Table of Contents

Dr. Mona Cherri
Business and Technology
North Lake College/DCCCD

An Introduction to Voice over the IP

1. Background Information

Since 1876 and beginning with Alexander Graham Bell, the public switched telephone network (PSTN) has been evolving. Today, the PSTN or the Plain Old Telephone Service (POTS) is almost entirely digitized in technology except for the final link from the central/local telephone office to the user.

In the mean time, the computer industry developed and grew at an incredible speed. Many input, output and storage devices were conceived. Communication systems expanded in type and in geographical coverage. Various different devices can be connected together and reliable paths can be established between different locations. Computer systems and **endpoints** are sources or sinks of data that physically or logically exist within a network and/or an entity.

Thus, endpoints and/or computer systems are programmed to put data into frames, i.e. packets with checksum to ensure that the data is transferred correctly. Either rejecting the data or retransmitting it, rectified errors detected within the packets. Naturally, voice introduces stringent timing restrictions that communication systems have to account for.

Traditionally, voice fax and other types of information were carried over the PSTN's circuit-switched connections. Circuit-switched networks *dedicate* a physical path to a single connection between two end-points for the duration of the connection.

Today, Internet Protocol telephony (IT), which is an emerging technology, uses the Internet Protocol (IP, packet-switched) to exchange multimedia information. Communication is broken down into packets and packets are routed through the network according to the destination address. This type of communication between sender and receiver is classified as *connectionless* rather than dedicated.

Voice over the IP (**VoIP**) is the set of facilities used in IP telephony for voice information delivery/exchange. Generally, voice data is sent in digital form using discrete packets rather than the circuit-committed protocols of the PSTN. Standards for sending voice (audio) and video using IP on the public Internet and within an intranet are developed. An effort by major equipment providers including Cisco is made to promote the use of International Telecommunication Union (ITU) **H.323** standard for transmitting audio/video over IP networks.

In addition to IP, a real-time protocol (**RTP**) is applied to ensure that packets are delivered in a timely fashion. In order to ensure faster packets delivery some VoIP solutions contact all possible gateways and the fastest path is chosen prior to establishing a Transmission Control Protocol (TCP) sockets connections with the other end.

2. *Voice over IP versus PSTN*

This section describes the basic components of circuit based-switched (i.e. PSTN) as well as packet-oriented networks. In addition, a parallel and a correspondence between their infrastructures are given.

2.1 PSTN Basics

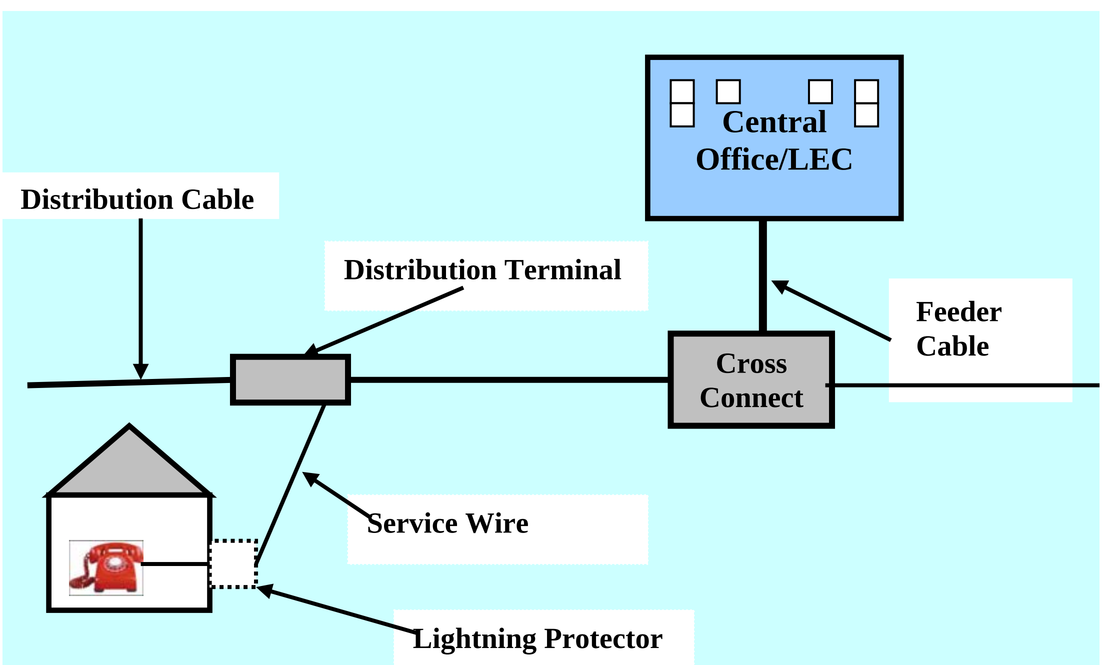
The structure of the PSTN is rather complex and involved. This section attempts to cover the most important elements and/or components of the PSTN.

2.1.1 PSTN Network Architecture

The first two telephones connected via physical wires and voice transmission was accomplished via a *ring-down* circuit (i.e. one person picks up the phone on one end and the other person is at the other end). Hence if an individual needs to call 10 different people, the location from which the phone call is initiated needs 45 pairs of lines running (i.e. n*(n-1)/2) to other locations.

Consequently, the need of a mechanism, the *switch*, to map any pair of phones rises to the surface (i.e. it is impossible to run a physical cable between any existing pair of telephones). In 1878**,** the first telephone exchange opens in New Haven, Connecticut under a license from Bell Telephone and within a few years, licensed telephone exchanges open in every major city in the country.
Now, telephone users need only a simple pair of copper wires running through their location. All telephone jacks are wired in parallel and a continuous wired connection is made from the jacks to the centralized switch office (CO). The CO is a secured building managed by a local exchange carrier (LEC), that terminates physical wires from thousands or ten thousands of households. This physical cabling is called the *local loop* (see Figure1*)*.

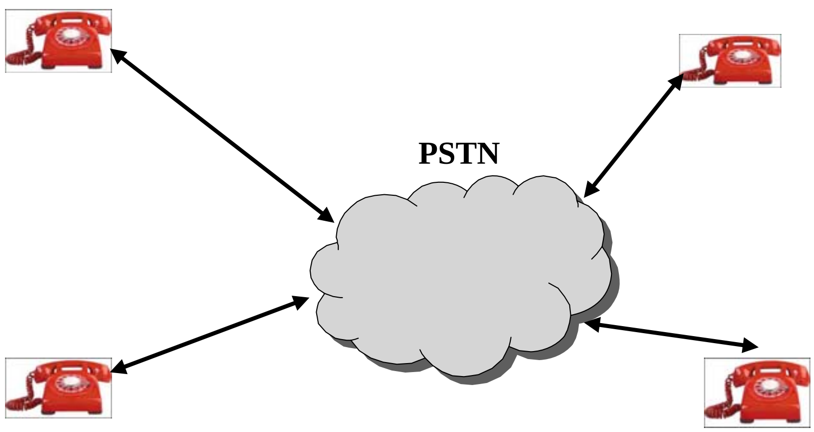
Figure 1: *Basic elements of a residential local loop*



In addition to the wire terminations, the CO (also known as *Class 5 switch* or *end office*) has communication paths to other COs and *tandem switches* (also known as *Class 4 switches*). Tandem switches are at a higher level and are connected to local COs via trunks.

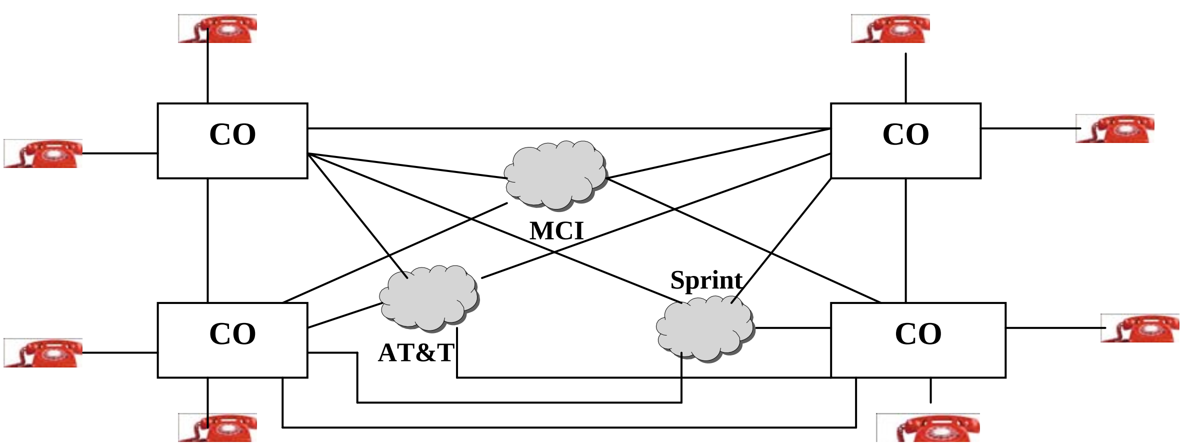
Frequently, in the PSTN network switches are deployed in hierarchies. Often, central office switches connect directly to each other and to local tandem switches, while higher layer tandem switches (also known as large voice switches) connect to local tandem switches.
The PSTN is depicted as the network cloud in the Figure below. From the point of view of a subscriber, its boundary is the jack in the wall that individual telephone sets plug into.

Figure 2: *PSTN cloud*



Looking inside the PSTN cloud we can see that it is made up of central offices (COs) that provides access to subscribers, and Inter-Exchange carriers (IXC) e.g., MCI, Sprint, and AT&T etc. that provide long distance services. Figure 3 presents a rough diagram of the PSTN.

Figure 3: *Inside the PSTN Cloud*



Business and other organizations have more complex telecommunication needs. On-premise business telephony system may be composed of but not limited to Private Branch Exchanges (PBX), business telephones, analog devices (fax machines), Key System Units (KSU), Automatic Call Distribution (ACD), Interactive Voice Response (IVR) units, voice mail and Auto-Attendant systems. At least one site and one analog phone at each site must have direct connections to the PSTN.

2.1.2 PSTN Signaling

Generally, users connect to the PSTN through analog, Integrated Services Digital Network (ISDN), or through T1/E1 carriers. There are two ways of signaling methods via the PSTN network: *in-band* and *out-of-band*. In-band signaling uses voice channels to carry the signal and out-band signaling is transported on a channel separate from the voice.

The most common signaling method for user-to-network analog communication is *Dual Tone Multi-Frequency* (DTMF). DTMF is an *in-band* signaling that passes a tone following every digit pressed from your phone to the CO. The mapping between digits and audible frequencies is based on the physical layout of the standard telephone keypad. A separate frequency is assigned to each row and column of the keypad, such that any key corresponds exactly with two frequencies, as shown in table 1.

Table 1: DTMF Assigned Frequencies

	1209 Hz	1336 Hz	1477 Hz	1633 Hz
697 Hz	**1**	**2**	**3**	**A**
770 Hz	**4**	**5**	**6**	**B**
852 Hz	**7**	**8**	**9**	**C**
941 HZ	*****	**0**	**#**	**D**

ISDN is an *out-of-band* user-to-network digital signaling method. The channel on which the voice is carried is called the *bearer* channel (B channel) and the signaling channel is called the data channel (D channel). The ISDN interfaces are required for subscribers to take advantage of the capabilities of the Advanced Intelligent Network (AIN) of North America.
The most common ISDN interfaces are the basic rate interface (BRI) and the primary rate interface (PRI). The BRI bundles two bearer channels with one data channel (2B+D). The PRI service is a 23B+D for a T1 facility and 30B+D for an E1 facility. ISDN uses a message-oriented protocol, specified in ITU-T recommendations Q.921 and Q.931 across the data channel. In addition ISDN supports many data types in the bearer or B channels, such as 3.1 kHz audio, speech, video, and other unrestricted data.

In-band network–to-network signaling includes Mutli-Frequency (MF) and Robbed Bit Signaling (RBS). MF is similar to DTMF; tones are sent in-band and use a set of different frequencies. But signals are sent from switch to switch. RBS uses some of the bits from the audio payload for signaling information.

For example, for every six frames the bits that represent the signaling state for each timeslot overwrite the encoded audio information in the least significant bit of their respective timeslots.

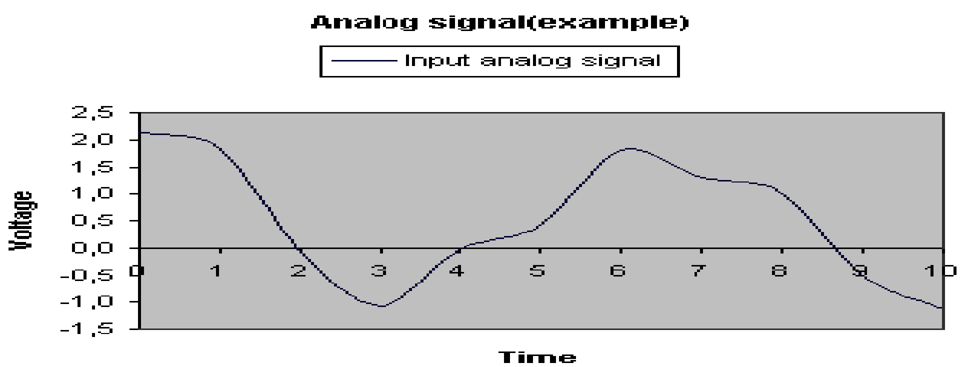
Out-of-band network-to-network signaling method is known as *Signaling System 7* (SS7) (or C7 in European countries). SS7 is a suite of protocols used by telephony service providers to support out-of-band call signaling and advanced calling features. SS7 supports the PSTN by handling call set up, teardown messages, database queries, trunk status, routing, billing, instructions for remote phone switches, and so on. An SS7 network is sometimes referred to as an Intelligent Network (IN), or an AIN, because the signaling message structure allows many more functions than just simple call setup and teardown.

The inter-working of the SS7 protocol within the PSTN is critical to the acceptance, development and ultimately the success of VoIP solutions. SS7 provides a common protocol for signaling, messaging, and interfacing for which VoIP-type devices can be developed.

2.1.3 Voice Digitization and Coding

Everything we hear, including human speech, is analog in form. Analog signals are continuously variable within a given range. Figure 4 shows an example of an analog signal that varies over time.

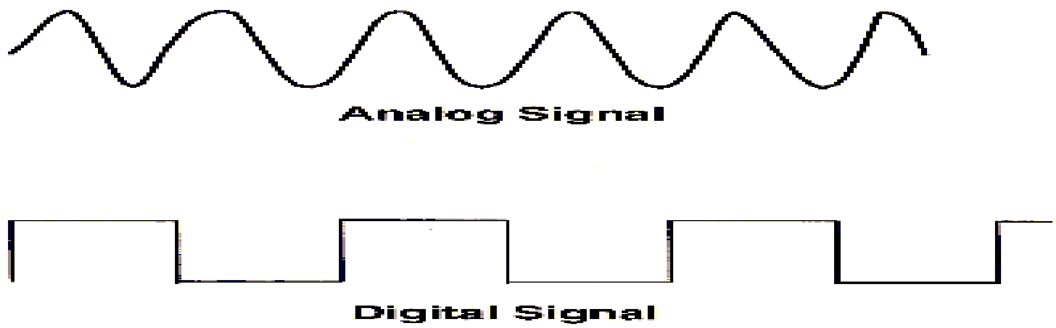
Figure 4: *Analog Signal Example*



Although analog communication is ideal for human interaction, line noise causes problems in audio transmission. Early telephony networks introduced amplifiers to boost the signal especially if the telephone users were at a considerable distance from the *end office* switch. As a result of this practice, analog amplifiers boost the entire input signal, including any noise introduced in the path (i.e. since an amplifier does not clean the signal as it enhances it). Consequently, this line noise accumulated and resulted is an unusable connection.

Digital signals have specified and/or discrete values within a given range. Figure 5 shows an analog signal versus the same digital signal varying over time. Note that the curve of the digital signal has characteristic straight lines and discontinuities.

Figure 5: *Analog versus Digital Signal*

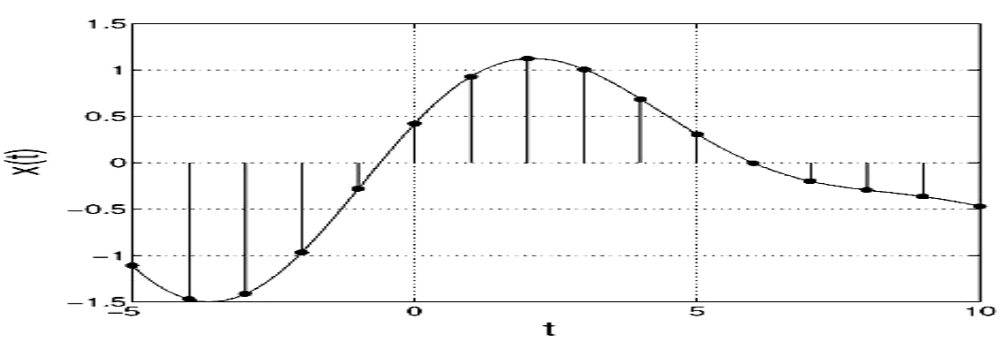


In digital networks line noise is not a problem. The digital signal that carries the same audio information is subject to degradation. But instead of boosting the signal waveform, a digital repeater is added. The repeater interprets the signal, and regenerates a clean version of that signal. With this method there is no accumulation of error and the signal propagates appropriately.
In comparison, note that analog signals are continuous while digital signals are discrete. In addition, the value of an analog signal reflects the value of the information source at any instant, the value of a digital signal reflects the signaling state. Instead of relying on the instantaneous value of the signal to encode the information source, digital signals consider a series of signal values to represent a single instant of the information source. A binary digital signal has two states, 0 and 1, and is far more efficient and appropriate for communication.

The most common method of encoding an analog signal into a digital stream of 0s and 1s is called pulse code modulation (PCM). Several steps must be taken in order to complete the PCM process as follows:

a. Analog signals are preconditioned and passed through frequency filter in order to retain audible frequencies. Although humans can hear frequencies as high as 20 kHz, only frequencies lower than 4 kHz are kept.
b. Filtered analog signals are sampled twice as high as their highest input frequency in accordance with the *Nyquist theorem* to achieve good-quality voice transmission. Therefore, the signal is sampled at 8 kHz so that frequencies as high as 4 kHz may be recorded. Every 125 microseconds (1/8000th of a second), the value of the analog signal is passed through the quantizer (see Figure 6).

Figure 6: *Sampled Signal*



c. Sampled signals are converted into a discrete digital form. The quantization function rounds the sampled values to the nearest predefined discrete value. This enables every value of the pulse to be represented by a binary bit-stream.
d. Digitized signals are encoded into a more efficient way for transmission or storage using *codecs*. The value of every sample is encoded using one of the two coding laws: mu-law and A-law. Both laws emphasize signal quality in the quiet audio ranges, at the expense of signal quality in the loud audio ranges.

2.1.4 End Office Switch Call-Flow
This section will give an example of a call-flow using two subscribers Bill, and Jody on the same local end office switch. Therefore, no SS7 is needed and the call is completed using the following steps:

a. Bill picks up his handset (off hook).
b. The local end office switch gives Bill a dial tone.
c. Bill dials Jody's phone number.
d. The end office switch collects and analyzes the dialed digits to determine the destination of the phone call. The end office recognizes that Bill is calling, since a specific port is dedicated to Bill
e. The switch analyzes the digits dialed to determine which switch can serve the called number. If the number is local the same switch can serve it. Otherwise the end office routes the call to the appropriate switch.
f. The switch determines Jody's specific subscriber line.
g. The end office switch then signals Jody's circuit by ringing Jody's phone.
h. A voice path is established so Bill can hear Jody's phone ringing (ring-back tone that the switch is sending).
i. Jody picks up the phone (off hook).
j. The end office seizes a voice path (trunk) from Bill to Jody. This is a 64 Kbps, full-duplex DS-0 in the end office switching fabric to enable voice transmission.

2.1.5 PSTN Services, Applications and Drawbacks
PSTN services are divided in two common categories: custom calling and Custom Local Area Signaling Services (CLASS). Custom calling rely upon the end office switch, not the entire PSTN, to carry information from circuit-switch to a circuit-switch. CLASS features require SS7 connectivity to carry these features from end to end in the PSTN.

Examples of custom calling features are: call waiting, call forwarding and three-way calling. Examples of CLASS/advanced features that are possible with the deployment of the SS7 network are: Automatic Number Identification (ANI), Call blocking, Call return, Calling cards, 800/877/888 numbers, Virtual Private Networks (VPNs), Private leased lines, Automatic call back and Calling line ID blocking.

Although the PSTN is efficient and effective to what is was built for (switch voice calls), it cannot create and deploy features fast enough. Data has overtaken voice as the primary traffic on many networks built for voice. The PSTN as currently built cannot accommodate the convergence of data, voice and video. Modifications to the current architecture of the PSTN are needed.

It is also important to note that a permanent 64-kbps dedicated circuit is required between two telephones for circuit-switched calls. Whether the caller or the called person is talking, the 64-kbps connection cannot be used for any other purpose. This means that the telephone company cannot use this bandwidth and must Bill the parties for consuming these resources. On the other hand, data networking has the capability of using bandwidth whenever it is needed.

2.2 VoIP Basics

At the beginning the Internet was not designed to support real time traffic data such as voice and video. Now, the Internet is modified to support voice traffic and products are being made to link the data and voice networks. This section introduces basic network configurations for VoIP solutions as well as standards, services and benefits.

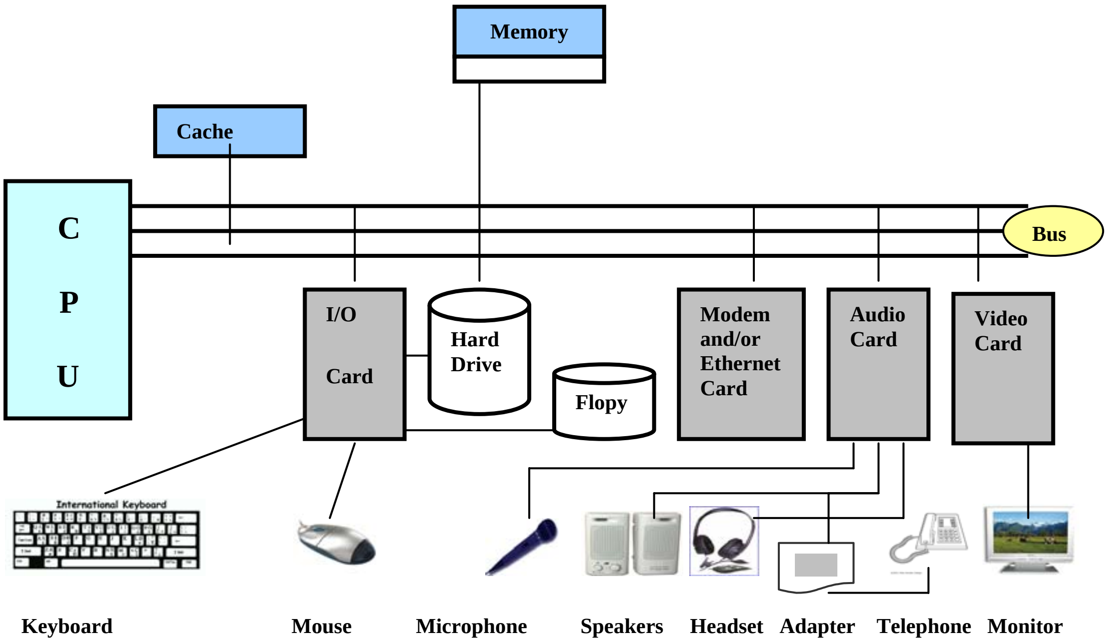
2.2.1 *VoIP Network Configurations*

VoIP calls have several flavors: personal computer (PC) to PC, phone to phone and PC to phone. This section will investigate network structures/configurations for the various types of VoIP calls.

2.2.1.1 PC-to-PC VoIP Network Configuration

In February 1995, Vocaltec Inc. introduced the first Internet Phone Software. This IP software was designed to run on a 486/33-MHz personal computer (PC) equipped with a sound card, speakers, microphone, and modem. The PC is an IP endpoint and usually speaks H.323 for media stream. The software compresses the voice signal and translates it into IP packets for transmission over the Internet. Figure 7 shows the PC configuration for VoIP.

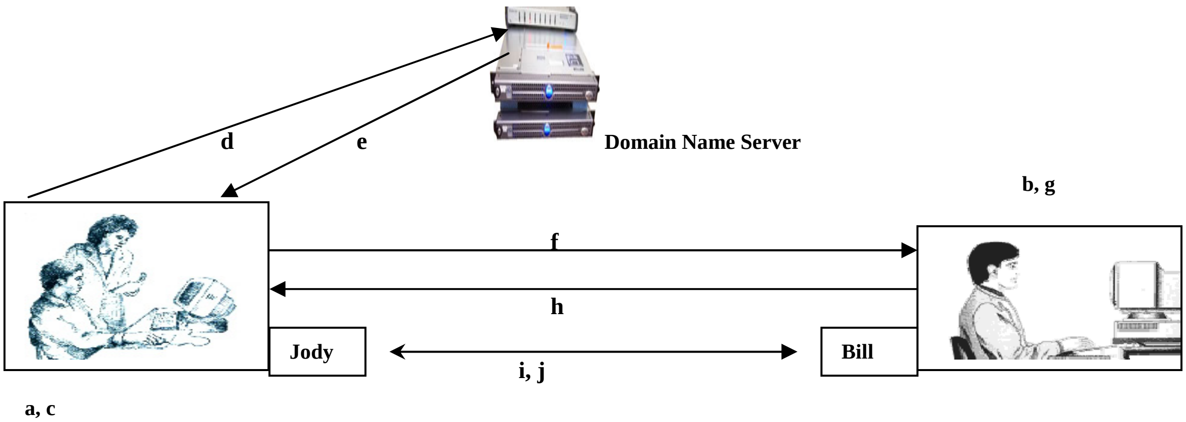
Figure 7: *PC configuration for VoIP*



This PC-to-PC Internet telephone works only if parties are using Internet-Phone software.

Figure 8 demonstrates the call-flow necessary to complete an Internet phone call using a PC I-phone application.

Figure 8: *Calling with a PC Internet-Phone application*



Both customers (Jody and Bill) must be connected to an IP network (Internet) and must have launched their I-phone application (H.323). In H.323 networks, call control procedures are based on International Telecommunication Union (ITU) recommendation H.225.

The purpose of initiating call control messages (H.225) is to connect, maintain and disconnect calls. On the other hand, H.245 procedures establish logical channels for transmission of audio, video, data, and control channel information. The following steps illustrate a PC-to-PC call flow.

a. Jody starts her I-phone application (H.323 compatible)
b. Bill has his I-phone application running
c. Jody knows that Bill's Domain Name System (DNS) entry, is Bill_friend1.neighboor.com, so the name is entered in the "who to call" section in the calling I-phone application and the Return key is pressed.
d. The I-phone application resolves via a DNS server Bill's name and converts the name into an actual IP address.
e. The DNS server passes back to Jody the IP address of Bill.
f. Jody's I-phone application takes Bill's IP address and issues a call setup (H.225) message to Bill.
g. The H.225 message signals to Bill to begin ringing.
h. Bill clicks the Accept button, which tells his I-phone application to send back a connect message (H.225).
i. Jody's I-phone application begins negotiation with Bill to establish logical channels (H.245).
j. H.245 negotiation finishes and logical channels are opened. Jody and Bill can now speak to one another via a packet-based network.

2.2.1.2 PC-to-Phone VoIP Network Configurations

In the relatively short time more software as well as hardware products were developed to act as interfaces between the Internet and the PSTN. Different vendors produced gateway servers, gatekeepers and IP phones.

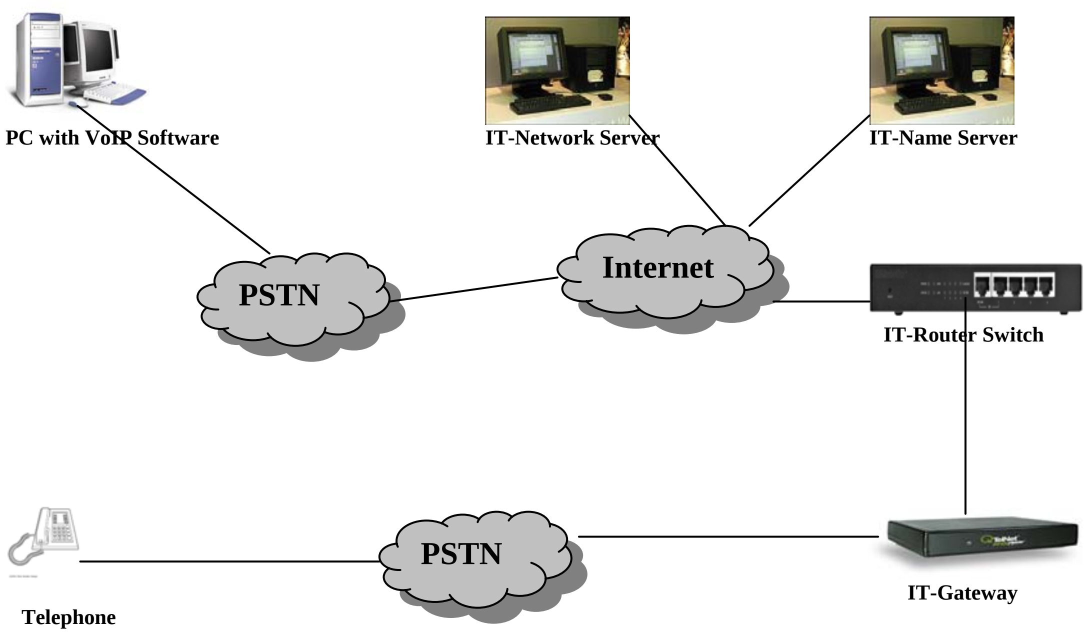
Gateways are devices that communicate between the telephone signals and the IP endpoint. These gateway servers, if equipped with voice processing cards, will enable users to communicate via standards telephones. Analog telephone signals coming into a trunk to a gateway are digitized by the gateway into a format useful for transmission, usually 64 Kbps PCM. Then, the signal is classified as voice, and is compressed by a digital signal-processing device (DSP) from 64 Kbps to a 5.3 Kbps signal (G.723.1 standard).

Sophisticated gateways handle both voice and fax signals. When the signal is a fax, it is demodulated by the DSP back into its original 2.4-14.4 Kbps digital format. Then, the digitized signal is divided into IP packets fro transmission. The demodulated information is remodulated back into its original analog format by the remote gateway, for delivery to the remote fax machine.

IT/IP gateways are also used to place a call across the IP network. Therefore, when a gateway receives the called party number, it converts into an IP address of the far end gateway. Usually, this translation happens through a table lookup in the originating gateway or in a centralized directory server. Then, the originating gateway establishes a connection to the destination gateway, exchanges call setup, compatibility information and performs any option negotiation and security handshake.

Normally, a call goes through the local PSTN network to the nearest gateway server, which digitizes the analog voice signal, compresses it into IP packets, and moves it into the internet for transport to a gateway at the receiving end (see Figure 9)

Figure 9: *Example of a PC to Phone Connection using an IT gateway*



Gateways translate between audio, video and data transmission format as well as communication systems protocols. Gateways can serve as endpoints in a packet-switched network as well as circuit-switched network (i.e. PSTN). Their role includes call setup and teardown on both the IP network and the circuit-switched network.

Gateways are needed if interconnection between packet-switched network and the circuit-switched network is required. Consequently, endpoints over the packet network can communicate directly with each other without the need of a gateway.

Gatekeepers register endpoints and keep track of how many users are connected and where are they located. The collection of a gatekeeper and its registered endpoints is called a **zone**.

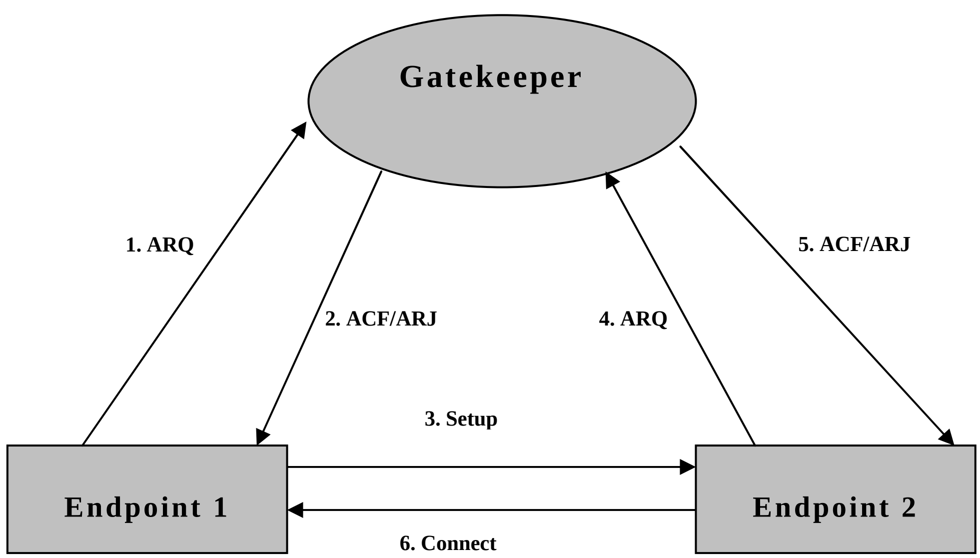
Gatekeepers are required to perform address translation, access authorization, bandwidth control, and zone management. Optionally, the gatekeeper can perform call control signaling, call authorization, and call management.

Gatekeepers perform a simple database query to remotely locate users and use *Admissions* messages for access control. The following messages provide admissions control between gatekeepers and endpoints:

- ARQ- An attempt message sent by an endpoint to initiate a call
- ACF- An authorization message sent by a gatekeeper to admit the call
- ARJ- A denial message of the endpoint's request to gain access to the network for this particular call sent by the gatekeeper

The ACF message contains the IP address of the terminating gateway or gatekeeper and enables the originating gateway to initiate call control signaling procedure immediately. Figure 10 illustrates call signaling messages sent directly between endpoints and gatekeeper.

Figure 10: *Direct Call Signaling between Gatekeeper and Endpoints*



Initially, the bandwidth is controlled through the admissions exchange between an endpoint and gatekeeper. However, if during the call the bandwidth changes, the following messages can be used to request and change the bandwidth:

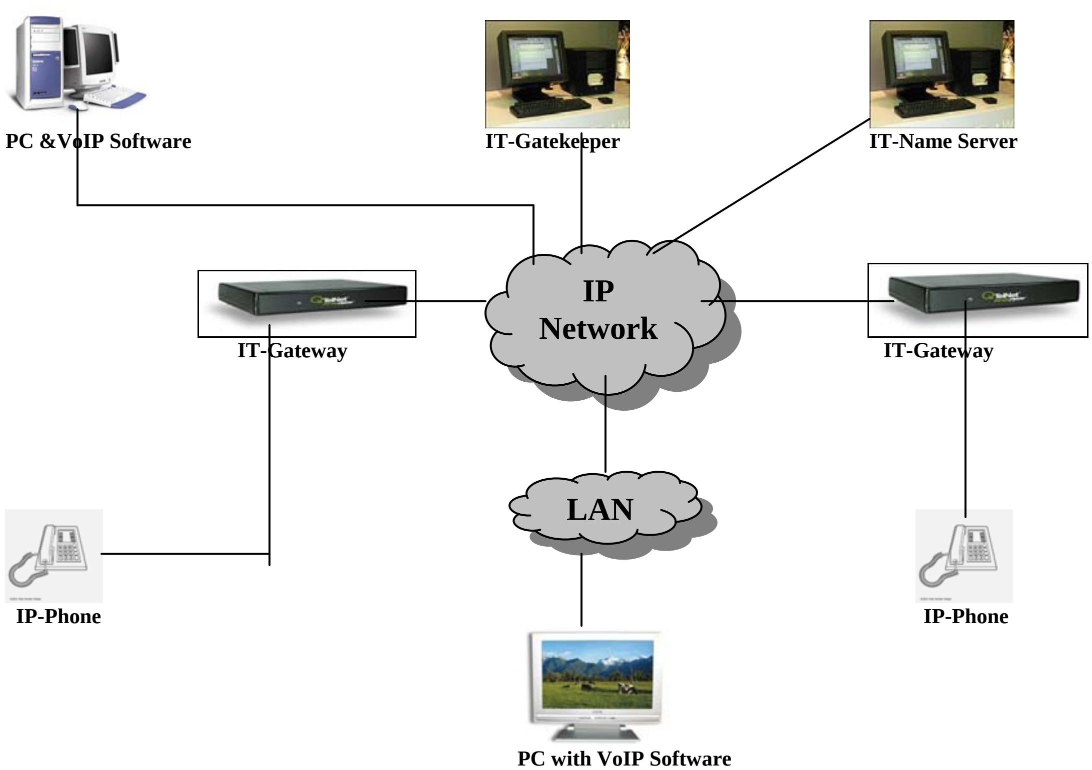
- BRQ- A request message of an increase or decrease in bandwidth sent by the endpoint
- BCF- A confirm message sent by the gatekeeper to accept the bandwidth change
- BRJ- A reject message sent by the gatekeeper denying the bandwidth change

Currently, the gatekeeper and/or gateway can control the bandwidth independently of the network state. A simple lookup in the static bandwidth table determines whether to accept or reject the bandwidth change.

Normally, Gatekeepers are logically separated from the other network elements. If two or more gatekeepers are needed, communications between gatekeepers are done in an unspecified manner.

IP telephones provide enhanced services suited for VoIP and will replace the existing telephones. In addition, IP telephones retain the capabilities of the original phones. Figure 11 shows a PC to IP-phone network configuration.

Figure 11: *Example of a PC to Phone Connection using one IT gatekeeper and two IP-phones*



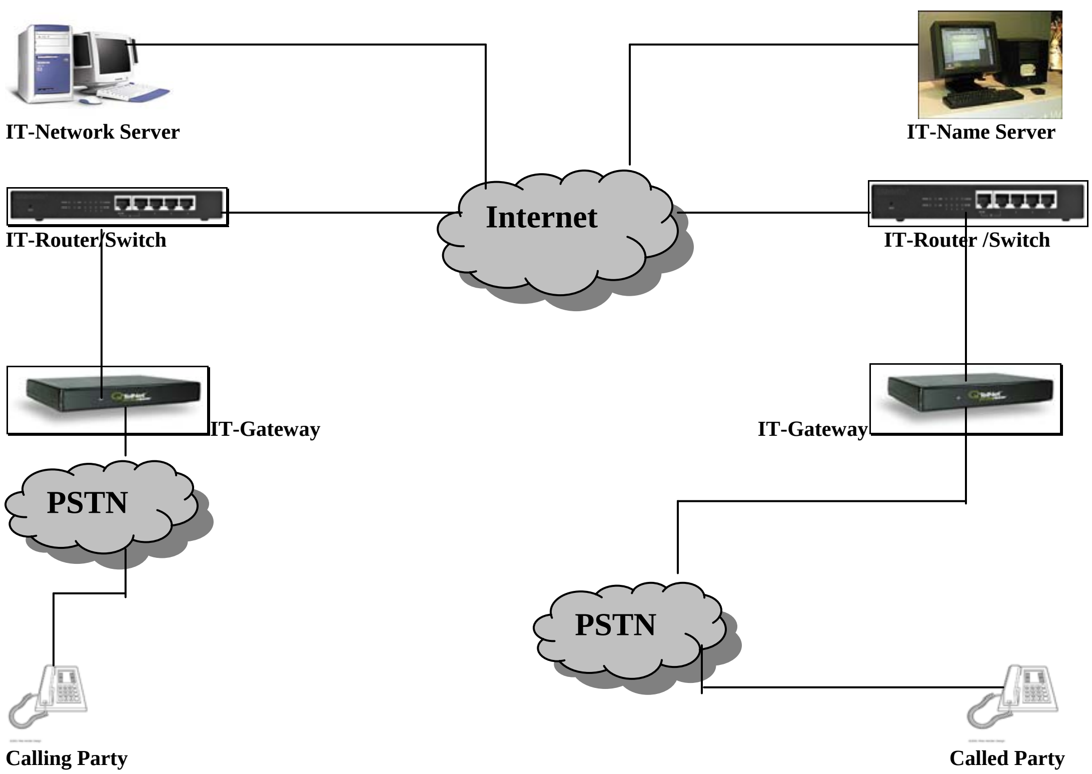
Note that the network configuration in Figure 11 accommodates PC to IP-Phone calls, PC-to-PC calls, and IP-phone-to-IP-phone calls.

VoIP network configuration depends upon the existing organization's network and its needs. The next section illustrates phone-to-phone configurations.

2.2.1.3 Phone-to-Phone Configuration

Internet telephony is becoming more attractive and popular. Users are able to bypass log-distance carriers and make their phone calls over the Internet for a flat monthly Internet-access fee. Thus, VoIP offers a tremendous cost savings over the PSTN. Figure 12 presents a phone-to-phone configuration.

Figure 12: *Example of a Phone-to-Phone Connection using gateways and regular phones*



User A (Calling Party) wants to make a phone call to user B (Called Party). User A picks the phone and dials the numbers. The CO connects with the originating gateway server (Origination ISP). The originating gateway, equipped with telephony board and compression-conversion software, digitizes the upcoming call. Then, the gateway transmits the (digitized, IP-packetized) call over the IP-based network (Internet) to the terminating gateway at the Called Party end (Destination ISP).

In the specific case where the telephone is directly connected to the gateway (see Figure 11); the user is required to dial an extension to connect to the gateway.

2.2.2 Voice over IP Standards and Protocols
The Internet industry is adopting standards and modifying protocols in order to increase network reliability and sound quality. Standards setting are focusing on the three central elements of Internet Telephony: audio *codec* format, transport protocols, and directory services.

2.2.2.1 H.323 Standard
In 1996, the ITU approved a standard (H.323) to promote compatibility in video conferencing transmissions over IP networks. At that time, local area networks (LAN) provided limited service quality (QoS). Consequently, H.323 was created to provide consistency in audio, video, and data packet transmission.
H.323 describes how multimedia communications occur between terminals, network equipments and services. H.323 is a part of a larger group of ITU recommendations for multi-media interoperability called H.3x. The latest of these recommendations is H.248 that was announced in August 2000, and provides a single standard for the control of gateway devices in multi-media packet transmissions. In addition, H.248 permits calls to connect from LANs to the PSTN as well as to other standard-based terminals.

The H.323 standard includes call signaling and control, multimedia transport and control, and bandwidth control for point-to-point and multipoint conferences. H.323 elements include terminals, gateways, gatekeepers, and multipoint control units (**MCU**).

Terminals often referred to as endpoints, provide point-to-point and multipoint conferencing for audio and, optionally, video and data. H.323 terminals are equipped with a system control unit, a media transmission device, an audio codec, and a packet-based network interface. MCU devices support conferences between two or more terminals or gateways.

The H.3232 protocol suite consists of three main areas. The first section covers Registration, Admission and Status (**RAS**) Signaling. RAS signaling provides pre-call control in gatekeeper-based networks. The second area consists of call control signaling which is necessary to connect, maintain, and disconnect calls between endpoints. The last topic is Media Control and Transport, which provides the reliable H.245 channel that carries media control messages.

The Transport uses either Transfer Control Protocol (**TCP)** or User Datagram Protocol (**UDP**) mechanism. TCP moves data in a continuous manner, with unstructured bytes that are identified by sequence numbers. In addition, TCP allows each station to send multiple packets prior to receiving an acknowledgment. After the sender receives an acknowledgment for an outstanding packet, the sender proceeds to send another packet. Within VoIP, TCP is used to ensure the reliability of the setup of a call.

UDP is also connectionless and a much simpler protocol than TCP. Within VoIP, and since flow control and retransmission of voice packets are not necessary, UDP is used to carry the actual voice traffic (*bearer channels*). Hence, UDP continues to transmit audio packets irregardless of the percentage of packets loss (i.e. 5 % or 50%).

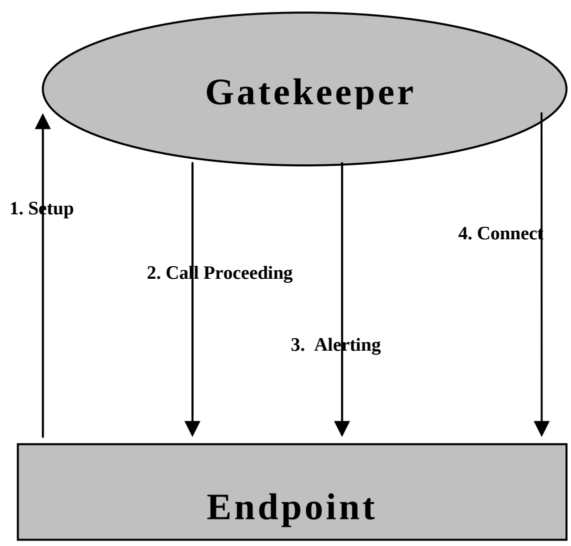
In H.323 gatekeeper-based networks endpoints are required to automatically discover and register with a gatekeeper.

Six messages enable an endpoint to register and cancel registration as follows: Registration Request (RRQ), Registration Confirm (RCF), Registration Reject (RRJ), Unregister Request (URQ), Unregister Confirm (UCF), and Unregister Reject (URJ).

The RAS channel, which is opened first, is used to obtain status information from an endpoint. This information is used to check whether an endpoint is online and/or offline due to failure condition. Three messages provide the status of an endpoint as follows: Information Request (IRQ), Information Request Response (IRR), and Status Enquiry.

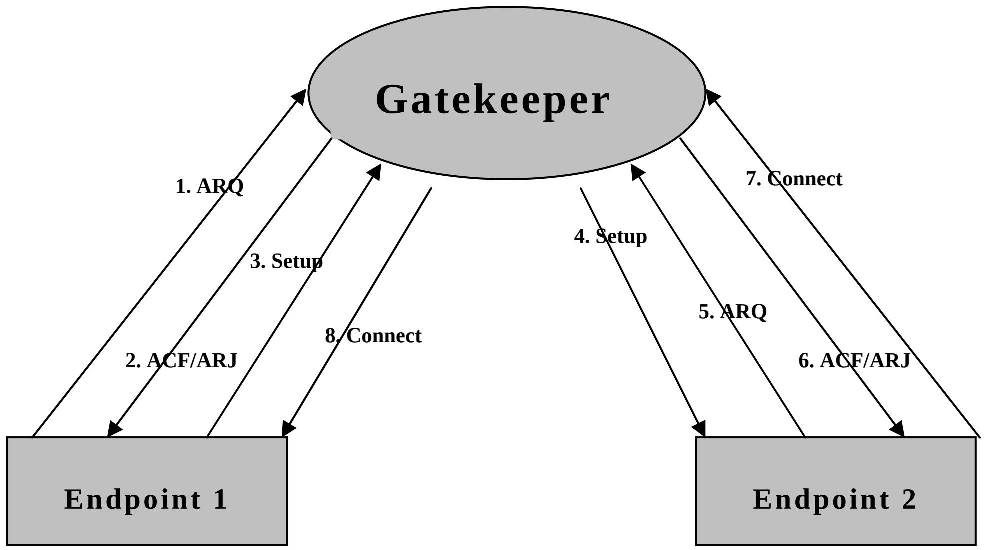
In H.323 networks, call control procedures are based upon the ITU recommendation H.225. A reliable call control channel is created across the IP network and call control messages are initiated between two endpoints for the purpose of connecting, maintaining, and disconnecting calls. The most commonly used signaling messages are: Setup, Call Proceeding, Alerting, Connect, Release Complete, and Facility. Figure 13 illustrates the call setup signaling messages.

Figure 13: *Call Setup Signaling between a Gatekeeper and an Endpoint*



In H.323 networks the call signaling channel can be routed in two ways: direct endpoint call signaling and gatekeeper routed call signaling. Figure 10 above shows a direct endpoint signaling and Figure 14 illustrates a typical routed call signaling between a gatekeeper and two endpoints.

Figure 14: *Routed Call Signaling between a Gatekeeper and two Endpoints*



In H.323 networks, H.245 and RTP handle end-to-end media control messages and media transport. H.245 establishes a reliable media control channel over IP using the dynamically assigned TCP port in the final call signaling message.

Several messages can be used in the media control operations as follows: Capability Exchange, Logical Channels Signaling, Round Trip Delay, and Master-Slave termination.

RTP provides media transport and enables, real-time, end-to-end delivery of interactive audio, video, and data. RTP ensures on-time delivery, reliability, and QoS. Payload, sequencing, time stamping, and monitoring are included in the transmission and packetization services. RTP-compliant equipments include control mechanisms for synchronizing different traffic streams. However, RTP does not possess any mechanisms to ensure on-time delivery of traffic signals nor to recover lost packets.

Currently, a draft of a signaling-protocol standard exists and is called Resource Reservation Protocol (RSVP). RSVP is aimed at strengthening the Internet's ability to handle real-time traffic data with high reliability. RSVP if adopted will be implemented in routers in order to establish and maintain requested transmissions path and quality of service (QoS) levels.

Finally, directories are required to ensure interoperability between the PSTN and the Internet. Current applications of the Internet telephony involve proprietary implementations of directory services. However, the lightweight directory access protocol (**LDAP**) is emerging to be the basis for a new standard.

Note that H.323 addresses the core Internet Telephony applications, defines how delay sensitive data gets the priority to transport, and ensures real-time communications over the Internet. But, H.324 specifies how voice, data, and video are transported over regular telephony networks and H.320 defines the protocol for transporting voice, data, and video over the ISDN.

In summary, H.323 addresses call control and management for both point-to-point and multipoint conferences, as well as gateway administration of media traffic. Therefore, H.323 is now considered to be the standard for interoperability in audio, video, and data transmissions over the IP.

2.2.2.2 Gateway Control Protocols

Gateway control protocols are used to control VoIP gateways from external call-control elements. Two Internet Engineering Task Force (IETF) gateway control protocols exist: Simple Gateway Control Protocol (**SGCP**), and Media Gateway Control Protocol (**MGCP**). These protocols support gateways with external intelligence such as large trunking gateways and residential gateways.

In **SGCP**, the call control intelligence is outside the gateway and is handled by *call agents*. Call Agents are identified in the VoIP network by their domain name not their IP address. These *call agents* handle call signaling functions and gateways provide audio translation functions.
The basic elements of SGCP are endpoints and connections. SGCP covers point-to-point or multipoint connections. By multipoint connection, we mean a connection between an endpoint to a multipoint session.

SGCP uses five primitives or commands for endpoint and connection handling functions.

- *Notification Request*- Issued by a call agent to a gateway in order to detect events such as off-hook, on-hook, and DTMF tones.
- *Notify*- issued by a gateway to advise the call agent of an event.
- *CreateConnection*- Issued by a call agent to create endpoint connections.
- *ModifyConnection*- Issued by call agents to request a change in an existing connection's parameters.
- *DeleteConnection*- Issued by both gateways and call agents to disconnect existing connections.

Connections operate in different modes: receive only (recvonly), send only (sendonly), send and receive (sendrecv), inactive, loop back (loopback), and test mode (contest). Return and error codes are a part of the SGCP acknowledgements message. These codes reflect the status of each acknowledgment request. For example a return code of 200 indicates a normal transaction execution. Individual vendors usually provide a table of the different return and error codes.

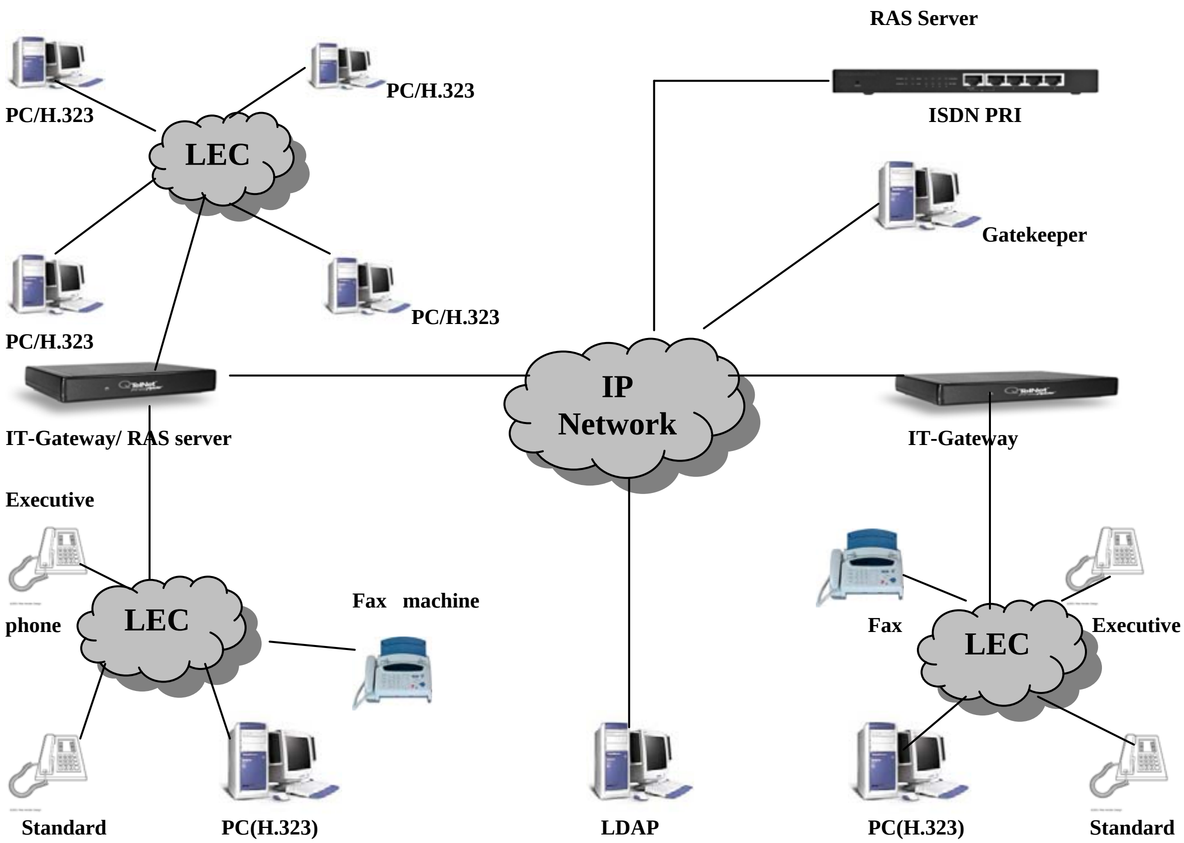
Similarly, MGCP controls VoIP through external call-control elements (i.e. call agents). However, MGCP enables connections to be created over IP networks and Asynchronous Transfer Mode (ATM) networks. In addition, internal connections are created in MGCP to permit the transmission of packets across the Time Division Multiplexed (TDM) back plane (i.e. occurs when a packet is not sent out into the packet network but sent back to the PSTN).

MGCP specifies New Return and Error codes, and modifies the SGCP five primitives by adding four additional functions as follows:

- *EndPointConfiguration* – Issued by a call agent to a gateway identifying coding characteristics of the line side of an endpoint (a-law, mu-law).
- *AuditEndPoint* and *AuditConnection* – Issued by a call agent to audit the status of a connection in the context of an endpoint.
- *RestratIn-Progress*- Issued by gateways to inform call agents when an endpoint is taken out of service or is back in service.

SGCP and MGCP are the basis for call agents and gateways to communicate and they are vital to the transition to a distributed platform. Figure 15 illustrates a VoIP network configuration where H.323 and SGCP are used.

Figure 15: *Example of Business VoIP Connections using gateways and regular phones*



3. Voice over IP Applications and Services

The single idea “*migrate to IP or risk of being left behind*” seems to dominate the minds of vendors who historically have been using circuit-switched infrastructures for the transportation of voice.

Products are being made to link the data and voice networks and the Internet is modified to support voice and video traffic. Eventually, data, voice and video networks will merge. This section will describe applications, services and benefits of the VoIP technology.

3.1 Consolidation of Voice and Data

For many years, businesses built networks based upon IP to take advantage of its networking power and the many services it provides. These services include Internet access for remote users, easy-to-use Web browsers, Intranets and Web servers, and Extranets with trading partners and suppliers.

The integration of voice and data traffic will enable businesses to bypass toll charges. Their intra-office voice and fax calls can be sent via their existing IP network.

Mutli-applications software requires the integration of voice and data. An example is the inevitable evolution of web servers capable of interacting with voice, data and images.

On the other hand, Microsoft Netmeeting provides integration between traditional telephone services and H.323 based video-conferencing. Thus, employees in different locations can easily collaborate on projects as well as reduce expenses by consolidating equipments and voice/data networks.

3.2 Simplification

Standardization and device management are directly dependent and affected by the network's infrastructure. Integrated infrastructures that support all forms of communication permit more standardization and fewer equipments management. Consequently, a fault tolerant network design is in place.

3.3 Network Efficiency

In TDM networks, users are given bandwidth whether they are talking or not. Data networks utilize bandwidth whenever necessary. Considering that 50% of the conversation is silence, the integration of voice and data will fill up the data communication channels efficiently. Thus, it will provide bandwidth consolidation (i.e. move away from the TDM scheme wherein the user is given bandwidth when he is not talking). In addition, removing the redundancy in certain speech patterns further boosts the network efficiency.

3.4 Cost Reduction

Using the Internet's backbone, users bypass The Public Switched Telephone Networks' toll services. This will result in reduction in prices of the long distance calls.

Although, Internet Service Providers (ISP) do not have to pay the local access fees to use the telephone companies' local access facilities (Enhanced Service Provider status). These access fees form a significant part of all long distance calls. The Federal communications Commission (FCC) will remove the Enhanced Service Provider (ESP) status granted to ISPs. But in spite of this, the circuit switched telephony would be expensive because of lack of bandwidth consolidation and speech compression techniques.

3.5 New Applications

As businesses and organizations become more comfortable with VoIP and toll-bypass, the next applications will be the ones that they can apply to customer service, interactive project groups, and distance-based learning and training. This section will describe some of these new services.

3.5.1 Directory Services over Telephones

Directory services will be reduced to simply submitting a name and receiving a reply. While, Ordinary telephones are enhanced to act as Internet access devices.

3.5.2 Inter Office Trunking over the Corporate Intranet

Intranet links can replace the tie trunks between companies owned PBXs. This will result in large savings at a good quality of service.

3.5.3 Remote Access to the Office from your home

Employee's home can be converted to a home office by gaining access to the company's voice, data and fax services using the company's Intranet.

3.5.4 IP-based Call Centers

In the last few years, E-commerce has gained popularity and companies have experienced a large increase in their web site inquiries. Although, not all inquiries will result in immediate financial transaction, people like to know about their products.

With VOIP there can be direct interaction between the customers and the business. Click-2-Dial enables businesses to put a link on their Web sites that automatically places a call from a customer to a customer service representative.

IP telephones look and feel like traditional handsets with the added functionality of IP connectivity. IP-based Private Branch Exchanges (IP-PBXs) interface with IP-phones to provide the traditional functionality of the PBXs (dial tone, voice-mail, and conferencing) as well as IP-based services available on the network. These new features allow the IP phone to use not only the VoIP services but also other IP-based multi-service applications available on the network.

Soft phones extend the handset functionality to the PC using a graphical user interface. PC-based soft phones eliminate the need to have an additional device on every desktop by using headsets and speakers. In addition, Soft phones integrate well with other multi-service applications such as Web browsing, or Net meeting.

3.5.5 Fax over IP

Traditionally, facsimile services use dial-up PSTN services. The latter services are affected by high cost for long distance, analog signal quality and machine compatibility. Eventually, a fax interface unit can convert the data to packet form, will handle the conversion of signaling and controlling protocols, and will ensure complete delivery of the data in correct order. Hence, real time facsimile transmission is an immediate application of Voice over IP.

4. References

1. Allan Sulkin, "PBX Systems for IP Telephony".
2. Bill Douskalis, "IP Telephony - The Integration of Robust VoIP Services**"**
3. Bill Douskalis, "Putting VoIP to Work: Soft switch Network Design and Testing".
4. Camp Ken, "IP Telephony Demystified".
5. Daniel Collins, Carrier Grade Voice Over IP
6. Daniel Minoli, Delivering Voice over IP Networks
7. David Lovell, "Cisco IP Telephony".
8. David Wright, Voice over Packet Networks
9. Frank Ohrtman, "Soft switch: Architecture for VoIP (Professional Telecom)".
10. Harte Lawrence, "Introduction to IP Telephony: Why and How Companies are Upgrading Private Telephone Systems to use VoIP Services".
11. John Alexander (Editor), Chris Pearce, Anne Smith, Delon Whetten, Cisco CallManager Fundamentals: A Cisco AVVID Solution
12. Jonathan Davidson , Tina Fox , Phil Bailey, Deploying Cisco Voice Over IP Solutions
13. Jonathan Davidson, James Peters, Brian Gracely, "Voice over IP Fundamentals**"**.
14. Mark A. Miller, "Voice over IP Technologies: Building the Converged Network".
15. Olivier Hersent, David, Gurle, and Jea-Pierre Petit, "IP Telephony: Packet-based multimedia communication systems".
16. Paul Mahler, VoIP Telephony with Asterisk
17. R.G. Cole and J.H. Rosenbluth, "Voice Over IP Performance Monitoring", Computer Communication Review, a publication of ACM SIGCOMM, volume 31, number 2, April 2001.
18. The European Online Magazine for the IT Professional http://www.upgrade-cepis.org Vol. II, No. 3, Jun. 2001.
19. Scott Keagy, "Integrating Voice and Data Networks".
20. William C. Hardy, "VoIP Service Quality: Measuring and Evaluating Packet-Switched Voice",

5. Useful URLs

1. http://www.acterna.com, "VoIP Fundamentals and Test Applications**"**.
2. http://www.eno.com/education, "Voice over IP Fundamentals**"**.
3. http://itresearch.forbes.com, "Voice over IP 101".
4. http://voip-voice-over-ip.com, "VoIP Voice over Internet Protocol Communication".
5. http://www.cwanett.org/courses.aspx , "Fundamentals of Voice and Data Cabling ".
6. http://www.mayflex.co.uk, "Design of Voice over IP Networks".
7. http://www.idc-online.com, "Practical Fundamentals of Voice over IP for Technicians and Engineers".
8. http://www.learningtree.co.uk, "Voice over IP".
9. http://www.cssa.net, "VoIP Technologies".
10. http://www.snrc.uow.edu.au, "Voice over IP".
11. IP Telephony
12. vovida.org contains source code for the *Vovida Open Communication Application Library (VOCAL)*, which includes the servers described in the course book.