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28 May 2006 | SearchVoIP.com

By now, you probably have a general idea of what VoIP is, but do you understand how it works? Do you know how many bits are in a conventional PCM voice sample or what "look-ahead" used for? Do you know which two modes a caller can use to set up a call with SIP? If you don't know the answers to these questions, you will after reading SearchVoIP.com's technical guide to VoIP. Learn how call signaling and gateways work and why VoIP calls are particularly susceptible to delay and echo. Uncover the functional elements of SIP, H.323 and MGCP and why softswitches are not really switches at all.

After you've read through guide, test how much you learned in the Do you really know VoIP? quiz. Let us know how many questions you got right and if there is something you'd like to learn more about. In the spirit of education, we'll enter your name in a drawing for a $50 gift certificate to Amazon.com. Contest ends July 31, 2006, but don't let that stop you from testing your VoIP acumen.

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Roger Freeman is a widely recognized expert in telecommunications system engineering. Roger has worked in telecommunications since 1946 when he joined the Navy and became an aviation radioman. Later, Roger served as a radio officer in the merchant marine for nearly 10 years. He then held several positions with ITT assigned to their Spanish Standard Electrica subsidiary. He also served the International Telecommunication Union as Regional Planning Expert for Northern Latin America based in Quito, Ecuador. Roger is bilingual. His last employee position was principal engineer with the Raytheon Company, Marlboro, MA where he took early retirement in 1991 to establish Roger Freeman Associates, Independent Consultants in Telecommunications. He has been giving seminars in telecommunication disciplines at the University of Wisconsin, Madison for nearly 20 years. Roger has been writing books on various telecommunication subjects for John Wiley & Sons since 1973. There are seven titles, which he keeps current including the two-volume work, Reference Manual for Telecommunication Engineers, now in 3rd edition. He holds two degrees from NYU. His Web site is www.rogerfreeman.com and his e-mail address is rogerf67@cox.net.

Here are Roger's landmark books:

- Radio System Design For Telecommunications, 2nd Ed. Wiley NY - 1997
- Reference Manual For Telecommunications Engineering, 3rd Ed. Wiley NY - 2001
- Telecommunication System Engineering, 4th Ed. Wiley NY - 2004
- Telecommunications Transmission Handbook, 4th Ed. Wiley NY - 1998
- Practical Data Communications, 2nd Ed. Wiley NY - 2001
- Fundamentals Of Telecommunications, 2nd Ed, Wiley NY - 2005
- Fiber Optic Systems For Telecommunications, Wiley NY - 2002

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Conventional voice telephony is transported in a full duplex mode on PSTN circuits optimized for voice. By the "full duplex mode" we mean that there are actually two circuits -- one for send and one for receive -- to support a normal telephone conversation between two parties. Today, once we depart the local area, all of these circuits are digital. The descriptive word "digital" may seem ambiguous to some.

Let's say that 10 years ago we looked ahead to our present time. All the circuits would consist of 8-bit words, which represent voltage samples of analog voice conversations in a PCM format. This is often characterized in the literature as G.711 service (i.e., ITU-T Rec. G.711). Data is also commonly transported in 8-bit sets called "bytes," but more properly called "octets." It is comparatively simple to replace 8-bit voltage samples of voice with 8-bit octets of data.

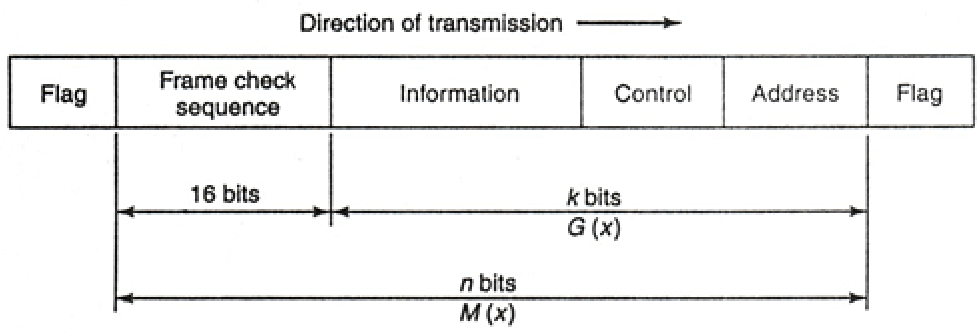
Essential philosophical differences remain, however, between voice in 8-bit octets and data transmission. A voice circuit is established when a subscriber desires to converse by telephone with some other telephone subscriber. The circuit between the two is set up by a signaling routine. The distant subscriber has a telephone address represented by a distinct telephone number consisting of between 7 and 12 digits. The digit sequence of the number sets up a circuit route and connectivity for conversation. The circuit is maintained in place for the duration of the conversation. It is terminated and the circuit is taken down when one or other party hangs up -- goes on-hook. The address sequence of dialed digits is sent just once, at the initiation of the connectivity. This whole process of setting up a circuit, holding the connectivity in place, and then taking down the circuit is called "signaling."

Signaling on data circuits is approached quite differently. There is the "permanent virtual circuit" (PVC), which has all the trappings of a voice circuit. The similarities stop here. Data transmission consists of frames or packets of data. A frame (or data packet) is made up of a "header" and payload. In some cases, a portion of the header may be appended at the end or on the tail of the data frame. But every data frame, at a minimum, has a header consisting of a destination address(es) and the originator's address. It will also nearly always contain some control information. This may be a word (or octet) count of the payload, a CRC sequence for error detection and/or correction, message priority, and/or some other type of control sequence(s).

Digital circuits on the PSTN have either 24-octet frames in the case of T1 (DS1) or 32-octet frames in the case of E1. Each 8-bit octet represents a voice circuit. Such a circuit may be set up using an Initial Address Message of CCITT Signaling System No. 7 or a sequence of DTMF tones where each frequency pair represents a digit in the range of 0 through 9. Once a circuit is set up, no more address messages or DTMF tones are required until the circuit is taken down.

This is not the case on a data circuit. Such a circuit also uses frames, but each frame has a standard header. A typical data frame is illustrated in Figure 1. The frame structure and how the various octets of the header (and tail) are utilized are governed by a "protocol." Various data protocols are discussed in *Practical Data Communications, 2nd ed., Wiley, N.Y., 2001.*

We can clearly see that there are two differing philosophies here, one for data communication, and the other for digital voice. Digital voice is sometimes called "circuit-switched voice." A majority in the telecommunication community saw how advantageous it would be if we could marry the two philosophies. That is, one singular approach for both voice and data. Meanwhile, data hobbyists were trying to transmit voice using data packets. The Internet protocol (IP) became the data protocol of choice, but there were many drawbacks.



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We have come to measure quality of packetized voice service by the equivalent service offered by the switched digital network, sometimes called G.711 (Ref. 1) service. The user expects a quality of service (QoS) as good as he/she would get on a PSTN dial-up connection.

To achieve this goal, VoIP designers were faced with the following degradations:

- Mouth-to-ear delay;
- Impact of erred frames (packets);
- Lost frames (packets);
- Variation of packet arrival time, jitter buffering;
- Prioritizing VoIP traffic over regular Internet and data services;
- Talker echo;
- Distortion;
- Sufficient bit rate capacity on interconnecting transmission media;
- Voice coding algorithm standardization;
- Optimized standard packet payload size;
- Packet overhead; and
- Silence suppression.

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Figure 2 shows a simplified block diagram of VoIP operation from an analog signal deriving from a standard telephone, which is digitized and transmitted over the Internet via a conversion device. Then, at the distant end, it is converted back to analog telephony using a similar device suitable for input to a standard telephone. A "gateway" is placed between the voice codec and the digital data transport circuit. An identical device will also be found at the far end of the link. This equipment carries out the signaling role on a telephone call, among other functions.

Moving from left to right in Figure 2, we have the spurty analog voice signal developed by the standard telephone. The signal is then converted to a digital counterpart using one of the seven or so codecs from which the VoIP system designer has to select. Some of the more popular codecs for this application are listed in Table 1. The binary output of the codec is then applied to a conversion device (i.e., a packetizer) that loads these binary 1's and 0's into an IP payload between 20 and 40 octets in length.

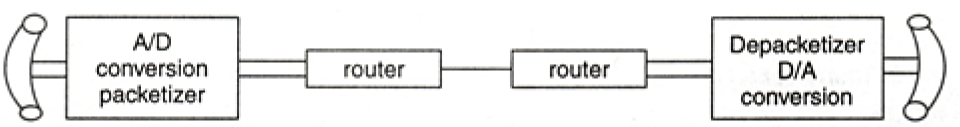

Figure 2. Elements of basic operation of VoIP, where the input signal derives from a conventional analog telephone

The output of this converter consists of IP packets* that are transmitted on the Web or other data circuit for delivery to the distant end.

At the far end, the IP packets are input to a converter that strips off the IP header, stores the payload, and then releases it in a constant bit stream to a codec. Of course, this codec must be compatible with its near-end counterpart. The codec converts the digital bit stream back to an analog signal that is input to a standard telephone.

The insightful reader will comment that many steps of translation and interface have been left out. Most of these considerations will be covered in Section 3.1, in our discussion of the "gateway." **The output may be ATM cells (See Chapter 16, Telecommunication System Engineering, 4th ed, Wiley, N.Y., 2004) if the intervening network is an ATM network.*

VoIP gateway

Gateways are defined in different ways by different people. A gateway is a server; it may also be called a "media gateway." Figure 3 illustrates a typical gateway. It sits on the edge of the network and carries out a switching function of a local, tandem or toll-connecting PSTN switch.

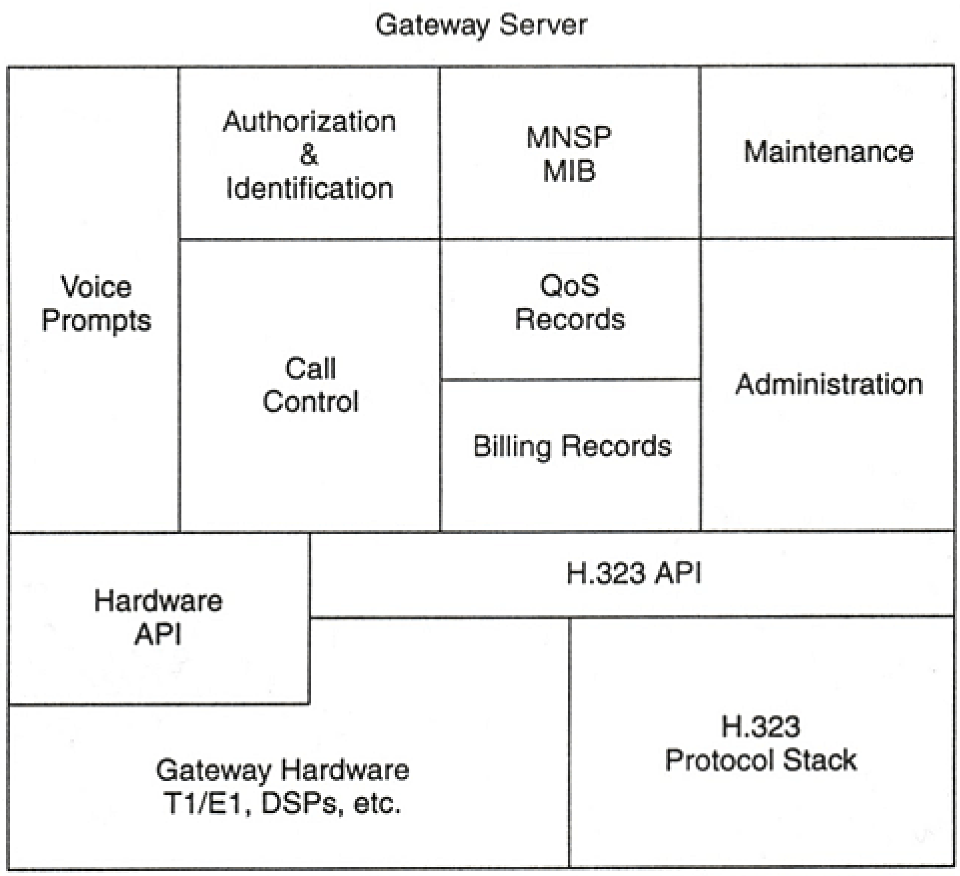

Figure 3. A media gateway, from one perspective. API = applications program interface. From IEC online (Jan. 2003)

Media gateways are part of the physical transport layer. They are regulated by a call control function housed in a media gateway controller. A media gateway, with its associated gateway controller, is necessary for the network transformation to packetized voice. Several of the media gateway functions are listed below:

- Carries out A/D conversion of the analog voice channel (called compression in many texts);
- Converts a DS0 or E0 to a binary signal compatible with IP or ATM;
- Supports several types of access networks, including media such as copper (including various DSL regimes), fiber, radio (wireless), and CATV cable. It is also able to support various formats found in PDH and SDH hierarchies;
- Competitive availability (99.999%);
- Capable of handling several voice and data interface protocols;
- Multi-vendor interoperability;
- It must provide interface between the media gateway control device and the media gateway. This involves one of four protocols: SIP (Ref. 2), H.323 (Ref. 3), MGCP and Megaco (H.248);
- It can handle switching and media processing based on standard network PCM, ATM and traditional IP; and
- Transport of voice. There are four transmission categories that may be involved:
    1. Standard PCM (E0/E1 or DS0/DS1)
    2. ATM over AAL1/AAL2
    3. IP-based RTP/RTCP
    4. Frame relay

Table 1 – Characteristics of speech codecs used on packet networks

Coding algorithm	Voice bit rate (Kbps)	Voice frame size (bytes)	Header (bytes)	Packets per second	Packet bit-rate (Kbps)

G.711 8-bit PCM (Ref. 1)	64	80	40	100	96
G.723.1 MPMLQ(1) (Ref. 4)	6.3	30	40	26	14.6
G.723.1 ACELP(2) (Ref. 5)	5.3	30	40	22	12.3
G.726 ADPCM (3) (Ref. 6)	32	40	40	100	64
G.728 LD-CELP(4) (Ref. 7)	16	20	40	100	48
G.729a CS-ACELP (5)(Ref. 8)	8	10	40	100	40

Notes:
(1) MPMLQ – Multi-pulse Maximum Likelihood Quantization
(2) ACELP -Algebraic Code-Excited Linear Prediction
(3) ADPCM – Adaptive Differential PCM
(4) LD-CELP – Low Delay Code-Excited Linear Prediction
(5) CS-ACELP – Conjugate Structure Algebraic Code-Excited Linear Prediction

The most powerful gateway supports the PSTN, requiring a high-reliability device to meet the PSTN availability requirements. It will be required to process many thousands of digital circuits. As shown in Figure 3, it has a network management capability most often based on simple network management protocol (SNMP -- see Chapter 21 of *Telecommunication System Engineering*, 4th ed., Wiley, N.Y.).

A somewhat less formidable gateway is employed to provide VoIP for small and medium-sized businesses. Some texts call this type of gateway an "integrated access device" (IAD) if it can handle data and video products as well. An IAD will probably be remotely configurable.

The least powerful and most economic gateways are residential. They can be deployed in at least five settings:

- POTS (telephony);
- Set-top box (CATV), which provides telephony as well;
- PC/modem;
- XDSL termination; and
- Broadband last-mile connectivity (to the digital network).

Figure 4 shows gateway interface functions via a block diagram. On the left are time slots of a PCM bit stream (T1, in this case). The various signal functions are shown to develop a stream of data packets carrying voice or data. The output on the right consists of IP packets.

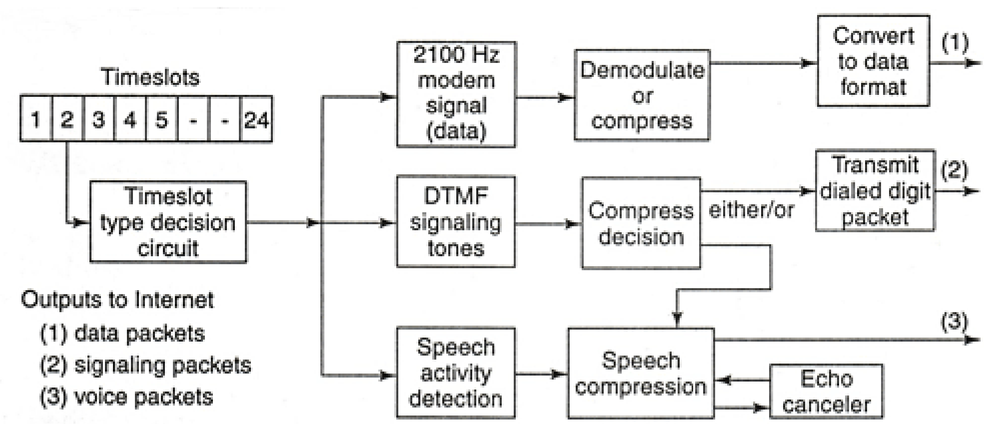

Figure 4. A simplified functional block diagram of a gateway providing an interface between a PCM bit stream deriving from the PSTN on the left and an IP network. The first functional block of the gateway analyzes the content on a time-slot basis. The time slot may contain an 8-bit data sequence where we must be hands-off regarding the content. A gateway senses the presence of data by the presence of a 2100 Hz tone in the time slot. The next signal type in the time slot it looks for is DTMF signaling tones (see Chapter 4). If there is no modem tone or DTMF tones in the time slot, the gateway assumes the time slot contains human speech. Three actions now have to be accomplished. "Silence" is removed, the standard PCM compression algorithm is applied, and an echo canceller is switched in. There are three digital formats used for voice over packet:

1. IP (Internet protocol, Chapter 11, Section 7);
2. Frame relay (Chapter 15); and

3. ATM (asynchronous transfer mode, Chapter 16).

Reference: Telecommunication System Engineering, Wiley, N.Y., 2004.

An IP packet, as used for VoIP
Assume for argument's sake that we use either a G.711 or G.726 IP packet. The packet consists of a header and a payload. Figure 5 shows a typical IP packet. Of interest, as one may imagine, is its payload.

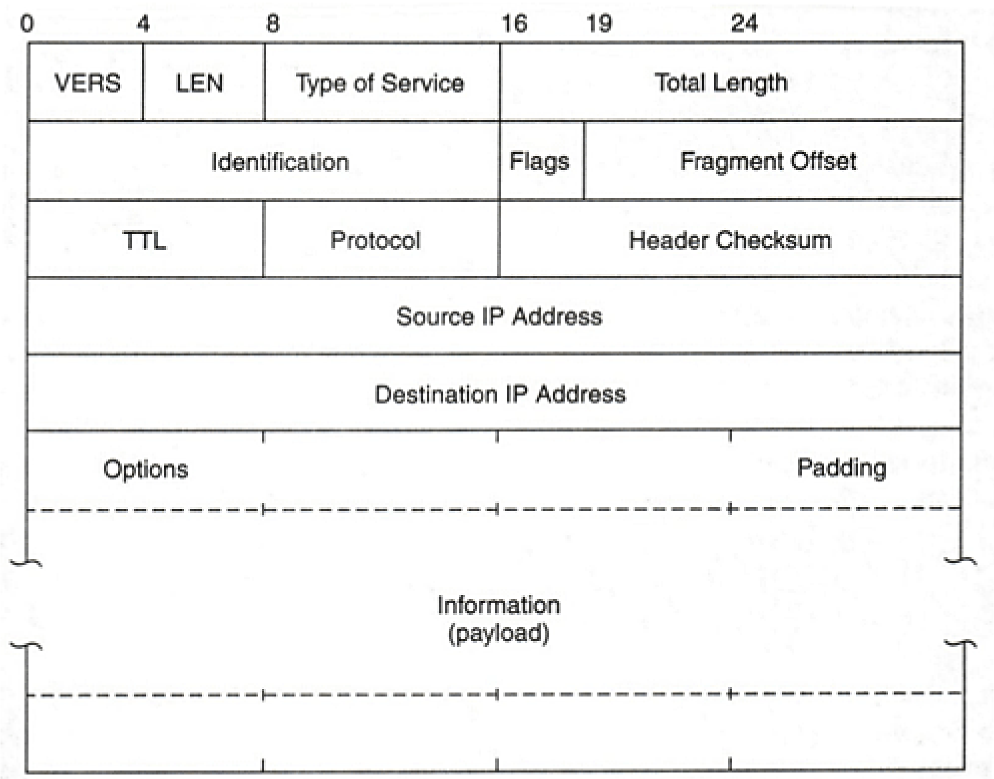

Figure 5. A typical IP packet. Based on RFC 791, Ref. 22. (also see Figure 11.28).

In the case of G.711, standard PSTN PCM, there may be a transmission rate of 100 packets per second with 80 bytes in the payload of each packet. Of course, our arithmetic comes out just right and we get 8,000 samples per second, the Nyquist sampling rate for a 4 KHz analog voice channel. Another transmission rate for G.711 is 50 packets per second, where each packet will have 160 bytes, again achieving 8,000 samples per second per voice channel.

The total raw bytes per channel come out as follows: Layers 3 and 4 overhead (IP): 40 bytes plus 8 bytes for Layer 2 (link layer) overhead. So we add 48 to 80 or 160 bytes and get 128 or 208 bytes for a raw packet. The efficiency is nothing to write home about. Keep in mind that the primary concern of the VoIP designer is delay.

The delay tradeoff
Human beings are intolerant of delay on a full-duplex circuit, typical of standard PSTN telephony. ITU-T Rec. G.114 (Ref. 10) recommends the total delay -- one-way -- in voice connectivity as follows:

- 0-150 msec acceptable;
- 150-400 msec acceptable but not desirable. Connectivity through a geostationary satellite falls into this category; and, above 400 msec, is unacceptable.

The delay objective -- one-way -- for a VoIP voice connectivity is less than 100 msec. With bridging for conference calls, that value doubles, owing to the very nature of bridging.
One-way components of delay are as follows:

- Packetization or encapsulation delay based on G.711 or other compression algorithm. In the case of G.711, we must build from 80 PCM samples at 125 µsec per sample, so we have consumed 80 x 125 µsec, or 10,000 msec, or 10 msec plus time for the header, or 48 x 125 µsec, or 6 msec, for a total of 16 msec. If we use 160 PCM samples in the payload, then allow 20 msec plus 6 msec for the header, or 26 msec. This is a fixed delay.
- Buffer delay is variable. As a minimum, there must be buffering of one frame or packet period. By definition, routers have buffers. Buffer delay varies with the number of routers in tandem. For G.711, the

packet buffer size is 16 or 26 msec.
- Look-ahead delay: This is used by the coder to help in compression. Look-ahead is a period of time when the coder looks at packet n+1 for patterns on which it can compress while coding packet n. With G.711, the look-ahead is 0.
- De-jitterizer: This is a buffer installed at the destination. It injects at least 1 frame duration (1-20 msec) in the total delay to smooth out the apparent arrival times of packets.
- Queuing delay: This is time spent in the queue because it is a shared network. One method to reduce this delay is to prioritize voice packets over data, with an objective of less than 50 msec.
- Propagation delay: Variable. Major contributor to total delay. Geostationary satellite relay of circuits is a special problem. The trip to the satellite and back is budgeted at 250 msec.

One way to speed things up is to increase the bit rate per voice data stream. To do this, the aggregate bit rate may have to be increased or the number of voice streams may be reduced on the aggregate bit rate so that each stream can be transmitted at a faster rate.

Lost packet rate
A second concern of the VoIP designer is *lost packet rate*. There are several ways a packet can be lost.

For example, Section 3.3 described a de-jitterizing buffer. It has a finite size. Once the time is exceeded by a late packet, the packet in question is lost. In the case of G.711, this would be the time equivalent to 16 or 26 msec -- duration of a packet including its header. Another cause of packet loss may be excessive error rate on a packet, whereby it is deleted. When the lost or discarded packet rate begins to exceed 10%, quality of voice starts to deteriorate. If high-compression algorithms -- such as G.723 or G.729 -- are employed, it is desirable to maintain the packet loss rate below 1%. Router buffer overflow is another source of packet loss.

IP through TCP has excellent retransmission capabilities for erred frames or packets, but they are not practical for VoIP because of the additional delay involved. When there is a packet in error, the receiving end of the link transmits a request (RQ) to the transmitting end for a packet retransmission and its incumbent propagation delay. This must be added to the transmission delay with some processing delay to send the offending packet back to the receiver again.

Concealment of lost packets
A lost packet causes a gap in the reception stream. For a single packet, we are looking at a 20 to 40 msec gap. The simplest measure to take for lost packets and the resulting gaps is to disregard them. The absolute silence of a gap may disturb a listener. In this case, artificial noise is often inserted.

There are packet loss concealment (PLC) procedures that can camouflage gaps in the output voice signal. The simplest techniques require a little extra processing power, and the most sophisticated techniques can restore speech to a level approximating the quality of the original signal. Concealment techniques are most effective for about 40 to 60 msec of missing speech. Gaps longer than 80 msec usually have to be muted.

One of the most elementary PLCs simply smooths the edges of gaps to eliminate audible clicks. A more advanced algorithm replays the previous packet in place of the lost one, but this can cause harmonic artifacts such as tones or beeps. Good concealment methods use variation in the synthesized replacement speech to make the output more like natural speech. There are better PLCs that preserve the spectral characteristics of the talker's voice and maintain a smooth transition between estimated signal and surrounding original. The most sophisticated PLCs use CELP (codebook-excited linear predictive) or a similar technique to determine the content of the missing packet by examining the previous one (Ref. 11). Lost packets can be detected by packet sequence numbering.

Echo and echo control
Echo is commonly removed by the use of echo cancellers that are incorporated on the same DSP chips that perform the voice coding. A good source for information and design of echo cancellers is ITU-T Rec. G.168 (Ref. 12). However, most vendors of VoIP equipment have their own proprietary designs. A common design approach is to have the echo canceller store the outgoing speech in a buffer. It then monitors the stored speech after a delay to see whether it contains a component the matches up against the stored speech after a delay. If it does, that component of the incoming speech is cancelled out instead of being passed back to the user, since it is an echo of what the user originally said. Echo cancellers can be tuned or can tune themselves to the echo delay on any particular connection. Each echo canceller design has a limit to the maximum delay of echo it can identify. Echo cancellers are bypassed if a fax signal or modem data is on the line.

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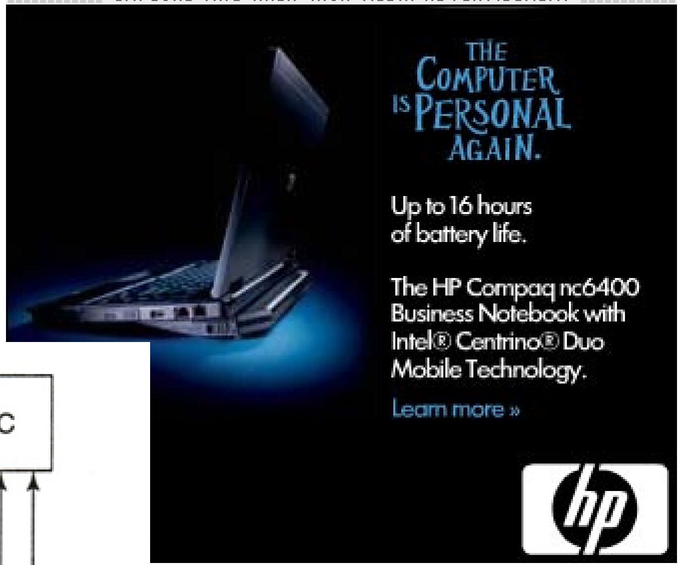

4. **Media gateway controller and its protocols**
The gateway controller or media gateway controller (MGC) carries out the signaling function on VoIP circuits. Some texts call an MGC a *softswitch*, even though they are not truly switches but servers that control gateways. This function is illustrated in Figure 6.

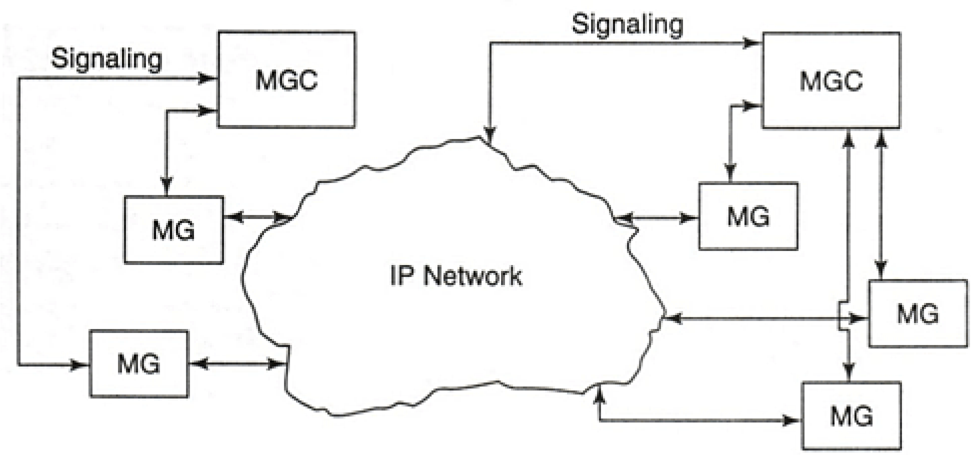

Figure 6. A media gateway controller (MGC) provides a signaling interface for media gateways (MGs), thence to the IP network.

An MGC can control numerous gateways, but to improve reliability and availability, several MGCs may be employed in separate locations with function duplication on the gateways they control. Thus, if one MGC fails, others can take over its functions. We must keep in mind that the basic topic of Section 4 of this chapter is signaling -- that is, establishing telephone connectivity, maintaining that connectivity, and taking down the circuit when the users are finished with conversation. There is a basic discussion of signaling in Chapter 4 of this text.

There are four possible signaling protocol options between an MGC and gateways. These are:

- 1. ITU-T Rec. H.323. This is employed where all network elements (NEs) have software intelligence.
- 2. SIP (Session Initiation Protocol, Ref. 2) is used when the end devices have software intelligence and the network itself is without such intelligence.

- 3. MGCP (media gateway control protocol) is another gateway control protocol.
- 4. MEGACO (ITU-T Rec. H.248, Ref. 13) is a gateway control protocol applicable when end devices are without software intelligence and the network has software intelligence.

Overview of the ITU-T Rec. H.323 standard
In May 1996, the ITU ratified the H.323 specification, which defines how voice, video and data traffic should be transported over IP-based LANs. It also incorporates the ITU-T Rec. T.120 (Ref. 21) data-conferencing standard. The H.323 recommendation is based on RTP/RTCP (real-time protocol/real-time control protocol) for managing audio and video signals.

What sets H.323 apart is that it addresses core Internet applications by defining how delay-sensitive traffic such as voice and video get priority transport to ensure real-time communication service over the Internet. Related protocols are ITU-T Rec. H.324 (Ref. 14) specification, which defines the transport of voice, data and video over regular telephone networks. Another related protocol is ITU-T Rec. H.320 (Ref. 15), which covers the transport of voice, video and data over the integrated services digital network (ISDN).

H.323 deals with three basic functional elements of VoIP. These are:

- Media gateway;
- Media gateway controller (MGC) -- in some settings this is called the *gatekeeper*; and
- Signaling gateway.

H.323 is an umbrella protocol covering:

- H.225 (Ref. 16), which covers the setup of multimedia channels and,
- H.245 (Ref. 17), which deals with the setup of single channel medium.

The standard H.323 (Ref. 3) prefers the use of the term gatekeeper to media gateway controller. Some of the more important responsibilities of a gatekeeper are:

- Security: It authenticates users of the H.323 network;
- It performs address translation between Internet addresses and ITU-T Rec. E.164 (Ref. 18) addresses;
- It polices the capacity of the network in question -- whether that network can accept another call;
- H.323 determines call routing -- to route through a gateway or be sent directly to the destination; and
- It keeps track of the network's bit rate capacity.

H.323 assumes that the transmission medium is a LAN that does not provide guaranteed delivery of packets. In the ITU H.323 standard we will find the term entity. An entity carries out a function. For example, a terminal is an endpoint on a LAN that can support real-time communications with another entity on that LAN. It has a capability provided by a voice or audio codec such as a G.711 or G.728 codec. It will also provide a signaling function for VoIP circuit setup, maintain and take-down. A VoIP terminal optionally can support video and data streams, including compression and decompression of these streams. Media streams are carried on RTP or RTCP. RTP deals with media content; RTCP works with the signaling functions of status and control. This protocol information is embedded in UDP, which is reliably transported by TCP. Other VoIP entities are gateways, and there is an optional gatekeeper.

The leading issue in VoIP implementation is guaranteed quality of service (QoS). H.323 is based on RTP, which is comparatively new. RTP-compliant equipment includes control mechanisms for synchronizing different traffic streams. On the other side of the coin, RTP has no mechanisms for ensuring on-time delivery of traffic signals or for recovering lost packets. It does not address the QoS issue related to guaranteed bit rate availability for specific applications. The IEC (Ref. 19) reports that there is a draft signaling proposal to strengthen the Internet's ability to handle real-time traffic reliably. This would dedicate end-to-end transport paths for specific sessions, much as the circuit-switched PSTN does. This is the resource reservation protocol (RSVP). It will be implemented in routers to establish and maintain requested transmission paths and QoS levels.

SIP
SIP (Session Initiation Protocol) is based on RFC 2543 (Ref. 3) and is an application layer signaling protocol. It deals with interactive multimedia communication sessions between end users, called *user agents*. It defines their initiation, modification and termination. SIP calls may be terminal-to-terminal, or they may require a server to intercede. If a server is to be involved, it is only required to locate the called party. For interworking with non-IP networks, Megaco and H.323 are required. Often, vendors of VoIP equipment integrate all three protocols on a single platform.

SIP is closely related to IP. SIP borrows most of its syntax and semantics from the familiar HTTP (hypertext transfer protocol). A SIP message looks very much like an HTTP message, especially with message formatting, header and multipurpose Internet mail extension support. It uses addresses that are very similar to URLs and to email. For example, a call may be made to so-and-so@such-and-such. SIP messages are text-based rather than binary. This makes writing easier and the debugging of software more straightforward.

There are two modes with which a user can set up a call with SIP. These are called *redirect* and *proxy*, and servers are designed to handle these modes. Both modes issue an invite message for another user to participate in a call. The redirect server is used to supply the address (URL) of an unknown called addressee. In this case, the "invite" message is sent to the redirect server, which consults the location server for address information. Once this address information is sent to the calling user, a second invite message is issued, now with the correct address.

One specific type of SIP is called SIP-T (T for telephone). This is a function that allows calls from CCITT Signaling System 7 (SS7) to interface with telephone in an IP-based network. The particular user part of SS7 for this application is ISUP.

Media gateway control protocol (MGCP)
This protocol was the predecessor to Megaco (see Section 12.3.4) and still holds sway with a number of carriers and other VoIP users. MGCP (Ref. 20) assumes a call-control architecture where the call-control intelligence is outside the gateways (i.e., at the network edge) and handled by external call-control elements. Thus, the MGCP assumes that these call-control elements, or call agents, will synchronize with each other to send coherent commands to the gateways under their command. There is no mechanism defined in MGCP for synchronizing call agents. It is, in essence, a master/slave protocol where the gateways are expected to execute commands sent by the call agents.

In the MGCP protocol, an assumption is made that the connection model consists of constructs that are basic endpoints and connections. Endpoints are sources or sinks of data and could be physical or virtual. The following are two examples of endpoints:

- (1) An interface on a gateway that terminates a trunk connected to a PSTN switch (e.g., local or toll-connecting, etc.). A gateway that terminates trunks is called a trunk gateway.
- (2) An interface on a gateway that terminates an analog POTS (plain old telephone service) connection to a telephone, a key system, PABX, etc. A gateway that terminates residential POTS lines (to telephones) is called a residential gateway.

An example of a virtual endpoint is an audio source in an audio-content server. Creation of physical endpoints requires a hardware installation, whereas creation of virtual endpoints can be done in software (Ref. 20).

Megaco or ITU-T Rec. H.248 (Ref. 13)
Megaco is a call-control protocol that communicates between a gateway controller and a gateway. It evolved from and replaces SGCP (simple gateway control protocol) and MGCP (media gateway control protocol). Megaco addresses the relationship between a media gateway (MG) and a media gateway controller (MGC). An MGC is sometimes called a *softswitch* or *call agent*. Both Megaco and MGCP are relatively low-level devices that instruct MGs to connect streams coming from outside the cell or packet data network onto a packet or cell stream governed by RTP.

A Megaco (H.248) connection model is illustrated in Figure 12.7. Two principal abstractions relate to the model: terminations and contexts. A termination sources and/or sinks one or more data streams. In a multimedia conference, a termination can be multimedia, and sources and sinks, multiple media streams. The media stream parameters, as well as modem and bearer parameters, are encapsulated within the termination.

A context is an association among a collection of terminations. There is a special type of context, called the *null context*, which contains all terminations that are not associated with any other termination. For example, in a decomposed access gateway, all idle lines are represented by terminations in the null context.

Let's look at three context possibilities.

- 1) A context with just one termination is call waiting. The caller does not hear anyone else.
- 2) A context with two terminations is a regular phone call. Of course, each person is expected to hear the other.
- 3) An example of more than two terminations is a conference call. All parties hear all of the others.

The maximum number of terminations in a context is a media gateway (MG) property. MGs that offer only point-to-point connectivity might allow at most two terminations per context. MGs that support multi-point conferences might allow three or more terminations per context.

The attributes of contexts are:

- Context ID;
- The topology -- who hears/sees whom;
- The topology of a context describes the flow of media between the terminations within a context. In contrast, the mode of a termination (send/receive) describes the flow of the media at the ingress/egress of the media gateway.
- The priority is used for a context in order to provide the MG with information about a certain precedence handling for a context. The MGC can also use the priority to control autonomously the traffic precedence in the MG in a smooth way in certain situations (e.g., restart), when a lot of context must be handled simultaneously; and
- An indicator for an emergency call is also provided, to allow preference handling in the MG.

Megaco uses a series of commands to manipulate terminations, contexts, events and signals. For example, the *add command* adds a termination to a context and may be used to create a new context at the same time. Of course, we would expect the *subtract command* to remove a termination from a context and may result in the context's being released if no terminations remain.

There is also the *modify command*, used to modify the description of a termination -- e.g., the type of voice compression in use. *Notify* is used to inform the gateway controller if an event occurs on a termination such as a telephone in an off-hook condition or digits being dialed. There is also a *service change command*.

Terminations are referenced by a TerminationID, which is an arbitrary schema selected by the MG. TerminationIDs of physical terminations are provisioned by the media gateway. The TerminationIDs may be chosen to have structure. For example, a TerminationID may consist of a trunk group and a trunk within the group (Ref. 9).

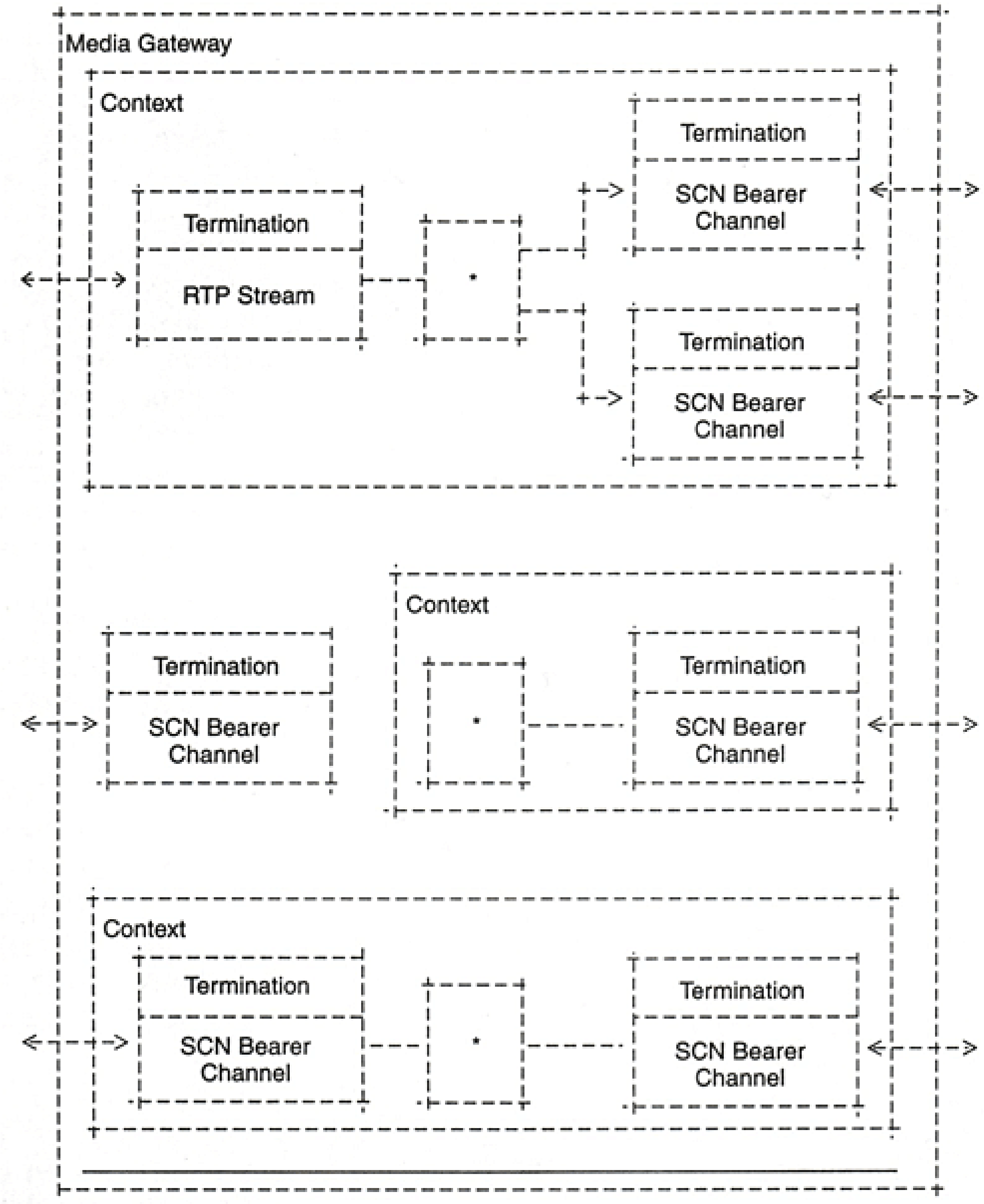

Figure 7. An example of H.248/Megaco connection model. SCN (switched circuit network). The asterisk in each box in each of the contexts represents the logical association of terminations implied by the context (based on Figure 1, RFC 3015, Ref 9).

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1. How many bits are in a conventional PCM voice sample? Answer
2. How is a telephone call on the PSTN usually terminated? Answer
3. Give the two basic components of a data message? Answer
4. Every data frame or packet carries two components (at least) in the header. What are they? Answer
5. In question 4, how does this differ from a PCM frame? Answer
6. What is a CRC sequence used for? Answer
7. Give at least two different ways the PSTN may setup a voice circuit. Answer
8. List at least five drawbacks or challenges designers faced in making VoIP a reality. Answer
9. List at least four tasks the gateway may do. Answer
10. List at least four transmission media and formats that a media gateway should be able to support. Answer
11. A media gateway interfaces a PSTN PCM bit stream. Give three types of traffic that may be transported in that bit stream, each requiring some sort of special handling. Answer
12. Given an IP packet used for voice transport. Assume G.711 or G.726 coders. How many bytes (octets) can we expect in the payload? Answer
13. Give some data you have learned about one-way delay. What is the one-way delay (in ms) that should not be exceeded as a design

goal? Answer

14. What one-way delay (in MS) should never be exceeded? Answer
15. What would you say is the biggest contributor to one-way delay? Answer
16. What is "look-ahead" used for? Answer
17. There is buffer delay. What is its cause and it varies with the number of xxxxx in tandem? Answer
18. What is one very obvious way to reduce delay? Answer
19. On G.711 circuits, the lost packet rate objective is? Answer
20. Give at least two ways to handle the presence of a lost packet. Answer
21. What is the basic function of a media gateway controller? Answer
22. Give at least three of the four signaling protocol options between an MGC and a media gateway. Answer
23. Give a primary function of an MGC regarding delay using the H.323 protocol on a VoIP circuit. Answer
24. Differentiate RTP and RTCP. Answer
25. In the SIP protocol, what are end-users called?Answer
26. What are the two modes with which a caller can set up a call with SIP? Answer
27. In MGCP gateways are expected to execute commands sent by____. Answer
28. Megaco (H.248) is a call-control protocol that communicates between a ____ and a____. Answer
29. What are the two basic abstractions relating to the Megaco model? Answer
30. A point-to-point connectivity must have at least ___ terminations. Answer
31. How are terminations referenced in Megaco? Answer

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1. 8

2. One party or the other goes "on-hook."

3. header and payload

4. destination address/es and originator's address

5. It does not carry any address information at all.

6. error detection

7. With DRMF tones or with an IAM in CCITT Signaling System No. 7

8. Mouth-to-ear delay, impact of errored frames, lost packets or frames, variation of packet arrival time, prioritizing VoIP traffic over other internet traffic, talker echo, distortion -- voice quality, voice coding algorithm for standardization, sufficient bit rate capacity, optimized standard packet payload size, packet overhead (efficiency), silence suppression)

9. Network management, authentication, voice prompts, PSTN interface, QoS records, H.323 functions

10. copper pair, ADSL, CATV cable, fiber (SONET/SD), radio, PDH PCM formats

11. POTS voice, data octets, DTMF tones

12. 80 or 160

13. 100 ms

14. 400 ms

15. Propagation time

16. To detect patterns in voice transmission to eventually save on bit rate capacity use

17. A buffer may be found in packet routers where at least one frame or packet must be stored. Thus buffer delay would be a function of the number of routers in tandem.

18. Speed up the bit rate.

19. 10%

20. a) disregard it
b) use a PLC procedure to camouflage the gap

21. Carry out a signaling function

22. SIP, H.323, MGCP and Megaco H.248

23. How to prioritize VoIP packets versus other data packets

24. RTP carries out media content function whereas RTCP works with signaling functions of status and control.

25. User agents

26. Redirect and proxy

27. Call agents

28. A gateway controller and a gateway

29. Terminations and contexts

30. 2

31. By terminationID.

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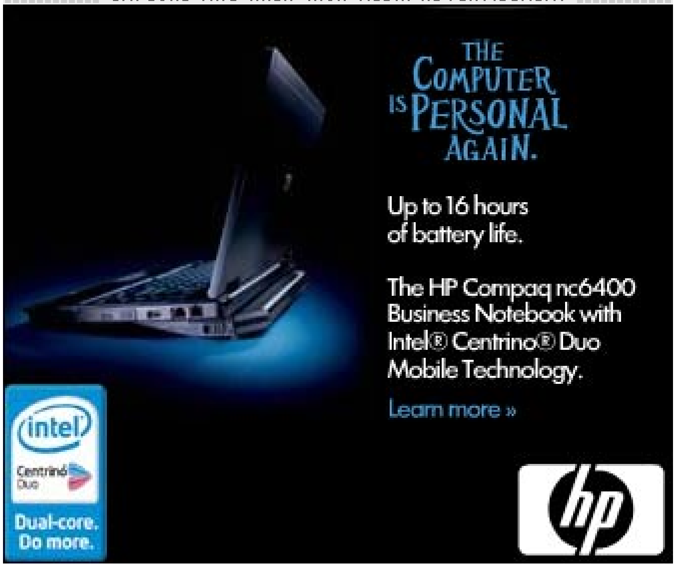

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