refresh state on  
voice coding profile "cp4" caller id off
```

!

Voice Coding Profile cp5

!

```
voice coding profile "cp5"  
voice coding profile "cp5" codec type g.723.1 63  
voice coding profile "cp5" voice packet interval 60  
voice coding profile "cp5" voice activity detector on  
voice coding profile "cp5" voice network delay buffer nominal delay 120  
voice coding profile "cp5" voice network delay buffer max delay 240  
voice coding profile "cp5" voice echo canceller on  
voice coding profile "cp5" voice echo canceller non linear on  
voice coding profile "cp5" voice echo canceller tail 16  
voice coding profile "cp5" voice network delay buffer mode adaptive  
voice coding profile "cp5" voice dtmf relay on  
voice coding profile "cp5" fax rate 14400  
voice coding profile "cp5" call progress tone detection on  
voice coding profile "cp5" v.18 tone detection off  
voice coding profile "cp5" single frequency tone detection on  
voice coding profile "cp5" voice activity detection threshold mode adaptive  
voice coding profile "cp5" voice echo canceller comfort noise mode static  
voice coding profile "cp5" voice comfort noise level -40  
voice coding profile "cp5" echo canceller refresh configuration refresh  
voice coding profile "cp5" echo canceller refresh state on  
voice coding profile "cp5" caller id off  
voice coding profile "cp5" switchover off
```

!

voice coding profile all reset

Command Usage

Reset *all* coding profiles for voice switching daughtercard to defaults, and delete *all* existing coding profiles at the same time. (*Not available this release.*)

Syntax Options

```
voice coding profile all reset
```

Remarks

Coding profile factory default settings are currently available only from the source code.

voice channel available coding profile

Command Usage

Relate coding profile to specified voice channel.

Syntax Options

```
voice channel <slot/port/startChannel-endChannel> [un]available coding profile
{"codingProfName" | all}
```

Definitions:

slot Specifies the chassis slot number where VSM is installed, (e.g., **2**).

startChannel The first number in the range of voice channels (e.g., **1**).

endChannel The last number in the range of voice channels (e.g., **30**).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

all Indicates channel uses all available coding profiles. (*Not available this release.*)

Default:

None

Command Example:

```
voice channel 2/1/1-12 unavailable coding profile salemprof1
voice channel 2/2/13-24 unavailable coding profile calabprof2
voice channel 2/3/1-30 available coding profile cp1
voice channel 2/4 /1-12 available coding profile calabprof2
voice channel 3/1/13-24 unavailable coding profile all
voice channel 3/2/1-30 available coding profile cp2
```

Remarks

When “**all**” is used in the syntax of this command, it means that *all* coding profiles are marked accordingly as either available or unavailable for the specified channels.

By default, cp1 through cp5 default coding profiles are automatically made available to all VoIP channels in all ports and slots.

voice channel assign preferred coding profile

Command Usage

Specify preferred coding profile (voice, fax, modem, data) for calls on voice channel.

Syntax Options

voice channel <slot/port/startChannel-endChannel> **assign preferred** {voice | fax | modem} **coding profile** <"codingProfName">

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

voice	Assigns preferred voice coding profile to channel.
fax	Assigns preferred fax coding profile to channel.
modem	Assigns preferred modem coding profile to channel.
data	Assigns preferred data coding profile to channel.
<i>codingProfName</i>	Identifies coding profile by name, (e.g., salemprof1); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & / \ < > () [] { }
all	Indicates channel uses all preferred coding profiles.

Default:

None (see remarks below)

Command Example:

```
voice channel 2/1/1-12 assign preferred voice coding profile salemprof1
voice channel 2/2/13-24 assign preferred fax coding profile calabprof1
voice channel 2/3/1-30 assign preferred modem coding profile salemprof2
```

Remarks

Preferred coding profiles must be assigned before activating the voice switching daughtercard.

It is highly recommended that if voice calls are to be processed that this command be issued as it speeds up the process during the h.323 call setup procedure.

By default, cp5 default coding profile is automatically assigned as the preferred voice coding profile for every voice channel on the switch. (See **create coding profile** command.)

By default, cp3 default coding profile is automatically assigned as the preferred fax coding profile for every voice channel on the switch. (See **create coding profile** command.)

There is no default coding profile automatically assigned as the preferred modem coding profile; g.711 for every voice channel on the switch. (See **create coding profile** command.)

voice coding profile coding type

Command Usage

Specify codec type for coding profile. (See table of codec types on next page for more details.)

Syntax Options

```
voice coding profile <"codingProfName"> codec type <codec_type >
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

codec_type The specified codec type. Command choices include:

- **g.711 mulaw** (specifies PCM Mu-Law audio)
- **g.711 alaw** (specifies PCM A-Law audio)
- **g.723.1 53** (specifies g.723.1 5.3 Kbps speech and audio)
- **g.723.1 63** (specifies g.723.1 6.3 Kbps speech and audio)
- **g.729ab** (specifies g.729ab speech and audio (CS-CELP))
- **fax t.38** (specifies fax t.38 (real time) codec type)

Default:

The default codec type is **g.723.1 63** (also referred to as **cp5**, or default coding profile 5).

Command Example:

```
voice coding profile salemprof1 codec type pcm mulaw
voice coding profile calabprof1 codec type pcm alaw
voice coding profile salemprof2 codec type g.723.1 53
voice coding profile calabprof2 codec type fax t.38
```

Remarks

PCM stands for Pulse Code Modulation.

ADPCM stands for Adaptive Differential Pulse Code Modulation.

CS-CELP stands for Conjugate Structure Algebraic Code Excited Linear Prediction.

The H.245 (H.323) control functions allow the voice switching daughtercard to negotiate the codec type at runtime.

VoIP Commands

Use the table below to determine the VPI and VIF for the codec type selected as per the **voice coding profile codec type** command.

Codec Type	Description	Voice	Fax	Modem	Voice Packet Interval (VPI) Time Allowed	Voice Information Field (VIF) Size Reference
g.711 mulaw	G.711 PCM Mu-Law	Yes	Yes	Yes	20	640 bits
					30	1280 bits
					40	1920 bits
g.711 alaw	G.711 PCM A-Law	Yes	Yes	Yes	20	640 bits
					30	1280 bits
					40	1920 bits
g.723 53 g.723 63	G.723.1, 5.3 kbps coding G.723.1, 6.3 kbps coding	Yes	No	No	30 ms	192 bits
					60 ms	384 bits
g.729ab	G.729, Annex A, Annex B 8 kbps coding	Yes	No	No	10 ms	80 bits
					20 ms	160 bits
					30 ms	240 bits
					40 ms	320 bits
					50 ms	400 bits
					60 ms	480 bits
					70 ms	560 bits
80 ms	640 bits					
fax t.38	Fax Relay in T.38 mode	No	Yes	No	not applicable	not applicable

voice coding profile voice packet interval

Command Usage

Specify preferred coding profile voice packet interval (VPI) size and voice information field (VIF) size.

Syntax Options

```
voice coding profile <"codingProfName"> voice packet interval <packet_size >
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

packet_size Specifies voice packet interval in 10 millisecond increments. Supported values include 10, 20, 30, 40, 50, 60, 70, and 80.

Default:

The default packet size is **60**.

Command Example:

```
voice coding profile salemprof1 voice packet interval 10
voice coding profile calabprof2 voice packet interval 60
```

Remarks

This command is used to set the size of the voice information field (VIF), in bits, for a coding profile. The VIF size is derived from the specified voice packet interval and the voice coding algorithm. The voice coding algorithm must be specified before the desired voice packet interval.

Each codec type uses a different mathematical algorithm (method) to encode/decode voice (audio) data into IP packets. Any sound that is not a fax, modem, data or signaling tone frequency, including music, is considered voice.

The data or PCM stream coming into the voice channel is referred to as the voice packet interval (VPI) time of the packets coming into the voice channel. The encoder samples the voice traffic every 10 milliseconds, i.e., it determines how many milliseconds of a sample to take, e.g., 10 = 100 ms.

The codec puts the voice data into packet data form using this formula: 10 ms of data is converted into one data packet. The size of the converted packet is the voice information field (VFI) size, and what is ultimately sent over the IP network. The greater the VPI time the larger the VIF packet and the quantity of data transmitted onto the network; however, when the VPI time is shorter, more VIF packets are sent. In other words, the larger the VIF packets the better the quality, but the greater the traffic and use of bandwidth.

voice coding profile voice network delay buffer mode

Command Usage

Set coding profile buffer mode (adaptive/static).

Syntax Options

voice coding profile < "*codingProfName*" > **voice network delay buffer mode** {**adaptive** | **static**}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

adaptive Sets voice network buffer delay buffer mode for to adaptive for the specified coding profile.

static Sets voice network buffer delay mode to static for the specified coding profile.

Default:

The default setting is **static**.

Command Example:

```
voice coding profile salemprof1 voice network delay buffer mode adaptive
voice coding profile calabprof2 voice network delay buffer mode static
```

Remarks

This command is used to configure the adaptive playback function mode for coding profiles.

When the voice network delay buffer mode is set to static (adaptive playout disabled), the nominal and maximum playout values are valid.

When the voice network buffer mode is set to adaptive (adaptive playout enabled), the nominal and maximum playout values remain constant, and the DSP adjusts the nominal delay (playout point) to reflect any observed jitter.

The formula for calculating nominal delay (ND) and maximum delay (MD) for the voice playout buffer, given the specified voice packet interval (VPI) size is as follows:

$$ND / PT = k \text{ and } MD / PT = j$$

where:

k and j are integers

k is 2

j is k+2

voice coding profile voice network delay buffer nominal delay

Command Usage

Specify coding profile nominal delay.

Syntax Options

voice coding profile <“*codingProfName*”> **voice network delay buffer nominal delay** <*value*>

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

value Specifies the voice network buffer nominal delay, in milliseconds from 1 to 1000, (e.g., **30**).

◆ Syntax Notes ◆

Do not use commas when entering the voice network buffer nominal delay value, (for example, **1,000** will return a syntax error message).

The nominal delay should be at least twice the packet interval (in milliseconds); $NomDelay = k * \text{packet time}$, where $k > = 2$.

Default:

The default value is **120** milliseconds.

Command Example:

```
voice coding profile salemprof1 voice network buffer delay nominal delay 120
voice coding profile salemprof2 voice network buffer delay nominal delay 500
voice coding profile calabprof1 voice network buffer delay nominal delay 750
voice coding profile calabprof2 voice network buffer delay nominal delay 1000
```

voice coding profile voice network delay buffer max delay

Command Usage

Specify coding profile maximum delay.

Syntax Options

voice coding profile <“codingProfName”> voice network delay buffer max[imum] delay <value >

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

imum Optional command syntax. Type either **max** or **maximum** in the command line.

value Specifies the voice network buffer maximum delay, in milliseconds from 1 to 1000, (e.g., **30**). The maximum delay should be at least two times greater than the nominal delay (in milliseconds); MaxDelay = k * packet time, where k > j = 2.

◆ **Syntax Notes** ◆

Do not use commas when entering the voice network buffer maximum delay value, (for example, **1,000** will return a syntax error message).

Default:

The default value is **240** milliseconds.

Command Example:

voice coding profile salemprof1 voice network delay buffer maximum delay 240
voice coding profile salemprof2 voice network delay buffer max delay 500
voice coding profile calabprof1 voice network delay buffer max delay 1000

Remarks

Maximum delays must not be greater than the values shown in the following table.

Coding Profile Voice Network Buffer	
Codec	Maximum Delay
G.711 64 kbps	160 ms
G.729ab	500 ms
G.723.1	500 ms

voice coding profile voice activity detector

Command Usage

Set coding profile for voice activity detector (on/off).

Syntax Options

```
voice coding profile <"codingProfName"> voice activity detector {on | off}
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON voice activity detector for specified coding profile.

off Turns OFF voice activity detector for specified coding profile.

Default:

The default setting is **on**.

Command Example:

```
voice coding profile salemprof1 voice activity detector on
voice coding profile calabprof2 voice activity detector off
```

voice coding profile voice activity detection threshold mode

Command Usage

Set coding profile voice activity threshold mode (adaptive/relative).

Syntax Options

voice coding profile <“*codingProfName*”> **voice activity detection threshold mode**
{**adaptive** | **relative**}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

adaptive Indicates adaptive voice activity detection audio threshold mode for specified coding profile.

relative Indicates relative voice activity detection audio threshold mode for specified coding profile.

◆ Syntax Notes ◆

To use this command, the voice activity detector mode must be enabled via the **voice coding profile voice activity detector** command.

If this command is adaptive in the switch, the threshold value is automatically 32767 dBm.

If this command is relative in the switch, the voice activity detection has no bearing on the hardware or VoIP.

Default:

The default setting is **adaptive**.

Command Example:

voice coding profile salemprof1 voice activity detection threshold mode adaptive
voice coding profile calabprof2 voice activity detection threshold mode relative

Remarks

This command is used to set the audio threshold level (in dBm) for the voice activity detector (VAD) for a coding profile to be adaptive or relative to a reference level of -30 dBm.

voice coding profile voice activity detection threshold level

Command Usage

Specify coding profile VAD audio threshold level (adaptive/relative; adaptive if threshold mode enabled).

Syntax Options

```
voice coding profile <"codingProfName"> voice activity detection threshold level <threshold_level>
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

threshold_value Specifies voice activity detection threshold level for specified coding profile in dBm (decibels below 1 milliwatt; output signal power referenced to 1 milliwatt input signal power). Values may range from -20 through 20 (e.g., **-13**, **-2**, **0**, **4**, **13**, etc).

◆ Syntax Notes ◆

To use this command, the voice activity detector mode must be enabled via the **voice coding profile voice activity detector** command.

This command is only valid when the VAD mode is set to relative; if the VAD mode is adaptive, then this command is ignored.

If the supervisory disconnect connection command is turned OFF in the switch, this value is automatically 32767 dBm.

Default:

The default threshold level is **-13**.

Command Example:

```
voice coding profile salemprof1 voice activity detection threshold level -13
voice coding profile salemprof2 voice activity detection threshold level 0
voice coding profile calabprof1 voice activity detection threshold level 20
voice coding profile calabprof2 voice activity detection threshold level 1
```

Remarks

This command is used to set the audio threshold level (in dBm) for the voice activity detector (VAD) for a coding profile to be adaptive or relative to a reference level of -30 dBm.

voice coding profile voice dtmf relay

Command Usage

Set voice coding profile for Dual Tone Multi-Frequency (DTMF) relay (on/off).

Syntax Options

voice coding profile < “codingProfName” > voice dtmf relay {on | off}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON voice DTMF for specified coding profile. When using G.711 codec type, DTMF relay must be turned ON.

off Turns OFF voice DTMF for specified coding profile. When using G.723.1 and G.729a codecs, DTMF relay must be turned OFF.

◆ **Syntax Notes** ◆

This command is only valid with RTP (Realtime Transport Protocol) encapsulation.

Default:

The default setting is **on**.

Command Example:

voice coding profile salemprof1 voice dtmf relay on
voice coding profile calabprof2 voice dtmf relay off

Remarks

DTMF tones are detected during voice processing and separately packetized for transmission.

voice coding profile switchover

Command Usage

Set coding profile fax/modem switchover (enable/disable).

Syntax Options

voice coding profile < "*codingProfName*" > **switchover** {on | off}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON fax/modem switchover for specified coding profile.

off Turns OFF fax/modem switchover for specified coding profile.

◆ Syntax Notes ◆

If the coding type is set to either PCM Mu-Law or PCM A-Law via the **voice coding profile codec type** command, this parameter can be ON or OFF.

If the coding type is set to fax T.38 this command should be set to OFF, because no switchover is required.

Default:

The default setting is **off**.

Command Example:

```
voice coding profile "cp1" switchover on
voice coding profile "cp2" switchover on
voice coding profile "cp3" switchover off
voice coding profile "cp4" switchover on
voice coding profile "cp5" switchover on
```

Remarks

DSP tone detection on the voice channel must be ON in the specified coding profile if fax modem switchover is desired, as switchover relies on tone detection. Switchover can be either ON or OFF for a fax-only coding profile.

voice coding profile call progress tone detection

Command Usage

Set coding profile call progress tone detection (on/off).

Syntax Options

voice coding profile <“*codingProfName*”> **call progress tone detection** {**on** | **off**}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON secondary level of control over call progress tone detection for specified coding profile.

off Turns OFF secondary level of control over call progress tone detection for specified coding profile.

◆ **Syntax Notes** ◆

If the call progress detection control for a channel is set to relative via the **call progress tone detection** command, then this parameter determines whether or not detection is enabled.

Default:

The default setting is **off**.

Command Example:

voice coding profile salemprof1 call progress tone detection off
voice coding profile calabprof2 call progress tone detection on

voice coding profile voice dtmf relay

Command Usage

Set voice coding profile for V.18 Annex A call progress tone detection (on/off).

Syntax Options

```
voice coding profile <"codingProfName"> voice dtmf relay {on | off}
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON V.18 Annex A call progress tone detection for specified coding profile.

off Turns OFF V.18 Annex A call progress tone detection for specified coding profile.

Default:

The default setting is **off**.

Command Example:

```
voice coding profile salemprof1 voice dtmf relay off
voice coding profile calabprof2 voice dtmf relay on
```

Remarks

V.18 Annex A is a 1400 hz. tone used for channel configuration that is detected for 100 ms.

For more information on V.18 Annex A refer to the V.18 Annex A threshold commands.

voice coding profile single frequency tone detection

Command Usage

Set coding profile single frequency call progress tone detection (on/off).

Syntax Options

voice coding profile <“codingProfName”> single frequency tone detection {on | off}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON single frequency call progress tone detection control.

off Turns OFF single frequency call progress tone detection control.

Default:

The default setting is **off**.

Command Examples:

voice coding profile salemprof1 single frequency tone detection off

voice coding profile calapprof2 single frequency tone detection off

Remarks

The DSPs on the voice switching daughtercard support 2600 hz. tone detection.

voice coding profile voice echo canceller

Command Usage

Set coding profile voice echo canceller (on/off).

Syntax Options

```
voice coding profile <"codingProfName"> voice echo canceller {on | off}
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON voice echo canceller mode for specified coding profile.

off Turns OFF voice echo canceller mode for specified coding profile.

Default:

The default setting is **on**.

Command Example:

```
voice coding profile salemprof1 voice echo canceller on
voice coding profile calapprof2 voice echo canceller off
```

Remarks

Voice switching daughtercards perform echo removal on PCM samples using a proprietary double filter algorithm that provides stability and performance up to 128 ms echo cancellation tail length. See also ITU-T Recommendation G.165: Echo Cancellers.

For more information on echo cancellers, refer to the echo and acoustic echo cancellation commands used for Telephony Signaling.

voice coding profile voice echo canceller non linear

Command Usage

Set coding profile voice echo canceller non-linear processor mode (on/off).

Syntax Options

voice coding profile <“codingProfName”> voice echo canceller non linear {on | off}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON non-linear voice echo canceller processor mode for specified coding profile.

off Turns OFF non-linear voice echo canceller processor mode for specified coding profile.

◆ **Syntax Notes** ◆

To use this command, the voice echo canceller mode must be enabled via the **coding profile voice echo canceller** command.

Default:

The default setting is **on**.

Command Example:

voice coding profile salemprof1 voice echo canceller non linear on
voice coding profile calabprof2 voice echo canceller non linear off

voice coding profile voice echo canceller comfort noise mode

Command Usage

Specify coding profile voice echo canceller comfort noise mode.

Syntax Options

```
voice coding profile < "codingProfName" > voice echo canceller comfort noise mode
{static | adaptive}
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

static Turns ON the fixed voice comfort noise level.

adaptive Turns OFF the fixed voice comfort noise level at runtime for the duration of the call, if it is determined that the phone is digital (optional selection).

◆ Syntax Notes ◆

To use this command, the voice echo canceller mode must be enabled via the **coding profile voice echo canceller** command.

Default:

The default setting is **static**.

Command Example:

```
voice coding profile salemprof1 voice echo canceller comfort noise mode static
voice coding profile calabprof2 voice echo canceller comfort noise mode adaptive
```

voice coding profile echo canceller noise level

Command Usage

Specify coding profile echo canceller noise level.

Syntax Options

voice coding profile <“codingProfName”> echo canceller noise level <value >

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

value Specifies voice echo canceller noise level for specified coding profile, in dBm, (power referenced to 1 milliwatt input signal power), from **-70** to **-40**, (e.g., **-40**).

◆ Syntax Notes ◆

To use this command, the voice echo canceller mode must be enabled via the **coding profile voice echo canceller** command. The voice echo canceller comfort noise mode must also be set to static via the **coding profile voice echo canceller comfort noise mode** command.

Default:

The default value is **-40**.

Command Example:

voice coding profile salemprof1 echo canceller noise level -40
voice coding profile calabprof2 echo canceller noise level -70

voice coding profile voice echo canceller tail length

Command Usage

Specify coding profile voice echo canceller tail delay length.

Syntax Options

voice coding profile <“*codingProfName*”> **voice echo canceller tail length** <*value*>

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

value Specifies voice echo canceller tail length for specified coding profile, in milliseconds, from **0** to **16**, (e.g., **16**); **4**, **8**, **16** are the only legal values for this release.

◆ Syntax Notes ◆

To use this command, the voice echo canceller mode must be enabled via the **coding profile voice echo canceller** command.

Default:

The default value is **16**.

Command Example:

```
voice coding profile salemprof1 voice echo canceller tail length 16
voice coding profile calabprof2 voice echo canceller tail length 0
voice coding profile salemprof1 voice echo canceller tail length 128
```

Remarks

voice coding profile echo canceller refresh configuration

Command Usage

Specify coding profile echo canceller refresh configuration state.

Syntax Options

```
voice coding profile <“codingProfName”> echo canceller refresh configuration  
{frozen | refresh}
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

frozen Specifies voice echo canceller configuration refresh state for specified coding profile.

refresh Specifies voice echo canceller configuration refresh state is frozen for specified coding profile.

Default:

The default setting is **frozen**.

Command Example:

```
voice coding profile salemprof1 echo canceller refresh configuration frozen  
voice coding profile calabprof2 echo canceller refresh configuration refresh
```

Remarks

In the configuration refresh state the echo canceller automatically adjusts as the call progresses based upon the PCM sample.

voice coding profile echo canceller refresh state

Command Usage

Specify coding profile echo refresh state.

Syntax Options

voice coding profile <“*codingProfName*”> echo canceller refresh state {on | off}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON (resets) voice echo canceller refresh state for specified coding profile.

off Turns OFF voice echo canceller refresh state for specified coding profile (normal state).

Default:

The default setting is **on**.

Command Example:

```
voice coding profile salemprof1 echo canceller refresh state on
voice coding profile calabprof2 echo canceller refresh state off
```

voice coding profile fax rate

Command Usage

Specify coding profile maximum allowed fax modem data rate.

Syntax Options

voice coding profile <“codingProfName”> fax rate {2400 | 4800 | 7200 | 9600 | 12000 | 14400}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

2400 Specifies a maximum fax data baud rate of 2400 bps.

4800 Specifies a maximum fax data baud rate of 4800 bps.

7200 Specifies a maximum fax data baud rate of 7200 bps.

9600 Specifies a maximum fax data baud rate of 9600 bps.

12000 Specifies a maximum fax data baud rate of 12000 bps.

14400 Specifies a maximum fax data baud rate of 14400 bps.

Default:

The default baud rate is **14400**.

Command Example:

voice coding profile salemprof1 fax rate 14400

voice coding profile calabprof2 fax rate 9600

voice coding profile fax transmit level**Command Usage**

Specify coding profile fax modem transmit level gain.

Syntax Options

```
voice coding profile <"codingProfName"> fax transmit level <gain_value >
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

gain_value Specifies fax modem transmit (TX) gain for specified voice coding profile, in decibels. Values may range from -13 through 0 (e.g., **-13**, **-2**, etc).

Default:

The default value is **-13** decibels.

Command Example:

```
voice coding profile salemprof1 fax transmit level -13  
voice coding profile calabprof2 fax transmit level 0
```

voice coding profile fax carrier detect threshold

Command Usage

Specify coding profile fax modem carrier detect threshold.

Syntax Options

```
voice coding profile <“codingProfName”> fax carrier detect threshold {low | medium | high}
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

low Sets fax modem carrier detect threshold to -43 dBm for specified coding profile.

medium Sets fax modem carrier detect threshold to -33 dBm for specified coding profile.

high Sets fax modem carrier detect threshold to -26 dBm for specified coding profile.

Default:

The default setting is **high**.

Command Example:

```
voice coding profile salemprof1 fax carrier detect threshold high  
voice coding profile salemprof2 fax carrier detect threshold medium  
voice coding profile calabprof1 fax carrier detect threshold low
```

voice coding profile fax timeout

Command Usage

Specify coding profile inactivity detection time to automatically tear down fax.

Syntax Options

```
voice coding profile <"codingProfName"> fax timeout <value >
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

value Specifies voice coding profile fax timeout (no activity time) on fax modem connection before call is cleared, in milliseconds, from 10 to 32,000 seconds, (e.g., **20**).

◆ Syntax Note ◆

Do not use commas when entering the voice coding profile fax timeout value, (for example, **1,000** will return a syntax error message).

Default:

The default value is **20** milliseconds.

Command Example:

```
voice coding profile salemprof1 fax time out value 20
voice coding profile salemprof2 fax time out value 12000
voice coding profile calabprof2 fax time out value 32000
```

voice coding profile fax t.38 high speed packet rate

Command Usage

Specify coding profile T.38 high speed fax rate.

Syntax Options

voice coding profile <“codingProfName”> fax t.38 high speed packet rate {10 | 20 | 30 | 40}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

10 | 20 | 30 | 40 Specifies rate at which high speed data is sent across network for specified fax coding profile.

Default:

The default fax rate is **20**.

Command Example:

voice coding profile salemprof1 fax t.38 high speed packet rate 20
voice coding profile calabprof2 fax t.38 high speed packet rate 40

voice coding profile fax t.38 low speed redundancy

Command Usage

Specify coding profile T.38 low speed packet redundancy.

Syntax Options

voice coding profile <“*codingProfName*”> **fax t.38 low speed redundancy** {0 | 1 | 3 | 4 | 5}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

0 | 1 | 3 | 4 | 5 Specifies packet-level redundancy for low speed data transmission (i.e., T.30 handshaking info) for specified fax coding profile.

Default:

The default value is **4**.

Command Example:

```
voice coding profile salemprof1 fax t.38 low speed redundancy 4
voice coding profile calabprof2 fax t.38 low speed redundancy 5
```

voice coding profile fax t.38 high speed redundancy

Command Usage

Specify coding T.38 high speed packet redundancy.

Syntax Options

voice coding profile <“codingProfName”> fax t.38 high speed redundancy {0 | 1 | 2}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

0 | 1 | 2 Specifies packet-level redundancy for high speed data transmission (i.e., T.4 image data) for specified fax coding profile.

Default:

The default value is **2**.

Command Example:

voice coding profile salemprof1 fax t.38 high speed redundancy 1
voice coding profile calabprof2 fax t.38 high speed redundancy 2

voice coding profile fax t.38 training check field method

Command Usage

Specify coding profile T.38 data handling method (local/over the network).

Syntax Options

voice coding profile <“codingProfName”> fax t.38 training check field method {local | network}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

local Specifies local method of handling data over the network for specified coding profile.

network Turns network method of handling data over the network for specified coding profile.

Default:

The default setting is **network**.

Command Example:

```
voice coding profile salemprof1 fax t.38 training check field method local
voice coding profile calabprof2 fax t.38 training check field method network
```

Remarks

The local method (method 1) requires that the training check field (TCF) training signal be generated and checked locally by the gateway, and not be forwarded over the network. With the network method (method 2), TCF data is sent over the network. Both methods correspond to data management methods 1 and 2 in the T.38 UDP fax protocol specification.

voice coding profile silence detect time

Command Usage

Specify voice/fax coding profile silence detection time.

Syntax Options

voice coding profile <“*codingProfName*”> [**no**] **silence detect time** <*value*>

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

value Specifies voice/fax coding profile silence detection time, in milliseconds, from 5 to 32,000, (e.g., **2000**).

◆ Syntax Notes ◆

Do not use commas when entering the value for voice/fax coding profile silence detection time, (for example, **2,000** will return a syntax error message).

no Disables silence detection. A “no silence detect time 100” or any other number of milliseconds disables silence detection.

Default:

The default value is **5** milliseconds.

Command Example:

```
voice coding profile salemprof1 no silence detect time
voice coding profile salemprof1 silence detect time value 100
voice coding profile calabprof2 silence detect time value 100
```

voice coding profile silence detect level

Command Usage

Specify voice/fax coding profile silence signal level.

Syntax Options

```
voice coding profile <"codingProfName"> silence detect level <signal_level>
```

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

signal_level Specifies voice/fax coding profile silence signal level, in decibels. Values may range from -50 through -40 (e.g., **-42**, **-50**, **-40**, etc).

Default:

The default value is **-50** decibels.

Command Example:

```
voice coding profile salemprof1 silence detect level -50
voice coding profile calabprof2 silence detect level -40
```

voice coding profile g.711 modem resampling mode

Command Usage

Specify g.711 (PCM Mu-Law/PCM A-Law) modem coding resampling.

Syntax Options

voice coding profile <“codingProfName”> g.711 modem resampling mode {on | off}

Definitions:

codingProfName Identifies coding profile by name, (e.g., **salemprof1**); maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON g.711 (Mu-Law/A-Law) modem resampling mode for specified coding profile.

off Turns OFF g.711 (Mu-Law/A-Law) modem resampling mode for specified coding profile.

◆ Syntax Notes ◆

The following settings must be made to use this command:

The voice coding profile codec type must be set to g.711 via the **voice coding profile codec type** command.

The voice coding profile adaptive playout delay must be disabled via the **voice network delay buffer mode** command.

The voice coding profile maximum network buffer delay must be set to maximum via the **voice network delay buffer maximum delay** command.

The voice coding profile nominal network buffer delay must be set to half of the maximum buffer delay via the **network delay buffer nominal delay** command.

The voice activity detection must be disabled via the **voice coding profile voice activity detector** command.

Default:

The default setting is **off**.

Command Example:

```
voice coding profile salemprof1 g.711 modem resampling on
voice coding profile calabprof2 g.711 modem resampling off
```

voice coding profile caller id

Command Usage

Set caller ID for specified coding profile (on/off); this command must be set to apply all other caller ID settings as listed:

Telephony Signaling

FXS LS to generate outbound caller ID (on/off)

FXS LS to detect inbound caller ID (on/off)

FXO GS to generate outbound caller ID (on/off)

FXO GS to detect inbound caller ID (on/off)

Outbound Caller ID

Outbound caller ID name (private/unavailable) to transmit

Outbound caller ID number (published/unpublished) to transmit

Syntax Options

voice coding profile < “codingProfName” > caller id {on | off}

Definitions:

codingProfName Identifies the coding profile by name, (e.g., **cprofcallid1**). Consists of at least one ASCII character with quotes on each end of the name; maximum length of 40 characters. The following characters are permitted in the coding profile name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON caller ID for specified coding profile.

off Turns OFF caller ID for specified coding profile.

Default:

The default setting is **on**.

Command Examples:

```
voice coding profile "cprofcallid1" caller id off
voice coding profile "cprofcallid2" caller id on
voice coding profile "cprofcallid3" caller id off
voice coding profile "cprofcallid4" caller id on
voice coding profile "cprofcallid5" caller id off
voice coding profile "cprofcallid6" caller id on
```

Voice Network

The commands listed and described below are used to assign and configure the VoIP or voice network and the following related components: H.323 gateway discovery, operations and configuration.

Talking about network portion, i.e., H.323, of a physical voice daughtercard, this is not to be confused with the voice switching daughtercard commands which deal with general daughtercard features - dialing timers, gateway mode and voice ports. Actually, Network settings for specified voice daughtercard.

H.323 Gateway Discovery

gatekeeper control (on/off)

gatekeeper discovery mode (manual/off) (Auto discovery *not available this release.*)

gatekeeper IP address for gatekeeper discovery (manual mode only)

H.323 Gateway Configuration

calls allowed (or disallowed) without gatekeeper (no gateway endpoint regis.; true/false)

no. of registration attempts allowed (before gateway endpoint registration failure)

gateway endpoint registration type (if gatekeeper used)

associate (or disassociate) phone groups with gatekeeper (if gatekeeper used)

H.323 Gateway Operations

H.323 display name for voice switching daughtercard gateway

RTP/RTCP *port mode* for voice switching daughtercard gateway (dynamic/sequential)

starting RTP/RTCP *port number* for voice switching daughtercard gateway (if sequential)

voice network card h.323 gatekeeper control**Command Usage**

Set voice switching daughtercard to use gatekeeper (on/off).

Syntax Options

```
voice network card <slot/card_number> h.323 gatekeeper control {on | off}
```

Definitions:

slot Specifies the chassis slot number where VSM is installed, (e.g., **2**).

card_number Specifies the voice daughtercard position number, (e.g., **1**).

on Turns ON H.323 gatekeeper for specified voice network.

off Turns OFF H.323 gatekeeper for specified voice network.

Default:

The default setting is **off**.

Command Example:

```
voice network card 2/1 h.323 gatekeeper control off  
voice network card 2/2 h.323 gatekeeper control on
```

Remarks

Commands containing the syntax **voice network card** differ from commands using the syntax **voice daughtercard** in that “voice network” type commands deal specifically with network related operations of a voice switching daughtercard (VoIP gateway).

voice network h.323 gatekeeper mode

Command Usage

Set gatekeeper mode for voice switching daughtercard gateway discovery (manual/auto).

Syntax Options

```
voice network card <slot/card_number> h.323 gatekeeper mode {manual | auto}
```

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
manual	Turns ON H.323 gatekeeper manual discovery mode for specified voice network.
auto	Turns ON H.323 gatekeeper autodiscovery mode for specified voice network.

◆ Syntax Note ◆

To use this command, the **voice network card h.323 gatekeeper control** command must be turned ON.

Default:

The default setting is **manual**.

Command Example:

```
voice network card 2/1 h.323 gatekeeper mode manual  
voice network card 2/2 h.323 gatekeeper mode auto
```

Remarks

This command is used to control gatekeeper operations in conjunction with the **voice network card h.323 gatekeeper control** command. Internally, the setting affects two different variables in the switch configuration. These include the gatekeeper configuration field which controls the enable/disable operation of the gatekeeper, and the auto discovery gatekeeper configuration field (when enabled), which determines automatic or manual discovery.

voice network h.323 gatekeeper address

Command Usage

Specify gatekeeper IP address for voice switching daughtercard gateway discovery (manual mode only).

Syntax Options

```
voice network card <slot/card_number> h.323 gatekeeper address <ip_address >
```

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
<i>ip_address</i>	Specifies IP address of the voice daughtercard H.323 gatekeeper, (e.g., 224.0.1.41).

◆ Syntax Notes ◆

H.323 display name string must be specified before the voice switching daughtercard is activated. A non-null string value is required.

To use this command, the **h.323 gatekeeper mode** command for gatekeeper discovery must be set to MANUAL.

Default:

None

Command Example:

```
voice network card 2/1 h.323 gatekeeper address 224.0.1.41
voice network card 2/2 h.323 gatekeeper address 224.0.1.42
```

Remarks

This command is used to specify the address of the gatekeeper in the currently active (H.323) zone when configured for manual mode. Because port 1719 is used, only the IP address needs to be specified.

voice network h.323 allow calls without gatekeeper

Command Usage

Set calls allowed (or disallowed) without gatekeeper; voice switching daughtercard gateway endpoint not registered (true/false).

Syntax Options

```
voice network card <slot/card_number> h.323 allow calls without gatekeeper {true | false}
```

Definitions:

slot Specifies the chassis slot number where VSM is installed, (e.g., **2**).

card_number Specifies the voice daughtercard position number, (e.g., **1**).

true H.323 allows calls without gatekeeper.

false H.323 does not allow calls without gatekeeper.

Default:

The default setting is **true**.

Command Example:

```
voice network card 2/1 h.323 allow calls without gatekeeper true
```

```
voice network card 2/2 h.323 allow calls without gatekeeper false
```

voice network h.323 allow calls without gatekeeper max tries

Command Usage

Specify number of registration attempts allowed before voice switching daughtercard gateway endpoint registration failure occurs.

Syntax Options

```
voice network card <slot/card_number> h.323 allow calls without gatekeeper max[imum] tries <value>
```

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
imum	Optional command syntax. Type either max or maximum in the command line.
<i>value</i>	Specifies number of registration attempts made by voice switching daughtercard before it is allowed to fail registration, (e.g., 4).

Default:

The default value is **4**.

Command Example:

```
voice network card 2/1 h.323 allows calls without gatekeeper maximum tries 1
voice network card 2/2 h.323 allows calls without gatekeeper max tries 4
```

Remarks

Once the number of unsuccessful registration attempts is passed, the endpoint is only able to place calls if the **h.323 allow calls without gatekeeper** command is set to **true**.

voice network h.323 endpoint registration type

Command Usage

Specify voice switching daughtercard gateway endpoint registration type (if gatekeeper used).

Syntax Options

voice network card <slot/card_number> h.323 endpoint registration type {gateway | terminal}

Definitions:

- slot* Specifies the chassis slot number where VSM is installed, (e.g., **2**).
- card_number* Specifies the voice daughtercard position number, (e.g., **1**).
- gateway** Enables gateway endpoint registration.
- terminal** Enables v registration.

◆ **Syntax Note** ◆

The only time this command should be set to **terminal** is when an analog voice switching daughtercard is in use, and only one phone number and one analog port are in service.

Default:

The default setting is **gateway**.

Command Example:

voice network card 2/1 h.323 endpoint registration type gateway
voice network card 2/2 h.323 endpoint registration type terminal

Remarks

This command is used to set the H.225.0 endpoint registration type of the voice switching daughtercard. This should not be confused with the H.245 terminal type, although the two parameters should be programmed consistently. This parameter specifies how the endpoint will register itself with the gatekeeper, and has nothing to do with master/slave determination. See the RADVision H.323 Gatekeeper User Manual for more information.

voice network h.323 gatekeeper associate

Command Usage

This command is used to associate or (disassociate) one or more phone groups with a daughtercard gatekeeper, thereby enabling the daughtercard to generate the legal H.323 alias names that are sent to the gatekeeper in lieu of a telephone number.

Syntax Options

```
voice network card <slot/card_number> h.323 gatekeeper [dis]associate [phone group]
<"phoneGrpName">
```

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
dis	Optional command syntax. Type either dis or disassociate in the command line.
phone group	Optional command syntax.
<i>phoneGrpName</i>	Specifies the phone group to which the gatekeeper is associated (or disassociated).

◆ Syntax Notes ◆

This command must be issued if the **voice network card h.323 gatekeeper control** command is ON.

In order for the voice daughtercard to generate alias names using this command, either the **voice daughter card activate** command or the **voice numbering plan activate** command must be issued.

This command can only be issued from the Command Line Prompt, and *not* from within the **vsmbboot.asc** file.

Default:

None

Command Example:

```
voice network card 2/1 h.323 gatekeeper associate phone group salem_engr1
voice network card 2/2 h.323 gatekeeper associate salem_engr1
voice network card 2/1 h.323 gatekeeper disassociate phone group salem_engr1
voice network card 2/2 h.323 gatekeeper dis salem_engr1
```

Remarks

Only E.164 alias names are generated by the gateway. E.164 is an ITU ISDN/SMDs (Switched Multimegabit Data Service) phone line numbering scheme; SDMS is used in LAN to LAN metropolitan networks.

voice network h.323 display name

Command Usage

Specify H.323 display name for voice switching daughtercard gateway.

Syntax Options

voice network card <slot/card_number> h.323 display name <"string">

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
<i>string</i>	Any string up to 64 bytes, (e.g., salemgway). The display name is carried as an H.323_ID alias name.

◆ Syntax Note ◆

H.323 display name string must be specified before the voice switching daughtercard is activated. A non-null string value is required.

Default:

None

Command Example:

voice network card 2/1 h.323 display name salemgway
voice network card 2/2 h.323 display name calabgway

Remarks

This command is used to set the display name information that is carried in the H.323 setup messages. The display name string is inserted into the Q.931 display information and source address field of the H.323 setup (UUIE).

voice network h.323 rtp port mode

Command Usage

Set RTP/RTCP (Real Time Protocol/Real Time Conferencing Protocol) port mode for voice switching daughtercard H.323 gateway (dynamic/sequential).

Syntax Options

```
voice network card <slot/card_number> h.323 rtp port mode {dynamic | sequential}
```

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
dynamic	Assigns RTP/RTCP port numbers dynamically for H.323 gateway.
sequential	Assigns RTP/RTCP port numbers sequentially for H.323 gateway.

◆ Syntax Notes ◆

If the starting RTP/RTCP port mode for the gateway is set to dynamic, then the RTP port base value is automatically set to **0**.

Starting RTP/RTCP port numbers are specified via the **h.323 RTP port base** command.

Default:

The default setting is **dynamic**.

Command Example:

```
voice network card 2/1 h.323 rtp port mode dynamic
voice network card 2/2 h.323 rtp port mode sequential
```

Remarks

This command is used to set the port number assignment method for RTP and RTCP ports.

voice network h.323 rtp port base

Command Usage

Specify starting RTP/RTCP port number for voice switching daughtercard gateway (if sequential).

Syntax Options

```
voice network card <slot/card_number> h.323 rtp port base <value >
```

Definitions:

<i>slot</i>	Specifies the chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
<i>value</i>	Specifies starting RTP or RTCP port value from 1 to 65535, (e.g., 30000). If dynamic port assignment is preferred, the RTP port base should be set to a value of 0 ; if dynamic port assignment is used, the requirements of the H.323 specification may not be met, as no attempt is made to assure proper numbering upon assignment.

◆ Syntax Notes ◆

Do not use commas when entering the starting value for RTP/RTCP ports, (for example, **30,000** will return a syntax error message).

To use this command, the starting RTP/RTCP port mode for the gateway must be set to sequential via the **RTP port mode** command.

If the starting RTP/RTCP port mode for the gateway is set to dynamic, then this value is ignored.

Default:

The default value is **30000**.

Command Example:

```
voice network card 2/1 h.323 rtp port base 30000
voice network card 2/2 h.323 rtp port base 65535
```

Remarks

This command is used to specify the starting port number assigned to RTP/RTCP ports; when H.323 calls are made an RTP or RTCP port is opened for each call. The RTP port number should be an even number, and the RTCP port number should be one number greater than the RTP port value.

In order to accomplish this sort of controlled allocation, the port numbers are assigned starting at the RTP port base value. Call Control Block (CCB) 0 will use ports numbered RTP port base and RTP port base + 1. CCB 1 will use the next two successive ports, etc. When the call is terminated, the CCB number will eventually be reused.

For each new H.323 call, the CCB number is incremented. The first call would be CCB 0, the second would be CCB 1 and so on, as shown below where the RTCP port number is set one number higher than the RTP port number, e.g., the starting RTP port number is 30001 (with a base of 30000), and the starting RTCP port number is 30002.

- On the first incoming H.323 call setup message, the starting RTP port number would be 30003, and the starting RTCP port number would be 30004.
- On the next incoming H.323 call setup message, the starting RTP port number would be 30004, and the starting RTCP port number would be 30005.
- On the subsequent incoming H.323 call setup message, the starting RTP port number would be 30006, and the starting RTCP port number would be 30007, and so on.

Standard port assignments are as follows:

Gatekeeper User Datagram Protocol (UDP) Discovery Multicast Address: 224.0.1.41

Gatekeeper UDP Discovery Port (automatic discovery): 1718

Gatekeeper UDP Registration and Status Port (manual discovery): 1719

Endpoint Transmission Control Protocol (TCP) Call Signaling Port: 1720

Network Dialing Scheme

The commands listed and described below are used to configure a Network Dialing Scheme and related components as follows: destinations, phone groups and phone group parameters, and numbering plans including numbering plan hunt methods and descriptions.

Destinations

- H.323 endpoint destination name
- local channel destination
- delete destination
- view destination

Phone Groups

- create (or delete) voice phone group
- view voice phone group

Inbound/Outbound Digit Processing

- unique phone group site *prefix* for routing VoIP calls (on/off)
- unique phone group site *prefix digits* for routing VoIP calls
- voice phone group dialing type
- voice phone group format of tel. number and number of outbound digits to dial

Additional Inbound/Outbound Digit Processing

- number of outbound digits to strip in voice phone group site (before forwarding call)
- allow forwarding of phone group prefix
- phone group site digits to prefix (before forwarding call) (on/off)

Digit Dialing Ranges

- voice phone group site *numbers to include* in range of digits for phone format string
- voice phone group site *numbers to remove* from range of digits for phone format string

Numbering Plan

- create (or delete) numbering plan
- view voice numbering plan
- activate voice numbering plan

Numbering Plan Hunt Method

- outgoing hunting method of voice numbering plan (destinations group)

Numbering Plan Description

voice numbering plan (optional)

Associate Numbering Plan

associate (or disassociate) *destinations* with numbering plans

associate (or disassociate) *phone groups* with numbering plans

voice destination h.323 endpoint

Command Usage

Create an endpoint destination with specified name that uses the H.323 protocol.

Syntax Options

voice destination <“*endpointDestName*”> **h.323 endpoint** <*address*> [*portAddr*]

Definitions:

<i>endpointDestName</i>	Identifies the voice call endpoint destination string name unique across the H.323 network, (e.g., to_vsd1) any string up to 64 bytes (not including quotes). The following characters are permitted in the destination string name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & / \ < > () [] { }
<i>address</i>	H.323 transport address, i.e., IP or H.323 network address, (e.g., 225.0.1.41). See voice daughter card ip address command for details on configuring H.323 network address for Alcatel VSM-based gateways.
<i>portAddr</i>	Optional command syntax. Any H.323 RTP port address, e.g., (1720).

◆ Syntax Notes ◆

H.323 display name string must be specified before the voice switching daughtercard is activated. A non-null string value is required.

To use this command, the **h.323 gatekeeper mode** command for gatekeeper discovery must be set to MANUAL. At least one endpoint (or gateway), and one H.323 transport address must be specified to make an over-the-network call.

Default:

None

Command Example:

```
voice destination to_vsd1 h.323 endpoint 225.0.1.41
voice destination to_vsd2 h.323 endpoint 225.0.1.42 1720
```

Remarks

This command can be used to specify the address of the gatekeeper in the currently active (H.323) zone when configured for manual mode. When the destination is a gatekeeper, port 1719 should be specified (default).

When port 1720 is used only the IP address needs to be specified.

The endpoint name string is the user's logical name for the remote calling party, and is automatically used to create the card level destinations, e.g., **tosalem**. The endpoint name string is sometimes referred to as an alias name.

voice destination local channel

Command Usage

Create a local channel destination with specified name.

Syntax Options

```
voice destination < "channelDestName" > local channel
< "endpointDestName" / port / startChannel - endChannel >
```

Definitions:

<i>channelDestName</i>	Identifies the voice call endpoint destination by name, (e.g., tosalem); maximum length of 40 characters. The following characters are permitted in the call destination name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & / \ < > () [] { }
<i>endpointDestName</i>	Identifies the voice call endpoint destination string name unique across the H.323 network, (e.g., to_vsd1) any string up to 64 bytes (not including quotes). The following characters are permitted in the destination string name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & / \ < > () [] { }
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>startChannel</i>	The first number in the range of voice channels (e.g., 1).
<i>endChannel</i>	The last number in the range of voice channels (e.g., 30).

◆ Syntax Note ◆

Be sure to separate the start and end range numbers with a hyphen (e.g., **1-30**).

Default:

None

Command Example:

```
voice destination tosalem local channel to_vsd1 /1/1-12
voice destination tocalab local channel to_vsd2/1/13-24
```

Remarks

This command is used to specify a local VSM-based gateway (VSD, VSA and VSB) destination at the channel level and add it to the list of destinations. The endpoint name string is an H.323 endpoint destination as defined in the **H.323 endpoint destination** command.

voice no destination

Command Usage

Delete an H.323 endpoint or local channel destination with specified name.

Syntax Options

voice no destination <“*DestName*”>

Definitions:

DestName Identifies either the voice call H.323 endpoint destination or local channel destination by name (e.g., **to salem**), or the maximum length of 40 characters. The following characters are permitted in the call destination name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

Default:

None

Command Example:

voice no destination tosaalem
voice no destination to_vsd1

view voice destination

Command Usage

Display an H.323 endpoint or local channel destination with specified name.

Syntax Options

```
view voice destination < "DestName" >
```

Definitions:

DestName Identifies either the voice call H.323 endpoint destination or local channel destination by name (e.g., **to salem**), or the maximum length of 40 characters. The following characters are permitted in the call destination name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

Default:

None

Command Example:

```
view voice destination tosaalem
view voice destination tocalab
```

Screen Output

To view parameters for an H.323 voice endpoint destination, type **view voice destination** and a valid destination name, e.g., **view voice destination to VSD_1**, and then press **<Enter>**.

A screen similar to the following displays.

```
*****
Viewing Destination
*****
!
voice destination VSD_1 h.323 address 195.167.10.33 1720
!
voice destination VSD_2 h.323 address 195.167.10.34 1720
!
voice destination to VSD_1 port 1 local channel VSD_1/1/1-24
!
voice destination to VSD_2 port 1 local channel VSD_1/1/1-24
!
voice phone group Ext. of PBX__1
!
voice phone group Ext. of PBX__2
!
voice phone group Ext. of PBX__1 type local extensions
!
voice phone group Ext. of PBX__2 type local extensions
!
voice phone group Ext. of PBX__1 site prefix off
!
voice phone group Ext. of PBX__2 site prefix off
!
voice phone group Ext. of PBX__1 format "xxxx"
!
voice phone group Ext of PBX_2 format "xxxx"
!
voice phone group Ext. of PBX__1 strip digit length 0
!
voice phone group Ext. of PBX__2 strip digit length 0
```

voice phone group**Command Usage**

Create phone group with specified name to add to phone group list.

Syntax Options

```
voice phone group <“PhoneGroupName”>
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

Default:

None

Command Example:

```
voice phone group salem_engr1  
voice phone group calab_engr2
```

voice no phone group

Command Usage

Delete phone group with specified name from phone group list.

Syntax Options

```
voice no phone group < "PhoneGroupName" >
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

Default:

None

Command Example:

```
voice no phone group salem_engr1  
voice no phone group calab_engr2
```

view voice phone group

Command Usage

Display phone group with specified name in phone group list.

Syntax Options

```
view voice phone group <"PhoneGroupName">
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

Default:

None

Command Example:

```
view voice phone group salem_engr1
view voice phone group calab_engr1
```

Screen Output

To view parameters for a voice phone group, type **view voice phone group** and a valid phone group name, e.g., **view voice phone group PBX_1**, and then press **<Enter>**.

A screen similar to the following displays.

```
*****
Viewing Phone Groups
*****
!
voice phone group "Ext. of PBX_1
voice phone group "Ext. of PBX_1 type local extensions
voice phone group "Ext. of PBX_1 site prefix off
voice phone group "Ext. of PBX_1 format "xxx"
voice phone group "Ext. of PBX_1 strip digit length 0
!
```

voice phone group site prefix

Command Usage

Set unique phone group site *prefix* for routing VoIP calls (on/off).

Syntax Options

```
voice phone group <"PhoneGroupName"> site prefix {on | off}
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON site prefix for specified voice phone group.

off Turns OFF site prefix for specified voice phone group.

Default:

The default setting is **off**.

Command Example:

```
voice phone group salem_engr1 site prefix off  
voice phone group salem_engr2 site prefix on
```

Remarks

Site prefix is similar to a trunk group.

voice phone group site prefix digits

Command Usage

Specify unique phone group site *prefix digits* for routing VoIP calls.

Syntax Options

```
voice phone group < "PhoneGroupName" > site prefix digits < "string" >
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

string Identifies the phone group prefix string, (e.g., **81**); maximum length of 23 characters. The following characters are permitted in the phone group prefix string: 0-9, # and *; at least one character must be used.

◆ Syntax Notes ◆

If a terminating digit has been configured for the voice switching daughtercard via the **voice daughter card termination digit** command, the specified terminating digit cannot be used as a site prefix digit.

To use this command, the **voice phone group site prefix** must be turned ON.

To use this command, the associated voice numbering plan must first be activated via the **voice numbering plan activate** command.

Default:

None

Command Example:

```
voice phone group salem_engr1 site prefix digits "***"
voice phone group salem_engr2 site prefix digits 81
```

Remarks

The example in this command for the string portion of the syntax sets the site prefix digits to be used to **81**. This means that from any channel on the network, whenever the digits **81** are pressed, a certain number of digits is expected to follow. If the site prefix is set to **81**, and the format is **xxx** to get to extension **306**, for instance, the caller would dial **81306** to get to extension **306** from anywhere in the VoIP network. In this case, a two digit dialing prefix is used, so the VoIP network can support 999 sites, with **100** extensions per site.

A prefix is an indicator consisting of one or more digits allowing selection of different types of number formats (e.g., local, national or international), transit networks, and/or the service.

Prefixes are part of the number and are signaled over internetwork or international boundaries. When prefixes are used, the user or automatic calling equipment always enters them. Prefixes are used on the source voice switching daughtercard, and are sent to the remote end destination. H.323 gatekeepers receive prefixes as part of the alias name.

If the phone group dialing type is either NANP extensions or International extensions, then the first digit of the site prefix digit cannot be the same as the first digit of the extension, as specified via the **phone group numbers to include in range of digits** command.

voice phone group type

Command Usage

Specify phone group dialing type numbering scheme as either intra-VoIP network extensions or PSTN telephone numbers.

Syntax Options

voice phone group <“PhoneGroupName”> type { local extensions | nanp extensions | international extensions | nanp pstn | international pstn }

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**; maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

local extensions Indicates use of intra-VoIP network (PBX) extensions, (e.g., xxxx), between 1 and 11 digits, .

nanp extensions Indicates use of North American Numbering Plan (NANP) intra-VoIP network (PBX) extensions, (e.g., x-xxx-xxx-xxxx); consists of single-digit long distance designator, three-digit area code, three-digit (CO) exchange prefix, and unique four-digit code for the telephone subscriber. Any combination of three digits may now be used for the area code. NANP is the numbering scheme used to assign area codes and also to establish rules for call routing in Canada and the United States.

international extensions Indicates use of international intra-VoIP network (PBX) extensions, (e.g., xx-xxx-xxx-xxxx); consists of between 10 and 26 digits (includes field separators, i.e., hyphens; without field separators maximum extension length is 15 digits); 1-3 digit country code (CC), national destination code (NDC), and subscriber number (SN). (Field separators *not available this release*.)

nanp pstn Indicates use of North American Numbering Plan via the Public Switching Telephone Network; consists of local and long distance domestic telephone numbers, (e.g., 1-xxx-xxx-xxxx).

international pstn Indicates use of international long distance telephone numbers via the PSTN, (e.g., 01-xxx-xxx-xxxx).

◆ Syntax Notes ◆

Restrictions on numbering schemes from various telephone companies are beyond the scope of this document.

More detailed, syntax-related rules specific to using local, NANP, international extensions, and NANP PSTN and International PSTN dialing types are detailed below.

Default:

The default setting is **local extensions**.

Command Example:

```
voice phone group salem_engr1 type local extensions
voice phone group salem_engr2 type nanp extensions
voice phone group salem_engr3 type international extensions
voice phone group calab_engr1 type nanp pstn
voice phone group caleb_engr2 type international pstn
```

Syntax Notes

Local Extensions (intended for daughtercard to daughtercard calls)

- If local extensions used, then voice switching daughtercard recognizes *fixed* number of dialed extension digits, i.e., **0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10** or **11**. *No site prefix required.*
- If H.323 gatekeeper control is set to ON via the **voice network card h.323 gatekeeper control** command, then the telephone extensions are set via the phone group **numbers to include in range of digits** command, and automatically registered with the gatekeeper as an alias.
- Format string must have between one and seven numbering scheme placeholders or designators (**xxxxxxx**); **0** through **9** are allowed in any combination as long as at least one “**x**” is used, e.g., **1x, 2xx, 1xxxxxx, 2xxxxxx**, etc. A dialed digit cannot follow an “**x**”. Neither the # and * symbols nor letters A through E are allowed. Format string can only contain dialed digits (no field separators).
- Local extensions cannot be used in conjunction with any other dialing types.

NANP Extensions (intended for daughtercard to daughtercard calls)

- If NANP extensions used digital voice switching daughtercard recognizes *variable* number of dialed digits, i.e., between seven and 11 digits; 11 digits when long distance designator included.
- If H.323 gatekeeper control is set to ON via the **voice network card h.323 gatekeeper control** command, then the telephone extensions are set via the phone group **numbers to include in range of digits** command, and automatically registered with the gatekeeper as an alias.
- If long distance designator used it can only contain one digit which must be the number **1** followed by field separator. (Field separators *not available this release*.)
- Three-digit central office (CO) code, field separator and unique four-digit subscriber number are required.
- Format string must contain dialed digits and limited use of field separators. Field separators (hyphens) in sequence indicate empty field; two field separators are not allowed in sequence as part of a valid format string. A dialed digit cannot follow an “**x**”. At least one “**x**” is required.

If no format is specified, then the phone group **numbers to include in range of digits** command is not allowed. Command can use only **1, 2, 3, 4,** or **7** numbering scheme “**x**” designators. Designators are not allowed in area code or prefixes unless long distance designator is used, in which case three designators must be used in the prefix (but never the area code), e.g., **818-123-000x, 1-818-xxx-xxxx**. Allowed digits include **000** through **999**.

International Extensions (intended for daughtercard to daughtercard calls)

- If NANP extensions used then voice switching daughtercard recognizes *variable* number of dialed digits, i.e., between 10 and 26 digits; 26 digits when long distance designator(s) and site prefix included. (See international PSTN extensions for more information on site prefixes.)
- If H.323 gatekeeper control is set to ON via the **voice network card h.323 gatekeeper control** command, then the telephone extensions are set via the phone group **numbers to include in range of digits** command, and automatically registered with the gatekeeper as an alias.
- Format string must contain dialed digits and limited use of field separators. Field separators (hyphens) in sequence indicate empty field; two hyphens are not allowed in sequence as part of a valid format string. A dialed digit cannot follow an “x”. At least one “x” is required.
- Format is optional; if not specified, the phone group **numbers to include in range of digits** command is not allowed.

NANP PSTN (intended for daughtercard to PSTN calls)

- If NANP PSTN telephone numbers used then voice switching daughtercard recognizes *variable* number of dialed digits, i.e., between seven and 11 digits; *11 digits when long distance designator included*.
- If H.323 gatekeeper control is set to ON via the **voice network card h.323 gatekeeper control** command, then the telephone numbers are *not* registered with the gatekeeper as an alias.
- Format string must contain dialed digits and limited use of field separators. Field separators (two hyphens) in sequence indicate empty field; two field separators are not allowed in sequence as part of a valid format string. A dialed digit cannot follow an “x”. At least one “x” is required.
- Format is optional; if not specified, the phone group **numbers to include in range of digits** command is not allowed.

PSTN International (intended for daughtercard to PSTN calls)

- If PSTN international telephone numbers used then voice switching daughtercard recognizes *variable* number of dialed digits, i.e., between 10 and 26 digits; *26 digits when long distance designator(s) and required site prefix included*.

Length of site prefix digits, which can be any combination of numbers from 0 to 9, cannot exceed 26 digits total when added to the format string. Site prefix digits are set via the **voice phone group site prefix digits** and **voice phone group site prefix (on/off)** commands.

- One- to three-digit country code (CC), three-digit central office (CO) code, field separator and unique four-digit subscriber number (SN) are required; cannot exceed 15 digits when field separators not used.
- Format string must contain dialed digits and limited use of field separators. Field separators (hyphens) in sequence indicate empty field; two field separators are not allowed in sequence as part of a valid format string. A dialed digit cannot follow an “x”. At least one “x” is required.

Format must be compatible with ITU E.164.1 specification.

- Format is optional; if not specified, the phone group **numbers to include in range of digits** command is not allowed.

voice phone group format

Command Usage

Specify phone group format of telephone number and number of outbound digits to dial.

Syntax Options

```
voice phone group <“PhoneGroupName”> format <“formatString”>
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

formatString Identifies the phone group format string, (e.g., **xxx**); maximum length of characters **24**. The only character permitted in the phone group format string: x and X.

◆ Syntax Notes ◆

Valid field separators include () , . = _ + [] { } \ / ; ; < or space (no hyphens); field separators are not allowed for local extensions.

For this command only, a dialed digit can be **0, 1, 2, 3, 4, 5, 6, 7, 8, 9** and **x**.

For the format string, a dialed digit cannot follow an “x” designator there must be at least one “x” (up to seven placeholder or designators allowed).

Format is optional; if format specified, the phone group **numbers to include in range of digits** command is not allowed. Also, if format specified, then site prefix digits must be set and turned ON via the **voice phone group site prefix digits** and **voice phone group site prefix (on/off)** commands.

If a termination digit has been set via the **VSD termination digit** command, then that digit cannot be used in this command.

The digits specified in the site prefix digits and the number of digits implied by the format must be unique on the entire network numbering plan for different types.

Default:

None

Command Example:

```
voice phone group salem_engr1 format 1xxx
voice phone group salem_engr2 format 2xxx
voice phone group salem_engr3 format 81x xxxx
voice phone group calab_engr1 format 818 xxxx
voice phone group calab_engr2 format 31xx
voice phone group calab_engr3 format 41xx
```

Remarks

This command calculates the number of digits allowed and describes the flexible portion of a dialing number to collect.

voice phone group strip digit length

Command Usage

Specify number of outbound (collected) digits to strip in phone group before forwarding call (optional).

Syntax Options

```
voice phone group <“PhoneGroupName”> strip digit length <num >
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

num Identifies the phone group strip digit length string, (e.g., **2**); maximum string length is **24**; minimum is 0.

Default:

The default *num* value is **0**.

Command Example:

```
voice phone group salem_engr1 strip digit 0  
voice phone group salem_engr2 strip digit 1  
voice phone group salem_engr3 strip digit 2
```

voice phone group forwarding prefix

Command Usage

Set voice phone group to allow string of digits (prefix) to be forwarded prior to forwarding call to endpoint destination (on/off).

Syntax Options

voice phone group <“*PhoneGroupName*”> forwarding prefix {on | off}

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

on Turns ON forwarding of voice phone group prefix.

off Turns OFF forwarding of voice phone group prefix.

Default:

The default setting is **off**.

Command Example:

```
voice phone group salem_engr1 forwarding prefix off
voice phone group salem_engr2 forwarding prefix on
```

voice phone group forwarding prefix digits

Command Usage

Specify voice phone group digits to prefix before forwarding call (optional).

Syntax Options

```
voice phone group <“PhoneGroupName”> forwarding prefix digits <“vsmTelephonePrefixNum”>
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

PrefixNum Identifies the forwarding prefix digits in the specified phone group, (e.g., **9**); maximum string length is **14** (MAX_DIAL_DIGITS-1); each digit can be either 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, or #.

◆ Syntax Notes ◆

To use this command, the **phone group forwarding prefix** command must first be enabled.

Default:

The default *PrefixNum* value is **0**.

Command Example:

```
voice phone group salem_engr1 strip digit 0  
voice phone group salem_engr2 strip digit 1  
voice phone group salem_engr3 strip digit 2
```

voice phone group add numbers

Command Usage

Specify phone group *numbers to include* in range of digits for phone format string.

Syntax Options

```
voice phone group < "PhoneGroupName" > add numbers < "StartRange" > [ thru "EndRange" ]
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

StartRange Specifies starting digits to add to designated phone group as per format specified in **voice phone format string** command, (e.g., extension **4600**).

thru EndRange Specifies ending digits to add to designated phone group as per format specified in **voice phone group format string** command, (e.g., extension **thru 4800**). Optional command syntax.

◆ Syntax Notes ◆

Ranges must be unique across the entire network numbering scheme, and digits specified must match the allowed number of digits as specified in the format string via the **voice phone group format string** command.

Termination digits as specified via the **VSD termination digit** command cannot be specified in either range.

Multiples of this command can be issued to have a cumulative effect.

If no number is entered for the end range, then the start range value is used.

If no number is entered for the starting range, all zeroes will be used based upon the specified format string.

Default:

None

Command Example:

```
voice phone group salem_engr1 add numbers 4600 thru 4800
voice phone group salem_engr1 add numbers 4600
voice phone group calab_engr1 add numbers 2500 thru 2750
voice phone group calab_engr1 add numbers 2500
```

Remarks

If the phone group dialing type is NANP extensions or International extensions, then the first digit of the site prefix digits cannot be the same as the first digits as specified via the **phone group numbers to include in range of digits** command.

voice phone group delete numbers

Command Usage

Specify phone group *numbers to remove* from range of digits for phone format string. (*Not available this release.*)

Syntax Options

```
voice phone group <“PhoneGroupName”> delete numbers <“StartRange”> [[thru] “EndRange”]
```

Definitions:

PhoneGroupName Identifies the phone group by name, (e.g., **salem_engr1**); maximum length of 40 characters. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

StartRange Specifies starting digits to delete from designated phone group as per format specified in **voice phone group format string** command, (e.g., four-digit local extension **4600**).

thru Optional command syntax.

EndRange Specifies ending digits to delete from designated phone group as per format specified in **voice phone group format string** command, (e.g., four-digit local extension **4800**).

Default:

None

Command Example:

```
voice phone group salem_engr1 delete numbers 4600 thru 4800
voice phone group salem_engr1 delete numbers 4600 4800
voice phone group calab_engr1 delete numbers 2500 thru 2750
voice phone group calab_engr1 delete numbers 2500 2750
```

voice numbering plan

Command Usage

Create numbering plan with specified name.

Syntax Options

voice numbering plan <“*NumberingPlanName*”>

Definitions:

NumberingPlanName Identifies the numbering plan by name, (e.g., **saalem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

◆ Syntax Notes ◆

If the **voice network card h.323 gatekeeper control** command is ON, then the voice switching daughtercard automatically generates an additional alias name when the **activate voice numbering plan** command is issued.

Numbering plan names cannot be named “all” because “all” is a reserved numbering plan name.

Numbering plans do not take effect unless they are activated via the **activate numbering plan** command.

Default:

None

Command Example:

```
voice numbering plan saalem#plan1
voice numbering plan calab#plan2
```

Remarks

Numbering plans associate one or more groups to one or more destinations (hunting targets) to be called.

voice no numbering plan

Command Usage

Delete numbering plan with specified name.

Syntax Options

voice no numbering plan < “*NumberingPlanName*” >

Definitions:

NumberingPlanName Identifies the numbering plan by name, (e.g., **saalem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

Default:

None

Command Example:

voice no numbering plan saalem#plan1

voice no numbering plan calab#plan2

view voice numbering plan

Command Usage

Delete numbering plan with specified name.

Syntax Options

```
view voice numbering plan <“NumberingPlanName”>
```

Definitions:

NumberingPlanName Identifies the numbering plan by name, (e.g., **saalem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

Default:

None

Command Example:

```
view voice numbering plan saalem#plan1  
view voice numbering plan calab#plan2
```

Screen Output

To view a voice numbering plan, type **view voice numbering plan** followed by a valid numbering plan name, e.g., **view voice numbering plan PBX_1**, and then press **<Enter>**.

A screen similar to the following displays.

```
*****
Viewing Numbering Plan
*****
!
voice numbering plan to PBX_1
!
voice numbering plan to PBX_2
!
voice numbering plan to PBX_1 hunt method round robin
!
voice numbering plan to PBX_1 hunt method round robin
!
voice numbering plan to PBX_1 associate destination member to VSD_1
!
voice numbering plan to PBX_2 associate destination member to VSD_2
!
voice numbering plan to PBX_1 associate phone group member Ext. of PBX_1
!
voice numbering plan to PBX_2 associate phone group member Ext. of PBX_2
!
voice numbering plan to PBX_1 description trunk to route calls from VSD1 to PBX1
!
voice numbering plan to PBX_2 description trunk to route calls from VSD2 to PBX2
!
voice numbering plan to PBX_1
!
```

voice numbering plan activate

Command Usage

Activate all voice numbering plans at once or by specified name.

Syntax Options

voice numbering plan { all | “NumberingPlanName” } activate

Definitions:

all Activates all numbering plans at once.

NumberingPlanName Identifies the numbering plan by name, (e.g., **saalem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }. (*Not available this release.*)

◆ Syntax Notes ◆

If the **voice network card h.323 gatekeeper control** command is ON, when the **voice numbering plan activate** command is issued, the voice switching daughtercard automatically generates an additional alias name. (*Not available this release.*)

This command must be used directly from the CLI prompt rather than from within the **vsmbboot.asc** file. In the **vsmbboot.asc** file, the **voice daughter card activate** command should be used instead.

Default:

None

Command Example:

```
voice numbering plan all activate
voice numbering plan saalem#plan1 activate
voice numbering plan calab#plan2 activate
```

Remarks

Numbering plans do not take effect until this command is issued, or the voice daughtercard is activated, and use of this command will not affect calls that are already in place.

Once a numbering plan is activated, all new connections are temporarily halted until this command is completed. It can take up to approximately 10 seconds for a numbering plan to be activated. As a result, it is recommended that this command be issued only when it will have minimum impact on callers. It is also more efficient to activate all the numbering plans at once, rather than individually.

Once the VoIP gateway has been configured completely using the **vsmbboot.asc** file, this command can be issued later to update or “dynamically reconfigure” the VoIP gateway. This means certain configuration updates can be made to the VoIP gateway without rebooting the switch as follows:

- Add VoIP gateway to VoIP network.
- Add telephone number to VoIP gateway.
- Delete telephone destination number or destination.

The dynamic reconfiguration feature works only on the VoIP commands specifically listed or grouped here:

- **voice daughter card first digit wait duration**
- **voice daughter card inter digit wait duration**
- **voice daughter card dial time duration**
- **voice daughter card termination digit**
- **voice network card h.323 gatekeeper associate phone group**
- all “**voice destination**” commands
- all “**voice phone group**” commands
- all “**voice numbering plan**” commands

It should be noted that updates made to the configuration using this command are only temporary. In order to make the configuration changes permanent, it will subsequently be necessary to download the changes to the **vsmboot.asc** file, and then reissue the **voice numbering plan activate** command.

voice numbering plan hunt method

Command Usage

Specify numbering plan method of outgoing hunting (destination group).

Syntax Options

voice numbering plan <“*NumberingPlanName*”> hunt method {round robin | top down}

Definitions:

NumberingPlanName Identifies the numbering plan by name, (e.g., **saalem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

round robin Indicates numbering plan uses round robin hunting method to find open lines for incoming calls.

top down Indicates numbering plan uses top down hunting method to find open lines for incoming calls.

Default:

The default setting is **round robin**.

Command Example:

```
voice numbering plan saalem#plan1 hunt method round robin
voice numbering plan calab#plan2 hunt method top down
```

Remarks

This command also groups related destinations together.

The round robin hunting method starts from the destination member just after the last used destination member each time a hunt request is received. The last used destination member is “remembered” across sessions.

The top down hunting method starts from the first destination member in the hunt group each time a new session is started.

Hunt groups allow telephone lines to be organized so that when the first line tried is unavailable for an incoming call, the next available line (using either the round robin or top down method) is hunted until an open line is located.

voice numbering plan description

Command Usage

Define numbering plan of specified phone group (optional).

Syntax Options

```
voice numbering plan <“NumberingPlanName”> description <“string”>
```

Definitions:

NumberingPlanName Identifies the numbering plan by name, (e.g., **saalem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

string Text string that describes the specified voice numbering plan, (e.g., **“phone group a_eng”**); maximum length of characters **40**. The following characters are permitted in the numbering plan description string: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

Default:

None

Command Example:

```
voice numbering plan saalem#plan1 description “phone group a_eng”
```

```
voice numbering plan saalem#plan2 description “phone group b_eng”
```

```
voice numbering plan calab#plan1 description “phone group a_eng”
```

```
voice numbering plan calab#plan2 description “phone group b_eng”
```

Remarks

This command is used to store a description of this phone group for convenience. Because the command is optional it has no effect on the switch. It can also be used to hold the circuit identifier (see the **voice port circuit identifier** command).

voice numbering plan destination member

Command Usage

Associate (or disassociate) destinations, or hunting targets, with numbering plans by name.

Syntax Options

```
voice numbering plan <“NumberingPlanName”> {associate | disassociate} destination member
<“DestName”>
```

Definitions:

NumberingPlanName Identifies the numbering plan by name, (e.g., **saalem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

associate Indicates specified numbering plan associated with destination member.

disassociate Indicates specified numbering plan disassociated with destination member.

DestName Identifies either the voice call H.323 endpoint destination or local channel destination by name, (e.g., **to saalem**) or the c; maximum length of 40 characters. The following characters are permitted in the call destination name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

◆ Syntax Notes ◆

To use this command, the hunt method must first be specified via the **voice numbering plan hunt method** command.

It is recommended that a local channel destination be used for the destination name string (“*DestName*”).

Multiple destinations can be associated with a single numbering plan in order to create a hunt group. The hunting order is determined by the order in which the hunt groups are associated in the **vsmboot.asc** file.

Default:

The default setting is **associate**.

Command Example:

```
voice numbering plan saalem#plan1 associate destination member tocalab
voice numbering plan saalem#plan2 disassociate destination member to_vsd1
voice numbering plan calab#plan1 associate destination member tosaalem
voice numbering plan calab#plan2 disassociate destination member to_vsd2
```

Remarks

This command is used to append a destination (hunting target) to the associated numbering plan for telephone number hunting. The numbering plan destination list is used to hunt for destinations (targets). Each numbering plan destination list requires at least one member.

voice numbering plan phone group member

Command Usage

Associate (or disassociate) phone groups with numbering plans by name.

Syntax Options

```
voice numbering plan <“NumberingPlanName”> {associate | disassociate} phone group member
<“PhoneGroupName”>
```

Definitions:

NumberingPlanName Identifies the numbering plan by name, (e.g., **salem#plan1**); maximum length of 40 characters. The following characters are permitted in the numbering plan name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }.

associate Indicates specified phone group associated with specified numbering plan.

disassociate Indicates specified phone group disassociated with specified numbering plan.

PhoneGroupName Identifies the phone group by name, (e.g., **calab_engr1**); maximum length of characters **60**. The following characters are permitted in the phone group name: a-z, A-Z, 0-9, space and # * ~ ' ; : , . @ \$ % ^ _ & | / \ < > () [] { }

◆ Syntax Notes ◆

To use this command, the destination member must first be associated via the **associate destination member** command.

At least one phone group must be associated before activating the associated numbering plan.

Multiple phone groups can be associated with a single numbering plan.

Default:

None

Command Example:

```
voice numbering plan salem#plan1 associate phone group member calab_engr1
voice numbering plan salem#plan2 disassociate phone group member calab_engr2
voice numbering plan calab#plan1 associate phone group member salem_engr1
voice numbering plan calab#plan2 disassociate phone group member salem_engr2
```

System-Wide VoIP Commands

The commands listed and described below are used to display system-wide VoIP command settings and statistics as follows: various VSD level parameters and configured voice items, including telephony, telephony channel, voice play out, DSP (receive and transmit), errors, modem, fax and ISDN statistics.

View Voice Switching Daughtercard Parameters

View Voice Switching Daughtercard Port Parameters

View Voice Switching Daughtercard Channel Parameters

View Voice Switching Daughtercard Network Parameters

Statistics Collection (on/off)

View Statistics

- telephony statistics
- channel statistics
- voice play out statistics
- DSP receive and transmit statistics
- error statistics
- modem statistics
- fax statistics
- ISDN level 2 statistics

Reset Statistics

- telephony statistics
- channel statistics
- voice play out statistics
- DSP receive and transmit statistics
- error statistics
- modem statistics
- fax statistics
- ISDN level 2 statistics

view voice daughter card

Command Usage

Display voice switching daughtercard parameters at the daughtercard level.

Syntax Options

view voice daughter card <slot/card_number >

Definitions:

slot Specifies chassis slot number where VSM is installed, (e.g., **2**).

card_number Specifies physical port number on voice daughtercard, (e.g., **1**).

Default:

None

Command Example:

view voice daughter card 2/1
view voice daughter card 2/2
view voice daughter card 3/1
view voice daughter card 3/2

Screen Output

To view voice daughtercard parameters, type **view voice daughter card** followed by valid slot and daughtercard port numbers, e.g., **view voice daughtercard 4/1**, and then press **<Enter>**.

A screen similar to the following displays.

```
*****  
Viewing Daughter Card  
*****  
!  
voice daughtercard 4/1 ip address 127.0.0.0  
!  
voice daughtercard 4/1 ip mask 255.255.255.0  
!  
voice daughtercard 4/1 activate
```

view voice port**Command Usage**

Display voice switching daughtercard parameters at the port level.

Syntax Options

```
view voice port <slot/port >
```

Definitions:

slot Specifies chassis slot number where VSM is installed, (e.g., **2**).

port Specifies physical port number on voice daughtercard, (e.g., **1**).

Default:

None

Command Example:

```
view voice port 2/1
view voice port 2/2
view voice port 2/3
view voice port 2/4
```

Screen Output

To view voice daughtercard port parameters, type **view voice port** followed by valid slot and daughtercard port numbers, e.g., **view voice port 4/1**, and then press **<Enter>**.

A screen similar to the following displays.

```
*****
Viewing Port
*****
!
voice port 4/1 interface type T1
!
```

view voice channel**Command Usage**

Display voice switching daughtercard parameters at the channel level.

Syntax Options

```
view voice channel <slot/port/channel >
```

Definitions:

slot Specifies chassis slot number where VSM is installed, (e.g., **2**).

port Specifies physical port number on voice daughtercard, (e.g., **1**).

channel Specifies port channel number in which to view voice channel parameters, (e.g., **25**).

Default:

None

Command Examples:

```
view voice channel 2/1/1
view voice channel 2/1/15
view voice channel 2/1/17
view voice channel 2/1/30
view voice channel 2/1/60
view voice channel 2/2/1
view voice channel 2/2/15
view voice channel 2/2/17
view voice channel 2/2/30
```

Screen Output

To view voice daughtercard parameters, type **view voice channel** followed by valid slot, daughtercard port and channel number(s), e.g., **view voice channel 4/1/1-12**, and then press **<Enter>**.

A screen similar to the following displays.

```
*****
Viewing Channel
*****
!
voice channel 4/1/1 mode telephony
voice channel 4/1/2 mode telephony
voice channel 4/1/3 mode telephony
voice channel 4/1/4 mode telephony
voice channel 4/1/5 mode telephony
voice channel 4/1/6 mode telephony
voice channel 4/1/7 mode telephony
voice channel 4/1/8 mode telephony
voice channel 4/1/9 mode telephony
voice channel 4/1/10 mode telephony
voice channel 4/1/11 mode telephony
voice channel 4/1/12 mode telephony
```

!

view voice network card**Command Usage**

Display voice switching daughtercard parameters at the network level for H.323 purposes.

Syntax Options

```
view voice network card <slot/card_number >
```

Definitions:

slot Specifies chassis slot number where VSM is installed, (e.g., **2**).

card_number Specifies the voice daughtercard position number, (e.g., **1**).

Default:

None

Command Example:

```
view voice network card 2/1  
view voice network card 2/2  
view voice network card 3/1  
view voice network card 3/2
```

Screen Output

No screen output available at this time.

voice daughter card statistics collection

Command Usage

Sets statistics collection for specified voice daughter card (on/off). *(Not available this release.)*

Syntax Options

voice daughter card <slot/card_number> stat[istic]s collection {on | off}

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>card_number</i>	Specifies the voice daughtercard position number, (e.g., 1).
istic	Optional command syntax. Type either stats or statistics in the command line.
on	Turns ON statistics collection for specified voice daughtercard.
off	Turns OFF statistics collection for specified voice daughtercard.

◆ Syntax Notes ◆

This command must be issued and activated from within the **vsmboot.asc** file.

Default:

None

Command Example:

voice daughter card 2/1 statistics collection on
voice daughter card 2/2 stats collection off

Remarks

Use of this command can cause significant degradation in the performance of a VoIP daughtercard. In certain instances a daughtercard may appear non-operational. Symptoms exhibited may include dead air, h.323 connection failures, lost packets, jitter, etc. This command should only be used in a tightly controlled environment and under the discretion of qualified personnel. It is strongly recommended that this command be tested extensively under high traffic loads before using in a real time VoIP network.

voice channel reset all statistics

Command Usage

Reset all statistics for one or more specified voice channels. (*Not available this release.*)

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> **reset all stat[istic]s**

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number range in which to reset all telephony statistics, (e.g., 1-12).
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmbot.asc** file.

Default:

None

Command Example:

voice channel 2/1/1-12 reset all statistics
voice channel 2/1/13-24 reset all stats

view voice channel telephony level stats

Command Usage

Display cumulative telephony statistics. (*Not available this release.*)

Syntax Options

```
view voice channel <slot/port/channel > telephony level stat[istic]s
```

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number in which to view telephony statistics, (e.g., 25).
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmbot.asc** file.

Default:

None

Command Example:

```
view voice channel 2/1 25 telephony level statistics
view voice channel 2/1 25 telephony level stats
view voice channel 2/2 30 telephony level statistics
view voice channel 2/2 30 telephony level stats
```

Remarks

Displays the current receive levels, mean receive levels, transmit levels and the mean transmit levels at the PCM interface of the DSP for the specified channel (received from or transmitted to the telephony interface). Current levels are given in 0.1 dBm0 units. Mean values are in 0.1 linear PCM units.

view voice channel telephony channel stats

Command Usage

Display cumulative channel statistics. (*Not available this release.*)

Syntax Options

view voice channel <slot/port/startChannel-endChannel > **telephony channel stat**[istic]s

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number range in which to view channel statistics, (e.g., 25); cumulative since channel was placed in service.
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

```
view voice channel 2/1 25 telephony level statistics
view voice channel 2/1 25 telephony level stats
view voice channel 2/2 30 telephony level statistics
view voice channel 2/2 30 telephony level stats
```

view voice channel voice playout stats

Command Usage

Display cumulative voice play out statistics for call in progress. (*Not available this release.*)

Syntax Options

view voice channel <slot/port/startChannel-endChannel > voice playout stat[istic]s

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- channel* Specifies port channel number range in which to view DSP voice play out statistics, (e.g., **25**); cumulative for the current call on specified channel.
- istic** Optional command syntax. Type either **stats** or **statistics** in the command line.

◆ **Syntax Notes** ◆

This command only works from within the command line interface, and cannot be used from within the **vsmbot.asc** file.

Default:

None

Command Example:

view voice channel 2/1 25 voice playout statistics
view voice channel 2/1 25 voice playout stats
view voice channel 2/2 30 voice playout statistics
view voice channel 2/2 30 voice playout stats

view voice channel dsp stats

Command Usage

Display cumulative DSP receive and transmit statistics. (*Not available this release.*)

Syntax Options

view voice channel <slot/port/startChannel-endChannel > dsp stat[istic]s

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number in which to view DSP receive and transmit statistics, (e.g., 25); cumulative for the current call on specified channel.
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

```
view voice channel 2/1 25 dsp statistics
view voice channel 2/1 25 dsp stats
view voice channel 2/2 30 dsp statistics
view voice channel 2/2 30 dsp stats
```

view voice channel error stats

Command Usage

Display cumulative error statistics. (*Not available this release.*)

Syntax Options

view voice channel <slot/port/startChannel-endChannel > **error stat[istic]s**

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number in which to view error statistics, (e.g., 25); cumulative for the current call on specified channel.
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmbot.asc** file.

Default:

None

Command Example:

```
view voice channel 2/1 25 error statistics  
view voice channel 2/1 25 error playout stats  
view voice channel 2/2 30 error playout statistics  
view voice channel 2/2 30 error playout stats
```

view voice channel modem stats

Command Usage

Display cumulative modem statistics. (*Not available this release.*)

Syntax Options

view voice channel <slot/port/startChannel-endChannel > modem stat[istic]s

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number in which to view modem statistics, (e.g., 25); cumulative for the current call on specified channel.
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmbot.asc** file.

Default:

None

Command Example:

```
view voice channel 2/1 25 modem statistics
view voice channel 2/1 25 modem stats
view voice channel 2/2 30 modem statistics
view voice channel 2/2 30 modem stats
```

view voice channel fax stats

Command Usage

Display cumulative facsimile statistics. (*Not available this release.*)

Syntax Options

view voice channel <slot/port/startChannel-endChannel > fax stat[istic]s

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- channel* Specifies port channel number range in which to view facsimile statistics, (e.g., **1-12**); cumulative for the current call on specified channel.
- istic** Optional command syntax. Type either **stats** or **statistics** in the command line.

◆ **Syntax Notes** ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

view voice channel 2/1 25 fax statistics
view voice channel 2/1 25 fax stats
view voice channel 2/2 30 fax statistics
view voice channel 2/2 30 fax stats

view voice channel isdn level 2 stats

Command Usage

Display cumulative ISDN level 2 statistics. (*Not available this release.*)

Syntax Options

view voice channel <slot/port/startChannel-endChannel > isdn level 2 stat[istic]s

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number in which to view ISDN level 2 statistics, (e.g., 25).
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

```
view voice channel 2/1 25 isdn level 2 statistics
view voice channel 2/1 25 isdn level 2 stats
view voice channel 2/2 30 isdn level 2 statistics
view voice channel 2/2 30 isdn level 2 stats
```

voice channel reset telephony level stats

Command Usage

Reset cumulative telephony statistics for one or more voice channels. (*Not available this release.*)

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> **reset telephony level stat[istic]s**

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number range in which to reset telephony statistics, (e.g., 1-12).
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

voice channel 2/1/1-12 reset telephony level statistics
voice channel 2/2/13-24 reset telephony level stats

voice channel reset telephony channel stats

Command Usage

Reset channel statistics for one or more voice channels. (*Not available this release.*)

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> **reset telephony channel stat[istic]s**

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number range in which to reset all telephony channel statistics, (e.g., 1-12); cumulative since channel was placed in service.
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

voice channel 2/1/1-12 reset telephony level statistics
voice channel 2/1/13-24 reset telephony level stats

voice channel reset voice playout stats

Command Usage

Reset DSP voice play out statistics for one or more voice channels. (*Not available this release.*)

Syntax Options

voice channel <slot/port/startChannel-endChannel > **reset voice playout stat[istic]s**

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- channel* Specifies port channel number range in which to reset DSP voice play out statistics, (e.g., **1-12**); cumulative for the current call on specified channel.
- istic** Optional command syntax. Type either **stats** or **statistics** in the command line.

◆ **Syntax Notes** ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

voice channel 2/1/1-12 reset voice playout statistics
voice channel 2/1/13-24 reset voice playout stats

voice channel reset dsp stats

Command Usage

Reset DSP voice play out statistics for one or more voice channels. (*Not available this release.*)

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> **reset dsp stat[istic]s**

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number range in which to reset DSP receive and transmit statistics, (e.g., 1-12); cumulative for the current call on specified channel.
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

voice channel 2/1/1-12 reset dsp statistics
voice channel 2/1/13-24 reset dsp stats

voice channel reset error stats

Command Usage

Reset error statistics for one or more voice channels. (*Not available this release.*)

Syntax Options

voice channel <slot/port/startChannel-endChannel > **reset error stat**[istic]**s**

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number range in which to reset error statistics, (e.g., 1-12); cumulative for the current call on specified channel.
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

voice channel 2/1/1-12 reset error statistics
voice channel 2/1/13-24 reset error playout stats

voice channel reset modem stats

Command Usage

Reset modem statistics for one or more voice channels. (*Not available this release.*)

Syntax Options

voice channel <*slot/port/startChannel-endChannel*> **reset modem stat[istic]s**

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number range in which to reset modem statistics, (e.g., 1-12); cumulative for the current call on specified channel.
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

voice channel 2/1/1-12 reset modem statistics
voice channel 2/1/13-24 reset modem stats

voice channel reset fax stats

Command Usage

Reset facsimile statistics for one or more voice channels. (*Not available this release.*)

Syntax Options

voice channel <slot/port/startChannel-endChannel > **reset fax stat[istic]s**

Definitions:

- slot* Specifies chassis slot number where VSM is installed, (e.g., **2**).
- port* Specifies physical port number on voice daughtercard, (e.g., **1**).
- channel* Specifies port channel number range in which to reset facsimile statistics, (e.g., **1-12**); cumulative for the current call on specified channel.
- istic** Optional command syntax. Type either **stats** or **statistics** in the command line.

◆ **Syntax Notes** ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

voice channel 2/1/1-12 reset fax statistics
voice channel 2/1/13-24 reset fax stats

reset voice channel isdn level 2 stats

Command Usage

Reset ISDN level 2 statistics for one or more voice channels. (*Not available this release.*)

Syntax Options

reset voice channel <*slot/port/startChannel-endChannel*> **isdn level 2 stat[istic]s**

Definitions:

<i>slot</i>	Specifies chassis slot number where VSM is installed, (e.g., 2).
<i>port</i>	Specifies physical port number on voice daughtercard, (e.g., 1).
<i>channel</i>	Specifies port channel number range in which to reset ISDN level 2 statistics, (e.g., 1-12).
istic	Optional command syntax. Type either stats or statistics in the command line.

◆ Syntax Notes ◆

This command only works from within the command line interface, and cannot be used from within the **vsmboot.asc** file.

Default:

None

Command Example:

```
reset voice channel 2/1/1-12 isdn level 2 statistics
reset voice channel 2/1/13-24 isdn level 2 stats
```


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