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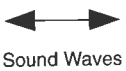
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destination. To support the transport of digital voice data over this type of network, the voice over IP device must take the voice data, encapsulate them into IP datagrams (packets), attach addressing information, and forward them into the network.

Transport

Intermediate nodes within the network inspect the addressing information attached to each of the IP datagrams and use that information to forward the datagram to the next hop along the path to its destination. The network links can be any topology or access method that supports IP traffic.

IP-to-Data Conversion

The destination voice over IP device receives the IP datagram and processes it. In processing the datagram, the addressing and control information is removed so that the original raw data remain. The raw data are then presented to the voice decoding process.

Conversion from Data Back to Voice

The voice decoding process interprets the raw data generated by the source station and runs them through the decoding function. The output from the decoding function is an analog signal resembling the original voice signal received by the source station line feed.

In summary, the transport of voice traffic over an IP network requires a conversion of the signal from analog to digital, packetization of the digital voice data, transport of the packetized information through the network, de-packetization of the voice data, and conversion of the digital voice data back to an analog signal. This process is depicted in Figure 2-2.

Voice over IP Basics

The previous example only dealt with the base functionality required to transport a voice signal across a simple data network. It lacked several

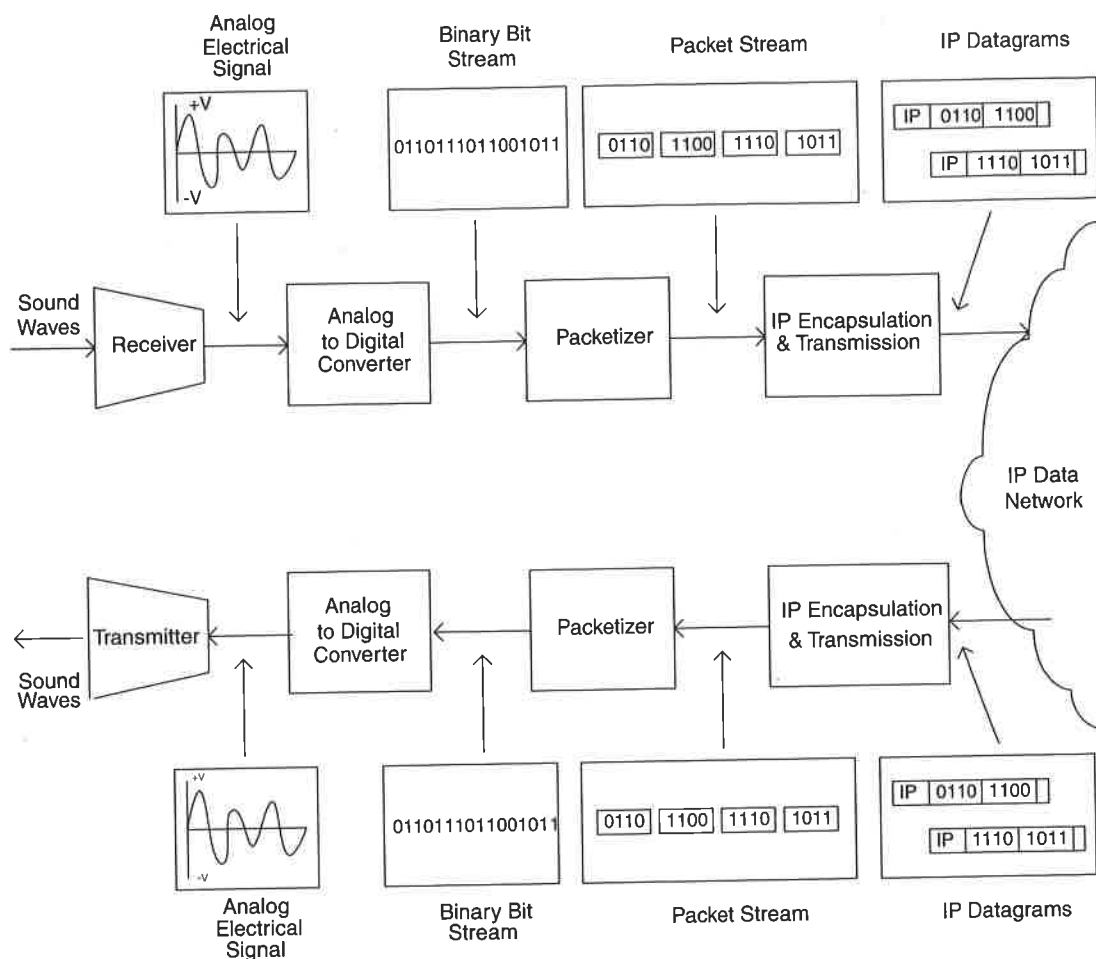


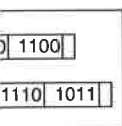
Figure 2-2 Simplified voice over IP transport process.

functions that are required to build a simple voice over IP network. The most basic are voice compression, phone number mapping, call-leg identification (peer establishment), and a protocol for the transport of voice data.

Voice Compression

In order to offset the overhead introduced by converting an analog signal to digital and transporting it over a packet switched network, compres-

Datagrams



Datagrams



IP network.
Mapping, call-leg
transport of

analog signal
work, compres-

sion algorithms can be used to reduce the amount of voice data that is transmitted over the packet switched network. Voice compression algorithms operate on the digital signal created by the initial analog to digital conversion and reduce the amount of bandwidth consumed by the voice signal by a factor of up to 8:1 or more. This reduction helps compensate for the addressing information applied (IP, UDP, and RTP headers) to the individual packets and allows for more voice calls to transit links with limited bandwidth.

Phone Number Mapping

The simple VoIP example above assumed that each voice over IP device would only transmit and receive calls from a single node. In more realistic environments, VoIP devices are required to dynamically form connections with multiple devices. In the voice world, destinations are selected using a dial string or phone number. In the IP world, destinations are selected using IP addresses. To allow a VoIP router to communicate with an array of other VoIP routers, a method of mapping the phone numbers dialed by the telephone user to the IP addresses used by the routers is required. This mapping can be performed statically within each router, or dynamically using Domain Naming System (DNS). The planning of connections and their associated phone numbers is called a *dial plan*.

Call Leg Identification

In order to properly identify, process, and forward voice calls, the router divides calls into discrete segments, referred to as call legs. Once identified, the router can apply services specific to each call leg, including terminating sessions.

Session Protocol

The Real Time Protocol (RTP) is a standardized way to transport voice and video traffic over IP networks. RTP adapts connectionless IP traffic to support real-time traffic through providing timing, sequencing, and other control functions. RTP is defined by an Internet Engineering Task Force (IETF) standard to ensure interoperability in multivendor environments.

Protocols

Real-Time Protocol/Real-Time Control Protocol

The Real-Time Protocol (RTP) and *Real-Time Control Protocol (RTCP)* were created to transport real-time traffic, such as video and audio, over the Internet and/or Internet Protocol-based networks. RTP is a session-layer protocol that usually runs on top of UDP/IP and is defined in RFCs 1889 and 1890. UDP is the favored transport-layer protocol because its connectionless nature enables its use in broadcast and multicast environments. RTP addresses some of the elements required for real-time traffic which are missing from UDP, including sequence numbering and time stamping. Sequence numbering enables the receiver to verify that datagrams have been received in order and that intermediate datagrams were not lost. Time stamping helps ensure proper playback rates regardless of datagram arrival times. Additionally, a payload-type field helps the receiving station identify which application or process to hand off the data to. Given this functionality Cisco and most of the VoIP community have chosen to use RTP for the transport of packetized voice in IP networks.

RTCP carries out the control functions for RTP streams. RTCP performs four discrete functions:

1. Communication of information and statistics about the RTP stream to the application
2. Identification of the RTP source
3. Limiting of control traffic
4. Secondary transport for small amounts of information

This protocol is important in voice environments for its ability to identify RTP sources and for the statistical information it provides applications.

Resource Reservation Protocol

Resource Reservation Protocol, or RSVP, is a protocol developed to allow applications to dynamically request specific levels of service from the network. RSVP's application-based approach enables it to be customized based on specific application requirements. RSVP-enabled applications communicate their bandwidth and network latency requirements to receiving applications. The receivers then issue RSVP requests to the network elements along the return path to the source station for these resources. Once a network element agrees to support a request, it is expected to honor that request for the duration of the reservation. In voice

protocol (RTCP) and audio, over P is a session-defined in RFCs because its broadcast environment-al-time traffic ring and time rify that data-grams were s regardless of eld helps the and off the data mmunity have P networks. as. RTCP per-

at the RTP

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over IP environments, a voice-enabled router can issue RSVP reservations for each individual voice call and rely on the intermediate network elements to dynamically allocate the appropriate resources for the call.

In addition to standard RSVP requests over packet networks, Cisco offers a means to map RSVP reservations to ATM SVCs called RSVP-ATM QoS interworking. This feature is supported by the 7500 series and enables the router to dynamically set up an ATM SVC with the appropriate QoS parameters to support the RSVP request. In hybrid packet and cell environments, RSVP-ATM QoS interworking provides enhanced QoS integration and leverages ATM's intrinsic QoS capabilities to support packet-based requests.

Diffserv/Committed Access Rate

Cisco's committed access rate (CAR) and the goals of the IETF's differentiated services (diffserv) working group are similar. The diffserv group is developing a standards-based approach to classifying IP traffic, setting bits within the IP header to identify a specific QoS, providing preferential service based on QoS levels, and supporting the QoS levels throughout the network. Cisco's committed access rate is an early implementation that provides rate control for IP traffic, interpretation and setting of IP precedence bits, and differentiated service throughout the network based upon IP precedence bits. This feature is significant in voice environments as it helps provide network-wide classes of service for IP traffic.

H.323

H.323 is an International Telecommunications Union (ITU) standard which was created in 1996 and updated in 1998. It provides a foundation for audio, video, and data communications across a packet-based network infrastructure. H.323 provides standards for voice-encoding, simple bandwidth management, admission control, address translation, call control and management, and links to external networks. The voice over IP community has adopted H.323 standards in an effort to foster the interoperability of equipment from multiple vendors. H.323 provides a solid foundation for building multimedia networks and has become critical for developing large-scale VoIP networks.

G.729/G.729a

G.729 and G.729a are standard for voice-encoding algorithms. These algorithms analyze standard PCM voice segments and use a complex algorithm to encode them. Once encoded, a lookup pointer for the audio

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Cisco Packetized Voice and Data Integration gives you the knowledge and insight to quickly get up to speed on planning, designing, and implementing Cisco-based voice and data networks. Through the use of real-world examples, case studies, and detailed configuration analyses, you will obtain the skills necessary to successfully deploy voice and data networks.

This essential sourcebook covers all topics intrinsic to the technology, including:

- Delivering QoS (quality of service) for both IP and frame relay
- Detailed discussions of Cisco's packetized voice-enabled products
- Leveraging IOS tools to implement voice over IP in diverse local and wide area Cisco networks
- Step-by-step instructions on integrating both voice and data in a Cisco-based network

In addition, an extensive quick reference section on the Cisco IOS command and an expansive listing of voice terminology puts the latest information at your fingertips. *Cisco Packetized Voice and Data Integration* is an ideal hands-on guide for any network administrator, engineer, or CCIE interested in using Cisco products for voice and data convergence.

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About the Author

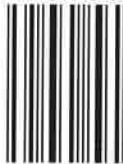
Robert Caputo, CCIE #1332, is a Lead Consultant in REALTECH Systems Corporation's Carrier/Service Provider group where he works with emerging service providers to plan, design, and build next generation multi-service networks. He actively consults on the deployment and implementation of Cisco-based products and has taught several seminars for Cisco Systems on voice and data convergence.

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