



Physical and Virtual Voice Interfaces Overview

Before configuring Cisco voice features, it is important to understand the concepts of voice ports and dial peers. In the Cisco implementation of voice, voice ports define the physical interfaces and dial peers define the virtual interfaces to and from which a call is established. Cisco voice ports and dial peers are not specific features, rather they are the foundations on which all other voice features are built. This overview provides a brief introduction to voice ports and dial peers.

For detailed information, please refer to these respective documents in the Cisco IOS Voice Configuration Library, Release 12.3:

- [Voice Port Configuration](#)
- [Dial Peer Configuration on Voice Gateway Routers](#)

Voice Ports

Voice ports on routers and access servers physically connect the router or access server to telephony devices such as telephones, fax machines, PBXs, and PSTN central office (CO) switches. The router's voice-port hardware and software need to be configured to transmit and receive the same type of signaling being used by the device with which they are interfacing so that calls can be exchanged smoothly between the packet network and the circuit-switched network.

Voice port commands define the characteristics associated with a particular voice-port signaling type. The Cisco implementation of voice supports both analog and digital telephony connections. The connection supported (and the associated signaling) depends on the type of voice network module (VNM) or voice feature card (VFC) installed in your Cisco router or access server. The voice port syntax depends on the hardware platform.

Voice ports support three basic analog voice signaling formats:

- **FXO**—Foreign Exchange Office interface. The FXO interface is an RJ-11 connector that allows a connection to be directed at the Public Switched Telephone Network (PSTN) central office (CO) (or to a standard PBX interface, if the local telecommunications authority permits). This interface is of value for off-premises extension applications.
- **FXS**—Foreign Exchange Station interface. The FXS interface is an RJ-11 connector that allows connection for basic telephone equipment, keysets, and PBXs; FXS connections supply ring, voltage, and dial tone.
- **E&M**—Ear and mouth (or recEive and transMit) interface. The E&M interface is an RJ-48 connector that allows connection for PBX trunk lines (tie lines). It is a signaling technique for 2-wire and 4-wire telephone and trunk interfaces.

Depending on the Cisco device you are configuring, the following digital signaling is supported:

- ISDN PRI
- ISDN BRI
- E1 R2
- T1 CAS

For more information, refer to the [Voice Port Configuration](#) document, Cisco IOS Voice Configuration Library, Release 12.3.

Dial Peers

Voice technologies use dial peers to identify call origin and destination and to define the characteristics associated with a call leg. A call leg is logical connection between two points in the connection. An end-to-end voice call comprises four call legs, two from the perspective of the source router, and two from the perspective of the destination router. Attributes applied to a call leg can include specific quality of service (QoS) features, DTMF relay method, compression/decompression (codec), voice activity detection (VAD), and fax rate.

There are two basic kinds of dial peers for each voice implementation:

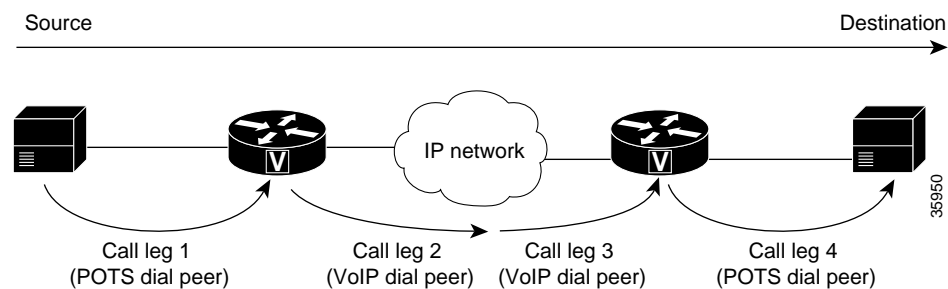
- Plain old telephone service (POTS)—Describe the characteristics of a traditional telephony network connection. POTS dial peers point to a particular voice port on the local router, normally the voice port connecting the router to the local PSTN, PBX, or telephone.
- Voice network (VoIP, VoATM, and VoFR)—Describe the characteristics of a packet network connection. A VoIP dial peer, for example, describes the characteristics of an IP network connection. Voice-network dial peers point to a specific network device, such as the destination router that is connected to the remote telephony device.

For calls coming into the router from a POTS interface, the router matches a POTS dial peer for the incoming call leg and a VoIP dial peer for the outbound leg.

Other applications, such as store-and-forward fax and VoiceXML voice store and forward, also use the following type of dial peers:

- Multimedia Mail over IP (MMoIP)—Describe the line characteristics generally associated with a packet network connection. For example, this is the IP network connection between an on-ramp or off-ramp gateway and an SMTP server.

Figure 2 Dial Peer Call Legs



Voice Configuration Task Overview

This section outlines the basic steps required for configuring voice on a Cisco router or access server:

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- Step 1** Configure your IP network for real-time voice traffic. Fine-tuning your network to adequately support voice involves a series of protocols and features designed to improve quality of service (QoS). To configure your IP network for real-time voice traffic, consider the entire scope of your network. Then select and configure the appropriate QoS tool or tools.
- Refer to the [Quality of Service for Voice](#) document for information about how to select and configure the appropriate QoS tools to optimize voice traffic on your network.
- Step 2** Configure voice ports. In general, voice-port commands define the characteristics associated with a particular voice-port signaling type. The following voice signaling types are supported:
- FXO—Foreign Exchange Office interface
 - FXS—The Foreign Exchange Station interface
 - E&M—The “ear and mouth” interface (also called the “earth and magnet interface, or the “recEive and transMit” interface)
- Under most circumstances, the default voice-port command values are adequate to configure FXO and FXS ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice-port values configured, depending on the specifications of the devices in your telephony network.
- For information about configuring voice ports, refer to the [Voice Port Configuration](#) document, Cisco IOS Voice Configuration Library, Release 12.3.
- Step 3** Configure dial peers. Each dial peer defines the characteristics associated with a call leg and identifies call origin and destination. There are two types of dial peers for VoIP:
- POTS—To configure a POTS dial peer, you must configure the associated telephone number and the logical interface. Use the **destination-pattern** command to associate a telephone number with a POTS dial peer. Use the **port** command to associate a specific logical interface with a POTS dial peer.
 - VoIP—To configure a VoIP dial peer, you must configure the associated destination telephone number and a destination IP address. Use the **destination-pattern** command to define the destination telephone number associated with a VoIP peer. Use the **session target** command to specify a destination IP address for a VoIP peer.
- For additional information about dial-peer characteristics and configuring dial peers, refer to the [Dial Peer Configuration on Voice Gateway Routers](#) document, Cisco IOS Voice Configuration Library, Release 12.3.
- Step 4** Configure number expansion. Use the **num-exp** command to configure number expansion if your telephone network is configured so that you can reach a destination by dialing only a portion (an extension number) of the full E.164 telephone number. For information about number expansion, refer to the [Dial Peer Configuration on Voice Gateway Routers](#) document, Cisco IOS Voice Configuration Library, Release 12.3.
- Step 5** Optimize dial peer and network interface configurations. You can use VoIP dial peers to define characteristics such as codec, VAD, and additional QoS parameters (when RSVP is configured). If you have configured RSVP, use either the **req-qos** or **acc-qos** command to configure QoS parameters. Use the **codec** command to configure specific voice coder rates. Use the **vad** command to disable voice activation detection and the transmission of silence packets.

For additional information about optimizing dial-peer characteristics, refer to the [Dial Peer Configuration on Voice Gateway Routers](#) document, Cisco IOS Voice Configuration Library, Release 12.3.
