Configuring Voice over IP

This chapter provides an overview of Voice over IP (VoIP) technology and gives step-by-step configuration tasks. The chapter contains the following sections:

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To identify the hardware platform or software image information associated with a feature in this chapter, use the Feature Navigator on Cisco.com to search for information about the feature or refer to the software release notes for a specific release. For more information, see the “Identifying Supported Platforms” in the “Using Cisco IOS Software” chapter.

Voice over IP Overview

VoIP is a Layer 3 network protocol that uses various Layer 2 point-to-point or link-layer protocols such as PPP, Frame Relay, or ATM for its transport. VoIP enables Cisco routers, access servers, and multiservice access concentrators to carry and send voice and fax traffic over an IP network. In VoIP, digital signal processors (DSPs) segment the voice signal into frames and store them in voice packets. These voice packets are transported via IP in compliance with a voice communications protocol or standard such as H.323, Media Gateway Control Protocol (MGCP), or Session Initiation Protocol (SIP). Table 3 shows the relationship between the Open System Interconnection (OSI) reference model and the protocols and functions of VoIP network elements.

Table 3 Relationship of OSI Reference Model to VoIP Protocols and Functions

<table>
<thead>
<tr>
<th>OSI Layer Number</th>
<th>OSI Layer Name</th>
<th>VoIP Protocols and Functions</th>
</tr>
</thead>
<tbody>
<tr>
<td>7</td>
<td>Application</td>
<td>NetMeeting/Applications</td>
</tr>
<tr>
<td>6</td>
<td>Presentation</td>
<td>Codecs</td>
</tr>
<tr>
<td>5</td>
<td>Session</td>
<td>H.323/MGCP/SIP</td>
</tr>
<tr>
<td>4</td>
<td>Transport</td>
<td>RTP/TCP/UDP</td>
</tr>
</tbody>
</table>
Cisco IOS software supports the following call control protocols and standards in Release 12.2:

- **H.323**—the International Telecommunication Union-Telecommunications Standardization Sector (ITU-T) specification for sending voice, video, and data across a network. The H.323 specification includes several related standards, such as H.225 (call control), H.235 (security), H.245 (media path and parameter negotiation), and H.450 (supplementary services). For more information, see the “H.323 Overview” chapter in this configuration guide.

- **MGCP**—Media Gateway Control Protocol, an Internet Engineering Task Force (IETF) draft standard for controlling voice gateways through IP networks. For more information, see the “Configuring MGCP and Related Protocols” chapter in this configuration guide.

- **SIP**—Session Initiation Protocol, defined in IETF RFC 2543. For more information, see the “Configuring SIP” chapter in this guide.

VoIP protocols typically use Real-time Transport Protocol (RTP) for the media stream or speech path. RTP uses User Datagram Protocol (UDP) as its transport protocol. Voice signaling traffic often uses Transmission Control Protocol (TCP) as its transport medium. The IP layer provides routing and network-level addressing; the data-link layer protocols control and direct the transmission of the information over the physical medium.

The main factor in choosing between VoIP and the Layer 2 VoFR and VoATM transport alternatives is interworking with other voice or multimedia applications. Generally speaking, Voice over Frame Relay (VoFR) and Voice over ATM (VoATM) are effective WAN transport technologies and are more bandwidth-efficient than VoIP. But VoFR and VoATM cannot be deployed over LANs or to the desktop. VoIP is the predominant form of voice-over-packet deployed today, and, for implementing voice applications, it is usually the only choice even if the first step in network deployment is pure transport between existing PBXs.

VoIP leverages the entire Internet and Intranet IP infrastructure for routing, making it easy to design any-to-any calling in a VoIP network. VoIP also allows multivendor interworking, which is more difficult to achieve with VoFR and VoATM applications because standards for those solutions have only recently emerged.

Cisco VoIP is frequently used in two primary applications:

- To provide a central-site telephony termination facility for voice traffic coming from multiple voice-equipped remote office facilities. Figure 2 illustrates this application using Cisco AS5300 universal access servers as the central-site telephony termination devices.
To provide Public Switched Telephone Network (PSTN) gateway functionality for Internet telephone traffic. Cisco VoIP used in this scenario 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), the functionality provided is equivalent to that of a carrier-class switch.

Figure 3 illustrates this application, using a Cisco AS5300 as the PSTN gateway.
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