



Multiservice Applications Overview

The *Cisco IOS Multiservice Applications Configuration Guide* shows you how to configure your Cisco router or access server to support voice, video, and broadband transmission.

Configuration Guide Overview

The *Cisco IOS Multiservice Applications Configuration Guide* document is divided into three parts:

- Voice
- Video
- Broadband

Each part contains one or more chapters that describe configuration procedures for each respective technology. The following sections describe the chapter contents for each part of this configuration guide.

Voice

Cisco offers the following implementations of voice technology, depending on the particular Cisco device you are using:

- Voice over IP (VoIP). VoIP uses IP to carry voice traffic. Because voice traffic is being transported via IP, you need to configure signalling parameters as part of the voice-port configuration in addition to feature-specific elements such as dial peers. VoIP is compliant with International Telecommunications Union-Telecommunications (ITU-T) specifications H.323 and H.323 version 2.

VoIP can be used to provide the following:

- A central-site telephony termination facility for VoIP traffic from multiple voice-equipped remote office facilities.
- A Public Switched Telephone Network (PSTN) gateway for Internet telephone traffic. VoIP used as a PSTN gateway leverages the standardized use of H.323-based Internet telephone client applications. In the case of a device with extensive capacity running VoIP (such as the Cisco AS5800 universal access server), it provides the functionality of a carrier class switch.
- Voice over Frame Relay (VoFR). VoFR uses Frame Relay to transport voice traffic. Because VoFR is transporting signals over Layer 2, you need to configure timing parameters in addition to feature-specific elements such as dial peers and voice ports. VoFR is compliant with FRF.11 and FRF.12 specifications.

- Voice over ATM (VoATM). VoATM uses ATM adaptation layer 5 (AAL5) to route voice traffic. Because VoATM is transporting signals over Layer 2, you need to configure timing parameters in addition to feature-specific elements such as dial peers and voice ports.
- Voice over HDLC (VoHDLC). VoHDLC uses HDLC to transport voice traffic. VoHDLC is used primarily to transport back-to-back voice data in a LAN. Because VoHDLC is transporting signals over Layer 2, you need to configure timing parameters in addition to feature-specific elements such as dial peers and voice ports.

Dial Peers

The key to understanding the Cisco voice implementation is to understand the use of dial peers. Dial peers describe the entities to or from which a call is established. All voice technologies use dial peers to define the characteristics associated with a call leg. A call leg is a discrete segment of a call connection that lies between two points in the connection. Four call legs comprise an end-to-end call, two from the perspective of the source route, and two from the perspective of the destination route. You use dial peers to apply specific attributes to call legs and to identify call origin and destination. Attributes applied to a call leg include specific quality of service (QoS) features (such as IP RTP Priority and IP Precedence), compression/decompression (codec), voice activity detection (VAD), and fax rate.

There are basically two different kinds of dial peers with each voice implementation:

- “Plain Old Telephone Service” (POTS)—Dial peer describing the characteristics of a traditional telephony network connection. POTS peers point to a particular voice port on a voice network device.

When you configure POTS dial peers, the key commands that you must be configure are the **port** and **destination-pattern** commands. The **destination-pattern** command defines the telephone number associated with the POTS dial peer. The **port** command associates the POTS dial peer with a specific logical dial interface, normally the voice port connecting the Cisco device to the local POTS network.

Direct inward dial (DID) is configured on a POTS dial peer. In this case, the key commands that must be configured are the **destination-pattern** and **direct-inward-dial** commands.

Specific applications, such as interactive voice response (IVR), are configured on the POTS dial peer as well.

- Voice network (VoIP, VoATM, VoHDLC, and VoFR)—Dial peer describing the characteristics of a packet network connection; for example, in the case of VoIP, this is an IP network. Voice-network peers point to specific voice-network devices.

When you configure voice-network dial peers, the key commands that you must configure are the **destination-pattern** and **session-target** commands. The **destination-pattern** command defines the telephone number associated with the voice-network dial peer. The **session-target** command specifies a destination address for the voice-network peer.

Other applications (like store and forward fax, which uses the infrastructure of VoIP but is not strictly a voice technology) also use dial peers to assign attributes to call legs.

Voice Ports

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

Voice ports provide support for three basic analog voice signalling formats:

- **FXO**—Foreign Exchange Office interface. The FXO interface is an RJ-11 connector that allows a connection to be directed at the PSTN central office (or to a standard PBX interface, if the local telecommunications authority permits). This interface is of value for off-premises extension applications.
- **FXS**—The 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**—The “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 signalling technique for 2-wire and 4-wire telephone and trunk interfaces.

The Cisco MC3810 multiservice concentrator also supports E&M Mercury Exchange Limited channel-associated signalling (MEL CAS), which is primarily used in the United Kingdom.

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

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

The voice port syntax depends on the hardware platform on which it is being configured.

Voice Technologies

Cisco IOS Release 12.1 offers the following voice and voice-related technologies:

- Voice over IP
- Multimedia Conference Manager
- Voice over Frame Relay
- Voice over ATM
- Voice over HDLC
- Store and Forward Fax

Voice over IP

VoIP enables Cisco routers and access servers to carry voice traffic (for example, telephone calls and faxes) over an IP network. In VoIP, the digital signal processor (DSP) segments the voice signal into frames that are then coupled in groups of two and stored in voice packets. These voice packets are transported using IP in compliance with ITU-T specification H.323. Because VoIP is a delay-sensitive application, you need to have a well-engineered network end-to-end to successfully use VoIP.

Fine-tuning your network to adequately support VoIP involves a series of protocols and features geared toward QoS. Traffic shaping considerations must be taken into account to ensure the reliability of the voice connection.

Multimedia Conference Manager

The Multimedia Conference Manager provides both gatekeeper and proxy capabilities, which are required for service provisioning and management of H.323 networks. With Multimedia Conference Manager you can configure your current internetwork to route bit-intensive data such as audio, telephony, video and audio telephony, and data conferencing using existing telephone and ISDN links, without degrading the current level of service in the network. In addition, you can implement H.323-compliant applications on existing networks in an incremental fashion without upgrades.

With Multimedia Conference Manager, you can provide the following services:

- Identify H.323 traffic and apply appropriate policies
- Limit H.323 traffic on the LAN and WAN
- Provide user accounting for records based on service utilization
- Insert QoS for the H.323 traffic generated by applications such as VoIP, data conferencing, and video conferencing
- Implement security for H.323 communications

Voice over Frame Relay

VoFR enables a Cisco device to carry voice traffic (for example, telephone calls and faxes) over a Frame Relay network. When voice traffic is sent over Frame Relay, the voice traffic is segmented and encapsulated for transit across the Frame Relay network. The segmentation engine uses FRF.12 fragmentation. FRF.12 (also known as FRF.11 Annex C) allows long data frames to be fragmented into smaller pieces and interleaved with real-time frames. In this way, real-time voice and nonreal-time data frames can be carried together on lower speed links without causing excessive delay to the real-time traffic.

The segmentation size configured must match the line rate, or the port access rate. To ensure a stable voice connection, you must configure the same data segmentation size on both sides of the voice connection. When voice segmentation is configured, all priority queueing, custom queueing, and weighted fair queueing is disabled on the interface.

When you configure voice and data traffic over the same Frame Relay DLCI, you must take traffic shaping considerations into account to ensure the reliability of the voice connection.

Cisco VoFR implementation supports the following types of VoFR calls:

- Static FRF.11 trunks
- Switched VoFR calls:
 - Dynamic switched calls
 - Cisco-trunk (private line) calls

Voice over ATM

VoATM enables a Cisco MC3810 multiservice concentrator to carry voice traffic (for example, telephone calls and faxes) over an ATM network. The Cisco MC3810 multiservice concentrator supports compressed VoATM on ATM port 0 only.

When voice traffic is sent over ATM, the voice traffic is encapsulated using a special AAL5 encapsulation for multiplexed voice. The ATM permanent virtual circuit (PVC) must be configured to support real-time voice traffic, and the AAL5 voice encapsulation must be assigned to the PVC. The PVC must also be configured to support variable bit rate (VBR) for real-time networks for traffic shaping between voice and data PVCs.

Traffic shaping is necessary so that the carrier does not discard the incoming calls from the MC3810. To configure voice and data traffic shaping, you must configure the peak, average, and burst options for voice traffic. Configure the burst value if the PVC will be carrying bursty traffic. The peak, average, and burst values are needed so the PVC can effectively handle the bandwidth for the expected number of voice calls.

Voice over HDLC

VoHDLC enables a Cisco MC3810 multiservice concentrator to carry live voice traffic (for example, telephone calls and faxes) back-to-back to a second Cisco MC3810 multiservice concentrator. VoHDLC on the Cisco MC3810 multiservice concentrator is supported on serial ports 0 or 1, or on 0:x (the T1/E1 trunk, where *x* represents the channel group number). VoHDLC traffic is carried over a serial line. As a result, configuration is simpler than for VoIP, VoFR, or VoATM.

Store and Forward Fax

The Store and Forward Fax feature enables Cisco AS5300 access servers to send and receive faxes across packet-based networks. This feature is an implementation of the RFC 2305 proposed standard from the Internet Engineering Task Force (IETF), which is the same as the T.37 recommendation from the ITU. With Store and Forward Fax, your access server becomes a multiservice platform, supplying both data and fax communication. Because Store and Forward Fax uses the infrastructure of VoIP, it has been included in the *Cisco IOS Multiservice Applications Configuration Guide* as a voice-related technology.

Store and Forward Fax functionality is facilitated through the Simple Mail Transfer Protocol (SMTP). Additional functionality is provided to confirm delivery using existing SMTP mechanisms, such as Extended Simple Mail Transfer Protocol (ESMTP), for those features.

Video

Cisco supports video traffic within a data stream in three ways:

- Video in pass-through mode—Using this method, video traffic received from a video codec connected to a universal I/O serial port can be transported on a dedicated time slot between systems using the time-division multiplexing (TDM) functionality of the T1/E1 trunk.
- Video over ATM AAL1—A serial stream from a video codec connected to a serial port can be converted to ATM and transported across an ATM network using AAL1 circuit emulation service (CES) encapsulation.
- Video over ATM PVCs and switched virtual circuits (SVCs)—A serial stream from a video codec connected to a Cisco MC3810 multiservice concentrator using the plug-in Video Dialing Module (VDM) can be converted to ATM and transported across an ATM network using AAL1 CES.

Broadband

Cisco offers broadband Internet or intranet access and packet telephone services through a shared two-way cable system and IP backbone network. At the cable service provider side of the network, Cisco provides the Cisco uBR7200 series universal broadband routers. Cisco uBR7200 series universal broadband routers are Data-over-Cable Service Interface Specifications (DOCSIS)-based cable modem termination systems (CMTSS) that serve as interfaces between a WAN backbone and a hybrid fiber-coaxial (HFC) cable network. The Cisco uBR7200 series cable routers support both two-way and telephone return cable modems on a single downstream channel. As many as six upstream channels are supported on a single cable modem card, and as many as five cable modem cards can be installed in a single Cisco uBR7200 series chassis.

At the subscriber end of the network, Cisco provides the DOCSIS-based Cisco uBR900 series cable access routers. Cisco uBR900 series cable access routers provide the residential or small office, home office (SOHO) subscriber with high-speed Internet or intranet access with data and VoIP services over the HFC network.