Lesson 2: Voice over IP Technologies

At a Glance

The basics of telephone service have remained the same for over 100 years. The same analog telephones used years ago are essentially the same as the analog telephones found in most homes today. However, with the new millennium the future of telephony is changing rapidly. With the prevalence of IP networks all over the world, it is only natural to progress to using these data networks as a transport mechanism for voice transmissions. Although the costs associated with local calls may remain the same, using voice over IP for long distance calls, especially overseas, can save individuals and corporations considerable money.

This lesson will cover the fundamentals of voice, basic telephone operations, and the technologies involved in implementing voice over IP networks.

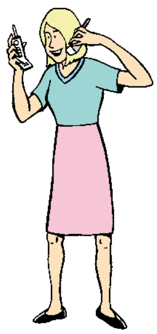
What You Will Learn

After completing this lesson, you will be able to do the following:

- Identify the three components of human speech
- Compare waveform, source, and hybrid coding methods used to digitize and compress analog signals
- Understand the basic function of the PBX
- Identify the advantages of implementing VoIP
- Identify the services provided by H.323, IGMP, RTP, RTCP, and RSVP
- Diagram and defend a Core WAN Network Design

Student Notes:

Tech Talk



- **Adaptive Differential Pulse Code Modulation (ADPCM)**—A variation of PCM. It is the most common speech compression technique in use today. Instead of quantifying the speech signal directly, ADPCM quantifies the difference between the speech signal and a prediction of what the speech signal will be.
- **Codec**—Coder-decoder. A hardware device or software that converts analog signals into digital signals and digital signals into analog signals.
- **Code Excited Linear Prediction (CELP)**—A hybrid coding method that analyzes incoming PCM bytes from the source against an indexed code book of pre-defined speech samples that are common to both the transmitter and the receiver. The code book is used to reconstruct the analog speech.
- **Conjugate-Structure Algebraic Code Excited Linear Prediction (CS-ACELP)**—A speech compression technique that is based on CELP. It was originally designed for packetized voice support. CS-ACELP operates at a transmission rate of only 8 Kbps.
- **Hybrid Coding**—A form of coding that combines aspects of waveform and source coding, to digitize analog speech at transmission rates below 16 kbps.
- **Internet Telephony Service Providers (ITSP)**—Internet service providers that also support VoIP telephone calls for a fee.
- **Plosive Sounds**—Sounds produced when the air is released with a sudden burst, such as when the mouth is closed and suddenly opened, such as the sound "p".
- **Pulse Code Modulation (PCM)**—A waveform coding technique that samples an analog signal and creates a digital copy with a 64 Kbps transmission rate.
- **Source Coding**—A form of coding that compresses speech to transmission rates as low as 2.4 kbps or less by analyzing the analog signal and creating a model of the source. A number of parameters are digitized by the source coder and then transmitted to the destination. Also known as Vocoders.
- **The International Telecommunications Union Standardization Sector (ITU-T)**—Formerly the International Telegraph and Telephone Consultative Committee (CCITT). An organization that establishes standards for the telephone industry.

- **Unvoiced Sounds**—Sounds produced when the air passes over an obstacle in the mouth, such as the sounds "s", "f" and "sh".
- **Voiced Sounds**—Sounds produced when the vocal chords vibrate, such as the vowel sounds.
- **Waveform Coding**—A form of coding that attempts to reconstruct an analog waveform, in a digital form as close to the original signal as possible without any knowledge of how the signal was produced.
- **VoIP Gateway**—A network component connected to the PBX that digitizes, compresses, and packetizes the analog calls and then addresses the calls for delivery to their destinations.

Voice Fundamentals

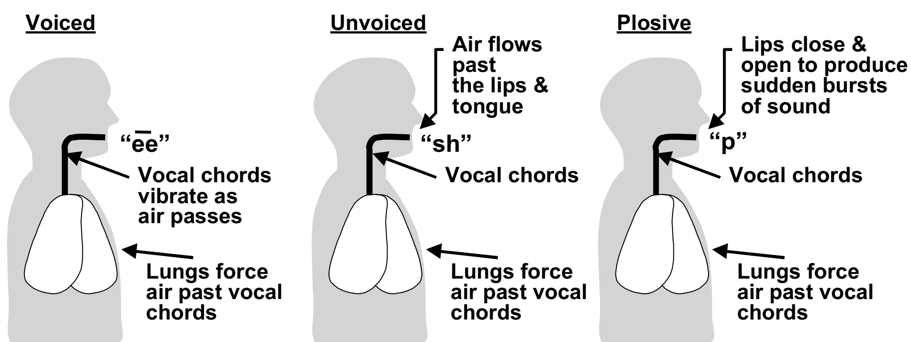
Human speech occurs as a result of air being forced from the lungs, through the vocal chords and along the vocal tract from the opening in the vocal chords to the mouth and nose. Speech is a compilation of three basic types of sounds: voiced, unvoiced, and plosive.

- **Voiced**—Sounds produced when the vocal chords vibrate, such as the vowel sounds. The vibration interrupts the flow of air from the lungs and produces sounds in the frequency range of 50 Hz to 500 Hz.
- **Unvoiced**—Sounds produced when the air passes over an obstacle in the mouth, such as the sounds "s", "f" and "sh". The frequency levels when unvoiced sounds are created is nearly flat.
- **Plosive**—Sounds produced when the air is released with a sudden burst, such as when the mouth is closed and suddenly opened, such as the sound "p". Plosive sounds occur in the low frequency range of all sounds.

The range of frequencies of these sounds mixed with individual characteristics of each person's nasal cavities and sinuses determine the unique sound of a person's voice. The normal frequency range of speech is from 50 Hz and up, with the majority of sounds occurring between 300 Hz and 3 kHz.

The human ear can detect sounds over a range of 20 Hz to 20 kHz. However, the ear's greatest sensitivity is in the range of 300 Hz to 10 kHz. Engineers of the telephone took these facts into account when designing the modern telephone. The goal was to maintain recognizable speech without wasting bandwidth by increasing quality beyond the levels the human ear could distinguish. With this goal in mind, the telephone system is limited to the frequency range of 300 Hz to 3.4 kHz. This range provides extremely understandable speech without wasting bandwidth.

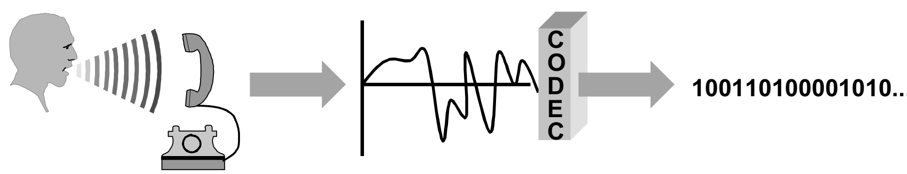
The Human Voice



Digitizing Voice

Digital voice is a representation of analog voice signals using binary 1s and 0s. When the talker speaks, they create variations in the air pressure. The telephone picks up the pressure changes and turns them into an analog signal. This analog signal is then converted by a codec into a digital stream of data bits that represent the digital voice signal. A codec stands for coder-decoder and is a hardware device or software that converts analog signals into digital signals and digital signals into analog signals.

Codecs Transform Analog Signals into Digital Signals



There are several methods defined by the International Telecommunications Union Standardization Sector (ITU-T) for digitizing and compressing analog voice signals based on two basic observations made about analog speech.

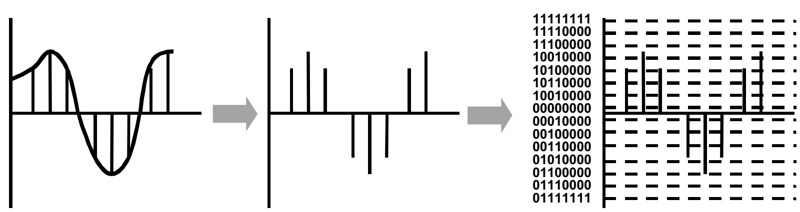
- Analog speech can be sampled at regular intervals over time. These samples could be used as a pattern to create a digitized replica of each sample period, resulting in a digital stream of data bits. This is accomplished using waveform coding, which attempts to reconstruct a waveform, in a form as close to the original signal as possible without any knowledge of how the signal was produced.
- Over a period of milliseconds, the sounds from analog speech do not change significantly. Since the sounds are predictable, it is possible to create a digitized model of analog speech. This is accomplished using source coders or vocoders, which analyze the signal and create a model of the source. The parameters of the model are sent to the destination, where they are used to rebuild the model and recreate the speech.

Waveform Coding

Pulse Code Modulation (PCM) is an example of a waveform coding technique. The ITU-T designation is G.711. PCM is a three-step process.

1. PCM samples the analog signal at a rate of 8000 times per second. The sampling is used to create a pulse amplitude modulation wave or PAM.
2. Each PAM sample is then quantified into a digital value. However, PCM does not use a one-to-one assignment of a value. Although doing so would produce an exact copy of the analog signal, a one-to-one assignment would also create an unnecessarily high bandwidth digital signal, one beyond the detection capacity of the human ear.
3. Using compression techniques, the PAM signal is coded into a specific digital value taking into account the need to reduce bandwidth and still maintain understandable speech. The process results in digitized speech with the transmission rate of 64Kbps.

Pulse Code Modulation

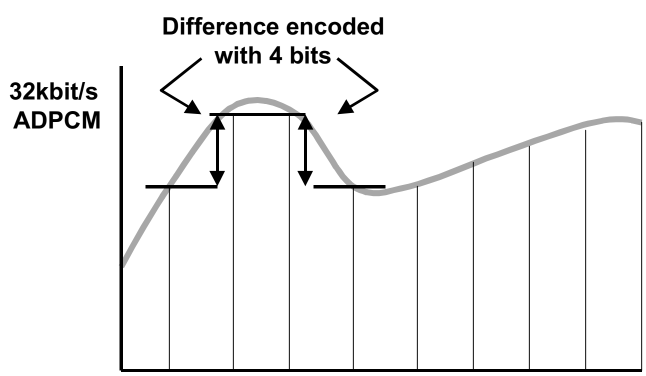


Another common waveform coding technique is Adaptive Differential Pulse Code Modulation (ADPCM), standardized in ITU-T's G.726. As its name implies, ADPCM is a variation of PCM and it is the most common speech compression technique in use today.

Instead of quantifying the speech signal directly, ADPCM quantifies the difference between the speech signal and a prediction of what the speech signal will be. It does this by using the fact that the adjacent samples are usually similar to each other. The value of the current sample can be predicted by using the value of the sample just before it. The difference between one sample and the next sample is then encoded.

ADPCM, also adapts the number of bits used to encode a sample depending on the range of amplitudes occurring in the analog signal over time. By doing this, fewer bits are needed to encode the signal that results in lower transmission rates than PCM, 16 Kbps to 40 Kbps.

Adaptive Differential Pulse Code Modulation



Source Coding

Source coders or Vocoders are more complicated than waveform coders, but they can compress speech to transmission rates as low as 2.4 Kbps or less. Vocoders analyze the analog signal and create a model of the source. A number of parameters are digitized by the vocoder and then transmitted to the destination. The vocoder at the destination uses this information to rebuild the model and recreate the speech. These parameters include such information as whether the sound is voiced or unvoiced.

Although the digitized speech is understandable, the general quality is poor and it is difficult to recognize who is speaking. The reproduced speech tends to sound synthesized instead of natural. Vocoders also do not provide quality support of non-speech signals such as modems and faxes. Source coding is typically used in military environments where natural sounding speech is not required.

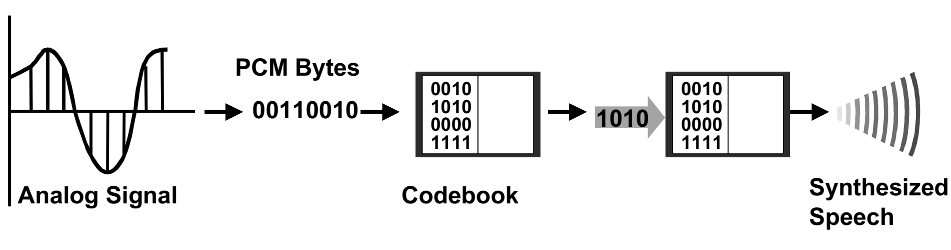
Hybrid Coding

Using waveform coders, the quality of digitized speech reduces tremendously at transmission rates around 16 Kbps. Using Vocoders, transmission rates can be lowered below 2.4 Kbps, but the digital voice sounds unnatural. Code Excited Linear Prediction (CELP) was developed as a hybrid coding technique, which combines aspects of waveform and source coding, to digitize analog speech at transmission rates below 16 Kbps.

A CELP encoder analyzes the incoming PCM bytes from the source against an indexed code book of pre-defined speech samples that are common to both the transmitter and the receiver. The encoder then transmits a number of parameters to the decoder at the destination. The decoder at the destination uses the code book to produce a synthetic speech signal for every input speech waveform.

CELP produces very high quality digital speech at transmission rates typically between 4.8 Kbps and 16 Kbps. However, since CELP is so tuned for human speech, it does not provide good support for modems and faxes.

Code Excited Linear Prediction



Conjugate-Structure Algebraic Code Excited Linear Prediction (CS-ACELP) is another speech compression technique that is based on CELP. It was specifically designed for packetized voice support. CS-ACELP operates at a transmission rate of only 8 Kbps yet it still provides the same speech quality of ADPCM at 32 Kbps. It also operates well when packets are lost. CS-ACELP is defined in ITU-T G.729.

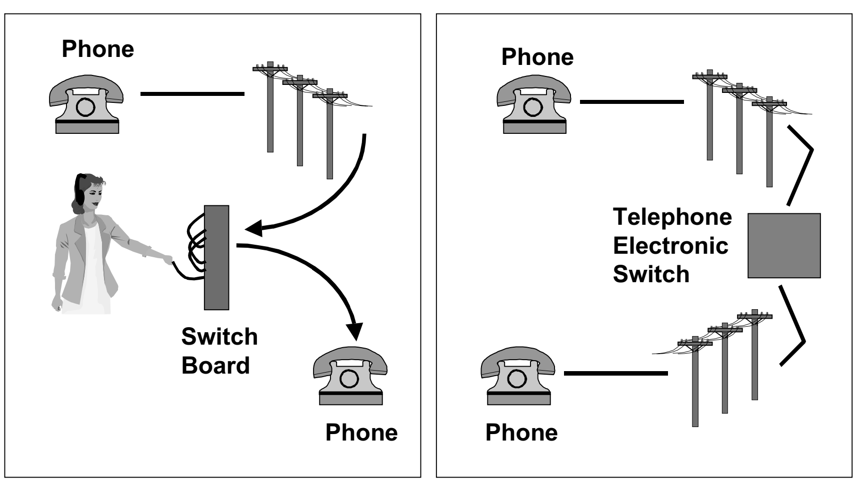
The Telephone

In the mid-1870's, Scottish born inventor Alexander Graham Bell had an idea for a device that would transmit sound over long distances by converting the sound to an electrical signal. This device was later called the telephone derived from the Greek words meaning “far” (tele) and “sound” (phone). The first telephone was patented in 1876.

Initially, the telephone had no mechanism for dialing another number. To make a call a handle needed to be turned, which generated an electric current. This current signaled the operator by turning on a light bulb above the caller's jack on the switchboard. To connect a caller to the called party, the operator connected his or her handset into the caller's plug and asked to whom the caller wished to speak. The operator then sent a ring signal to the receiving telephone. When the owner picked up the telephone receiver, the operator connected the two individuals together by placing the caller's plug into the receiver's socket on the switchboard.

An undertaker, Almon B. Strowger developed the automatic telephone exchange, in 1889. Strowger developed the exchange as a way of beating his business rival in Kansas City. The wife of Strowger's main competitor was the operator of the local exchange and whenever a call came in asking for an undertaker, naturally she passed it onto her husband. To eliminate this problem, Strowger developed the first automatic telephone exchange and the dial telephone. An electronic switch replaced the operator.

The Electric Switched Replaced the Telephone Switchboard Operator



Telephone networks have undergone many changes since those early days. However, many of the underlying principles remain the same. The basic "two wire" telephone set used in most domestic homes today still operates in essentially the same way as the telephones of over 100 years ago.

The Basic Telephone

There are two basic categories of telephones, analog and digital. Most telephones used in homes are still analog and are based on a two-wire system, often called POTS (plain old telephone service). In POTS, the telephone is connected to the telephone exchange via two wires that carry the voice signals in both directions simultaneously. The wires also carry the dialed digits to the exchange and the incoming ring to the telephone.

Some telephones operate as a four-wire system. In a four-wire system, speech is carried in each direction on separate pairs of wires or, in a digital system, on separate channels, as in ISDN.

The Private Branch Exchange

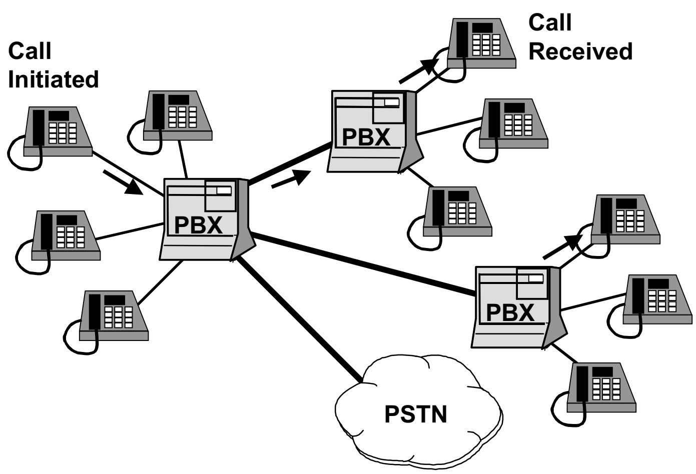
A private branch exchange (PBX) is a telephone exchange or switch that is privately owned by a large organization rather than the Telephone Company. A PBX is able to interconnect hundreds of telephones within an office building. Most PBXs today are digital, which means they first convert speech from analog to PCM before routing the call to its destination.

There are three voice interfaces in a PBX network.

- **Line Interfaces**—These are the lines used to connect the PBX to the telephone on an employee's desk.
- **Private Trunk Interfaces**—These are the links between PBXs within a multi-PBX private network. They allow calls to be routed from one PBX to another without involving the PSTN. An employee can place a call within the organization by merely dialing the extension number of the other employee.
- **Public Trunk Interfaces**—These are the links between the PBX and the PSTN for outgoing and incoming calls. To place an outside call requires the call to be routed via the PSTN. Usually the caller must dial an access code, such as "9" or "0", to connect to a telephone line outside the PBX system.

The PBX behaves much like a router does within a computer network. When a user dials a destination number, the PBX needs to determine how to route the connection in the most efficient manner. These choices are made based on information in its routing table, which is configured by an administrator. Within a network of PBXs, the PBX directly connected to the caller's line will examine a call. That PBX then decides which trunk to route the call on. The next PBX in line repeats this process until the call is routed to the destination telephone.

The PBX Network



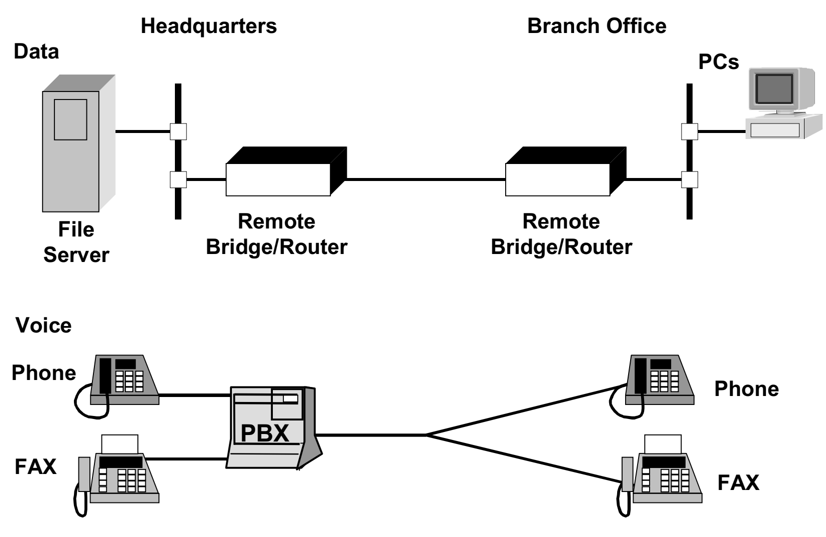
Check Your Understanding

- What advantages does ADPCM have over PCM?
- Speculate why vocoders tend to create voice reproductions that have a mechanical sound.

Voice over IP

Enterprise networks typically have separate, independent data and voice infrastructures. The data infrastructure supports financial, e-mail, and other business applications. The voice infrastructure supports telephone calls, voice mail, and fax mail. These separate networks have separate costs associated with management, leased lines, applications, and support personnel. Additionally, the demand for Internet and intranet applications and business strategies, such as e-commerce, has also increased dramatically. As a result of these demands, corporations are constantly seeking the best combination of technologies that reduce costs and accommodate the demands of the Internet.

Data and Voice Networks are Traditionally Separate



IP telephony is an important part of the convergence of computers, telephones, and video into a single integrated information network. IP or Internet Protocol telephony is a general term for the technologies that use IP's packet-switched connections to exchange voice, fax, and other forms of information that is traditionally carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN).

Using the data wide area network, calls travel as packets of data on shared lines, avoiding the toll charges of the PSTN. The challenge in IP telephony is to deliver the voice, fax, or video packets in a dependable flow to the user. Voice over IP, commonly known as VoIP, is an organized effort to standardize IP telephony.

VoIP has two main goals.

- The first is to reduce telephone charges by sending voice transmission over data lines. This is done by sending voice packets free of charge across corporate data networks, or the Internet, thus bypassing the toll charges of the PSTN.
- The second goal is to provide voice networks some of the same flexibility available to data networking. VoIP makes phone-to-phone, phone-to-PC, PC-to-phone, and PC-to-PC calls possible. It can also allow multiplexing several telephone calls on one physical link.

VoIP Protocols

VoIP takes advantage of the one network solution available nearly everywhere, the Internet Protocol. Although IP networks are highly accessible, they are in fact networks based on best effort quality of service and for that reason are not inherently ideal for voice and video data. There are many issues, such as reliability, delay, security, packet loss, packet size, and interoperability between vendor products that can negatively affect the performance of voice transmissions over IP. In an attempt to resolve some of these issues, several standards have been developed.

H.323

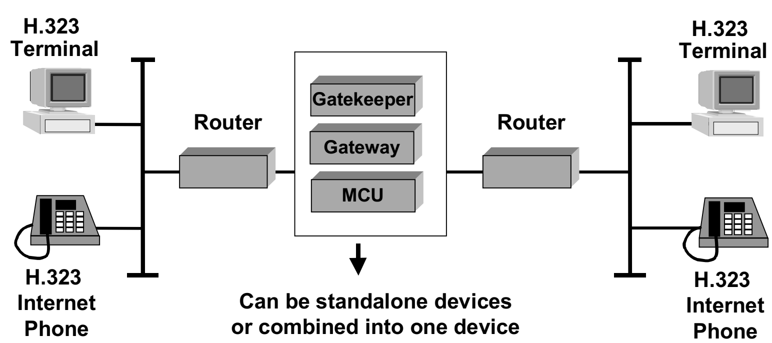
The H.323 is an umbrella term for a suite of standards that were adopted by the ITU-T as a foundation for audio, video, and data communications across IP networks, including the Internet. H.323 provides the specifications required for multimedia products and applications developed by different vendors to interoperate. As long as both the source and the destination equipment and applications are H.323 compliant audio, voice, video, and data communications are possible, independent of which vendor built the equipment.

The H.323 standard defines four major components for IP network communications.

- **Terminals**—Terminals are client endpoints on the LAN that are capable of real-time, two-way voice, video, and data communications. All terminals must support voice communications, but video and data are optional. Terminals can be PCs, Internet phones, video conferencing stations, etc.
- **Gateways**—A gateway is an optional component that provides translation between H.323 LAN terminals and other ITU-T terminals using different transmission formats and communications procedures. For example, gateways establish links between the H.323 LAN terminal and an analog PSTN terminal. Gateways are not required if connections to other networks are not needed.

- **Gatekeepers**—The gatekeeper can be a separate component or the functions of a gatekeeper may be physically incorporated into the terminals, gateways or multipoint control units (MCU). The gatekeeper has three primary functions; address translation, call access control, and bandwidth control. The gatekeeper translates telephone numbers into IP addresses using a translation table. It determines if a call should receive access to the LAN based on whether the incoming call will interfere with the quality of a call in progress. The gatekeeper manages the bandwidth of a network by rejecting connections that will overcome a set threshold established by the network manager. The gatekeeper provides these functions for all H.323 terminals, gateways, and MCUs in what is referred to as a zone.
- **Multipoint Control Units (MCU)**—The MCU provides support for multipoint conferencing between three or more terminals. The MCU determines which audio and video data streams will be multicast and it mixes, switches, and processes the audio, video, and data bits for each participating terminal. All terminals must register with the MCU before participating in a conference call. MCUs may be a standalone component or included as part of other H.323 components.

H.323 Components



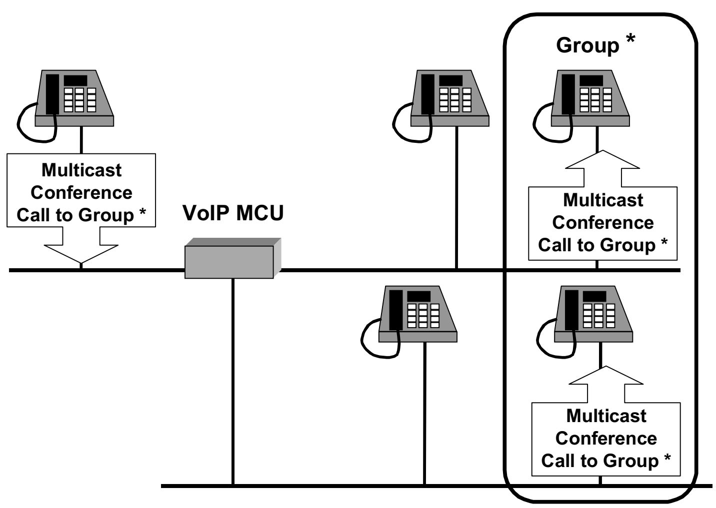
H.323 also provides the standards for speech and video coding methods and for audio mixing. All H.323 terminals must support PCM waveform coding (G.711), although the terminal may also support other coding methods. All terminals must be able to mix different simultaneous audio data streams that often occur during multipoint conferences.

H.323 also employs connection-oriented and connectionless modes of transmission. Connection-oriented transmissions are used over TCP for data transmissions and connectionless transmissions are used over UDP for audio and video messages. Where there are multiple audio and video data streams, H.323 employs IP multicasting and the Real-Time Protocol (RTP).

IGMP

IP multicasting provides a solution that allows a single message to be transmitted to multiple destinations, a multicast group, and still conserve bandwidth. The Internet Group Management Protocol provides procedures for nodes to join and leave a multicast group. IP multicasting supports two of the most important business activities that are rapidly becoming very popular; telephone conference calling and videoconferencing. In the past, a single message must be sent separately to each recipient in a group. If there were four group members, four copies of the message would be sent; one to each member. With IP multicasting using IGMP, only one copy is made and it is forwarded to each member of the group, thus saving up to 75 percent of the bandwidth.

Conference Calling using IP Multicast and the MCU



RTP

Real-time Protocol (RTP) was developed by the IETF (Internet Engineering Task Force) to work on top of UDP for audio or voice and video data that must operate in real-time. Real-time refers to the speed of simulated actions occurring at the same rate expected in the real world. Real-time data, such as audio, voice, or video, is encapsulated in an RTP packet that is preceded with an RTP header. The data and the header are then placed in a UDP packet for delivery.

RTP provides this end-to-end delivery of real-time traffic by providing several services.

- **Time Stamping**—RTP uses time stamping to identify where a packet falls in a sequence of packets transmitted over time. The RTP packet header contains a 32-bit time-stamp field. This field is used to determine if there was any delay in the arrival of the packet. Packets can be placed in a buffer and then, using the time stamp information, the packets can be removed in the correct order. RTP provides information only, it does not guarantee timely delivery of packets.
- **Sequence Numbering**—A 16-bit number is provided in the RTP header to indicate to the receiver if any packets have been lost during transmission. Again, RTP does not guarantee the arrival of a packet, it only provides a mechanism to determine if a packet is missing.
- **Data Identification**—The RTP header contains a 7-bit payload type field that identifies the format of the RTP contents by application. It can identify whether the payload is video or audio and what encoding was used (e.g., PCM).

The Real-time Protocol does not provide any quality of service guarantees. However, it does provide a standardized manner of transporting real-time data traffic, which increases interoperation between applications using the data. For example, when a voice data stream is transmitted an RTP header is attached that indicates the time and sequence of the data packet and what encoding was used (i.e., PCM). At the user end, this information is used to reconstruct the timing and encoding to reproduce the voice signal.

RTCP

The Real-time Control Protocol (RTCP) provides the mechanism to monitor the quality of service during the transmission of RTP packets. In RTCP, the sender transmits reports to the receiver during a session that informs the receiver of the quality of service that the sender is providing.

The contents of these reports contain information on the number of RTP packets and the number of bytes that the sender has sent. The contents also contain information on the transmissions received by the sender from the receiver. This information includes the percentage of lost packets and the total lost packets since the last report.

Based on the information received from these reports, network managers can determine performance issues and make adjustments on the transmission of data.

RSVP

The Resource Reservation Protocol (RSVP) provides a very important mechanism for reserving the bandwidth and the packet priority levels required for successful transmission of real-time traffic that is highly sensitive to transmission delays. RSVP is used to request timely delivery of transmissions and guarantee quality of service for a specific transmission session.

RSVP operates over IP and provides session layer services. Session layer protocols control mechanisms that establish connections and synchronize how two devices communicate. RSVP attempts to make a bandwidth reservation between each intermediate node, usually the end user PC, the servers, and the routers, along a transmission pathway.

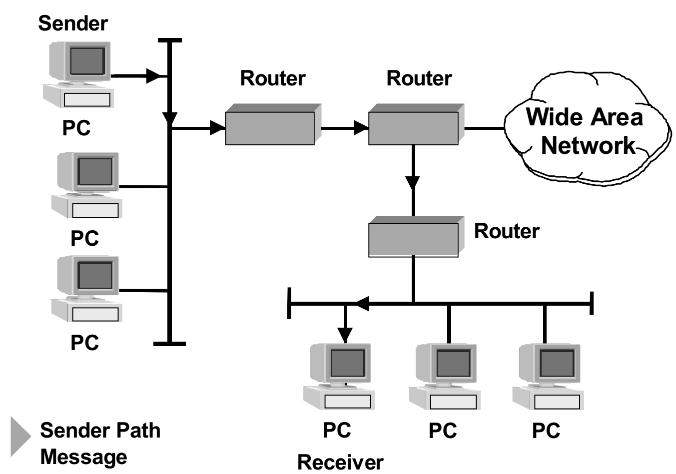
There are two levels of service RSVP may request for different types of real-time data. RSVP may request an assigned priority to packets that allows the packets to travel across the network without wasting time in router queues. Since the packets do not have to wait for evaluation in the routers, the packets can move along the network faster, reducing delays in transmission. These prioritized packets are referred to as controlled-load packets. There is a drawback to this type of service. If the network becomes particularly congested, controlled-load packets will be dumped if there are any no-priority packets waiting to be transmitted.

RSVP also has guaranteed service. With guaranteed service, RSVP requests a specific amount of bandwidth to be reserved for the transport of the packets between intermediary nodes. The more bandwidth reserved, the faster the packets will travel, thus reducing delays. These packets will not be dumped unless the bandwidth capacity of the network is exceeded.

There are basically three steps to transmitting real-time data via guaranteed or priority service.

1. The sender sends messages to the receiver, which contains information about the path from the sender to the receiver. The path messages contain information that is stored in each node along the path, such as the previous hop, characteristics of the sender's traffic, and characteristics of the path (e.g., bandwidth).

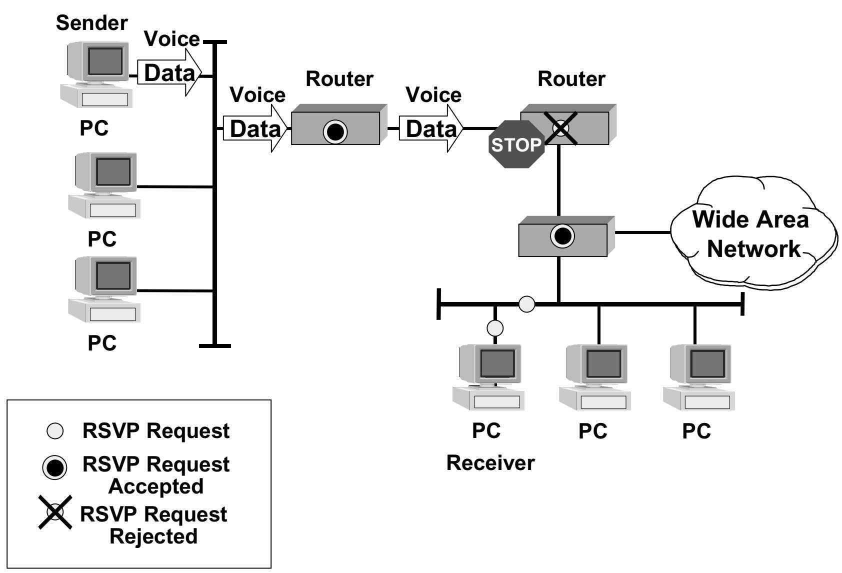
RSVP Step One



2. The receiver is a user who wishes to connect to a networked application or service, i.e., Netmeeting or a video conference. Once the receiver application receives the path message it generates a reservation request that specifies the source and destination addresses, as well as the quality of service requirements (priority or bandwidth). This request is forwarded to the routers on the network. The sender is informed of the path to transmit the data.

3. At each hop, the router either accepts or rejects the request. If all the routers along the path accept the request, then the path information will be maintained for the duration of the data flow from the sender to the receiver. Each router along the path must accept or reject the request. This process of request and acceptance continues from hop to hop until the packet arrives at its destination.

RSVP Steps Two and Three



There are limitations to RSVP. It is always possible that a request will be denied at any hop along the way, essentially preventing the completion of the transmission. RSVP is not a routing protocol. It cannot direct a packet down an alternative path to the destination based on bandwidth availability. Other routing protocols, such as the Routing Information Protocol or Open Shortest Path First, cannot help either. Both protocols route packets along the fastest or shortest path and can not determine what bandwidths are available along any given path. All the intermediate nodes and the end users must also support RSVP.

Check Your Understanding

- In the discussion about VoIP, there is a reference to issues surrounding the implementation of VoIP. List the issues mentioned.
- Speculate why packet size and packet loss can affect the quality of a VoIP call.
- Speculate why security is an issue with VoIP.
- Speculate why reliability is an issue for VoIP.

Try It Out: Test for Latency and Packet Loss

Materials Needed:

- Windows 95 PC
- Internet Connection
- MS-DOS Ping
- Stop Watch or Watch with second hand
- Any Word Processor (e.g., MS Word) or Spreadsheet (e.g., MS Excel)
- Pen/Pencil and Paper
- Student Portfolio

Network managers and designers must determine if the existing network infrastructure is capable of handling the implementation of Voice over IP. There are several factors they must investigate prior to implementation, such as bandwidth capacity, delay or latency, and dropped packets.

Most users notice problems with the quality of a call when the delay or latency reaches 250 milliseconds (ms) or higher. Additionally, packet loss should be under 10 percent, since higher packet losses can break up the voice and result in an unintelligible conversation.

In this activity, you will test over time the amount of latency, the time required for a signal to travel from one point on a network to another point, that occurs between your PC and a point on the Internet. You will also measure the level of packet loss over time.

Materials:

- Internet connected PC
- Windows 95 or 98 installed on the PC
- MS-DOS Ping

1. Work in a team of at least two students. One student is needed to run the ping command, the other student is needed to record the displayed statistics.
2. Open an Internet connection from your PC.
3. Determine what Internet site to ping. It could be any location, such as www.nortelnetworks.com. You do not need to have the IP address, but you can use one if you have that information. Or alternatively, you can acquire the IP address by using the ping command.
4. Select three days during the week that you will repeat this process. If your school is on a rotating block schedule, pick days when your class meets at a different time each day. In reality, a network manager or designer would pick peak times of the day to run this test to catch statistics that represent the heaviest network traffic.

5. If you want to use the ping command to acquire the IP address of a location follow these directions.
   a. Click Start.
   b. Click Programs.
   c. Click MS-DOS Prompt.
   d. When the C: prompt appears, type ping and the domain address of the location for which you desire to get the IP address. For example, ping www.nortelnetworks.com.
   e. The IP address will appear with the ping statistics.

   (Follow the steps outlined in Step 2 a-c each time you run this test during the three-day period.)

6. Once the C: prompt appears, type ping -n 10 <the IP address or domain address>. For example, ping -n 10 www.nortelnetworks.com. This will run the ping command 10 times before the final statistics are displayed.

7. Repeat Step 5 every 20 seconds for a 5 minutes each day. For each ping display, record the following information:
   a. Number of packets sent
   b. Number of packets received
   c. Number of packets lost
   d. The percent of lost packets
   e. Minimum round trip time in milliseconds
   f. Maximum round trip time in milliseconds
   g. Average round trip time in milliseconds

8. Using a spreadsheet program, record the data you have collected over the three-day period.

9. Determine the averages of each of the measured parameters and compare the results over the three-day period.

10. What patterns did you notice? Were there periods of greater packet loss or longer round trip travel time?

11. Looking just at the percentage of packet loss and the round trip travel time, do you think the network connection between your PC and the Internet point will support VoIP well? Why?

12. Write a conclusion of what your results demonstrated and a recommendation to implement VoIP or not and why.

13. If you do not recommend the implementation of VoIP, speculate what actions could be taken to resolve the issues that are preventing the network from supporting VoIP.

14. Include this activity in your portfolio.

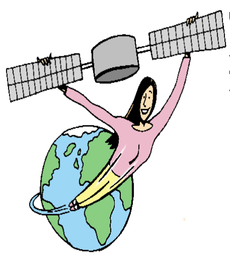
Rubric: Suggested evaluation criteria and weightings:

Criteria	**%**	**Your Score**
Professional team work and cooperation	25	
Organized collection of data and spreadsheet analysis	50	
Thorough analysis of data and written recommendations suitable for inclusion in portfolio	25	
TOTAL	**100**	

Stretch Yourself: Choose An Activity

Materials Needed:

- Windows 95 PC
- Internet Connection (optional)
- Any Word Processor (e.g., MS Word) or Spreadsheet (e.g., MS Excel)
- Pen/Pencil and Paper
- Optional Materials: Poster Board; 2 telephones with 9-volt batteries and 300 Ohm resistors; microphone and compression software



There are many facets to telephony that are interesting. In this activity, you may choose to design your own activity around the subject of telephony. The requirements for completion are simple.

1. The activity must center on telephony. Choose a subject that interests you; that gets you excited.

2. The results of your activity must be shared with your fellow classmates. You may create a poster presentation with a written report or do an oral presentation or create a game the demonstrating concepts you learned. How you share your results is negotiable with your teacher. Your grade depends on the project's level of creativity and professionalism.

3. Your grade also depends on how thoroughly you research your subject and the quality of your resources. All resources must be documented.

Below are suggested activities, but you are not restricted to these choices. Remember that you are required to share your results and experiences with the entire class.

a. Create your own telephone network. It is simple to create your own intercom telephone system using a 9-volt battery, a 300-ohm resistor, and two telephones.

b. Research PBX, Centrex, and Key telephone systems. Evaluate each system as a voice communications solution.

c. There are other protocols either in use or under development that can be implemented to support VoIP. Research DiffServ, Session Initiation Protocol (SIP), and Media Gateway Control Protocol (MGCP) and compare them against the protocols discussed in this lesson.

d. Locate audio samples of different voice compression techniques on the Internet. Use key search words such as H.323, VoIP demos, Pulse Code Modulation, audio samples, and digital and analog voice demos, to narrow your search to web sites of interest. Compare the quality of each sample. Or create your own audio samples using a microphone and compression software on your PC.

Rubric: Suggested evaluation criteria and weightings:

Criteria	%	Your Score
Thorough research of the subject	20	
Creative and professional presentation to fellow classmates	60	
Documented resources of high quality	20	
TOTAL	**100**	

Network Wizards: WAN Core Network Design

Materials Needed:

- Calculator (optional)
- Pen/Pencil and Paper
- Student Portfolio
- Student Network Design Proposal Working Draft

Part One: The Red Rock Unified School District Case Study

As a network designer, you are responsible to determine solutions for the wide area network requirements of your client. In Unit 2, Lesson 1, you determined Red Rock Unified School District's bandwidth requirements for conducting video conferencing from its four high schools. From that information, you made recommendations as to what WAN solutions could be implemented.

In part one of this activity, you will continue to resolve Red Rock's WAN requirements and create a WAN Core Network Design document. This diagram requires an overall picture of what the WAN will look like. It should have lines drawn between sites indicating the WAN connections, backup solutions, remote access, and access to the Internet. Although this diagram normally includes the location and number of routers required, you are not required to make this determination.

You may work in teams or individually.

RRUSD's Network requirements:

a. Red Rock Unified School District is comprised of four high schools; Blue HS, White HS, Green HS, and Red HS.

b. Each high school has 30 classrooms. Each classroom has one network connection to the school LAN and the district's WAN that can be used by the teacher and the students. Each high school has a computer lab that accommodates 30 students, each with a network connection to the school LAN. Each high school has 15 administrators who have a network connection to the school LAN and the district's WAN.

c. The Central Office has 50 administrators with connections to the Central Office LAN and the district WAN.

d. Over the next five years, the district expects a 10% total growth in utilization of the WAN.

e. Each high school must be connected to the School District's Central Office. The Central Office houses the database of all the student's grades and records.

f. Each school must have access to the Internet for student research. Teachers and administrators (both in the schools and the Central Office) need remote access to the district office and school LANs via the Internet.

g. The bandwidth requirement for day to day applications is 3 Kbps per user.

h. Each school depends on its connection to the Central Office and to the Internet. It is important that the District WAN has backup solutions, just in case a line is not accessible.

i. Each high school has a single video conferencing PC station capable of running the video conferencing software CUSeeMe. The district has already determined that they will not be using Timbuktu Pro.

j. An analysis of the current network and in-district telephone calls has been conducted and the following information is available:

   i. The average round trip latency is 200 ms.

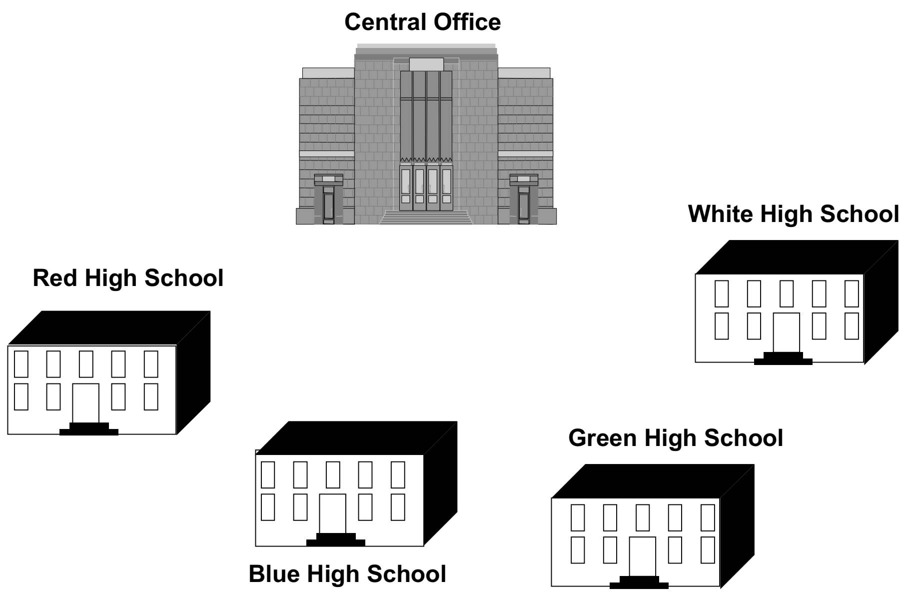
   ii. The average packet loss is 8 percent.

   iii. The maximum number of calls between the district's sites is 50.

   iv. The bandwidth requirement for each call is 8 Kbps.

   v. The district is currently using ISDN-PRI for their WAN connections.

1. Using the diagram below, draw lines between the high school and the Central office indicating the WAN connections. Do you think that each school must be directly connected to the other schools or should the schools reach each other through the Central Office?



2. Think about how each school will access the Internet. Will they each have separate Internet connections or will they connect to the Internet via the District's WAN connection? Indicate on the diagram how the schools will access the Internet. How will the teachers and administrators gain access to the district network?
   a. Calculate the total spare capacity by subtracting the peak network load from total available capacity from site to site (high school to high school or central office).
   b. Calculate the bandwidth required for the maximum number of simultaneous calls between the sites by multiplying the number of calls by the bandwidth required per call.
   c. The amount of spare capacity (#A) must be higher than the maximum bandwidth required for the peak number of calls (#B) for a network to handle the implementation of VoIP.
   d. Review the statistics on network latency and packet loss. Remember that the latency rate should be less than 200 ms and the packet loss less than 10 percent.

3. Make the necessary calculations to fill in the chart below with the bandwidth requirements for the WAN. (Round up all numbers.)

Maximum Bandwidth Requirements		
1.	All classrooms per school	
2.	All computer lab users per school	
3.	All administrators per school	
4.	Peak Network Load (Add 1,2, and 3 together) per school	
5.	Peak Network Load for all schools	
6.	Peak Network Load for the Central Office (assume all users on network at same time)	
7.	Peak Network Load for the District	
7.	Five-year projection of increase in peak network load per school and district.	
8.	Current WAN Capacity (ISDN-PRI)	
9.	Maximum telephone calls	
10.	Videoconference to all four schools (from Unit 2 Lesson 1)	

4. Given the information you have calculated, will the current WAN solution (ISDN-PRI) support the district's bandwidth requirements, without video conferencing and VoIP? Why?

5. Will the current WAN support video conferencing? Why?

6. Will the current WAN support VoIP? Why?

7. Make recommendations for WAN solutions that will accommodate not only the daily applications used by each user, but also video and VoIP. Support your recommendation with specifications about the WAN solution, including cost comparisons.

8. Indicate on your Core WAN diagram the WAN solutions you recommend for each network connection, including Internet access and remote access.

9. Be prepared to share your diagram and defend your recommendations with the class.

10. Place the results of this activity in your portfolio.

Rubric: Suggested evaluation criteria and weightings:

Criteria	**%**	**Your Score**
Critical analysis of network requirements of case study with accurate calculations	50	
Quality Core WAN Network Design presentation with a defendable recommendation	50	
TOTAL	**100**	

Part Two: Network Design Portfolio Case Study

In Part One, you completed a Core WAN Network Design for the Red Rock Unified School District. Using this experience, gather the same information from your portfolio case study and create a WAN network design to include in your final network design proposal.

Remember that you are a novice and there is no expectation that you will understand all the complications that can arise when creating a WAN network design. This is a learning experience.

Summary

Voice Over IP Technologies

In this lesson, you learned the following:

- The three components of human speech
- Waveform, source, and hybrid coding methods used to digitize and compress analog signals
- The basic function of the PBX
- The advantages of implementing VoIP
- The services provided by H.323, IGMP, RTP, RTCP, and RSVP
- How to diagram and defend a Core WAN Network Design

Review Questions

Voice Over IP Technologies

Part A:

1. What types of sounds are found in the word "sharp"?
    a. Voiced
    b. Unvoiced
    c. Plosive
    d. A and B only
    e. A, B, and C
2. Voiced sounds are produced air is released with a sudden burst.
    a. True
    b. False
3. The sounds "p" and "b" are examples of plosive sounds.
    a. True
    b. False

4. Unvoiced sounds are produced when…
    a. The vocal cords vibrate and interrupt the flow of air from the lungs.
    b. When air passes over an obstacle in the mouth.
    c. When air is released with a sudden burst.
    d. A and B only
    e. None of the above
5. Telephone systems have a frequency range of 300 Hz to 3.4 kHz because this range provides high-quality voice without wasting bandwidth by including levels the human ear cannot distinquish.
    a. True
    b. False

Part B:

1. What is the most commonly used voice compression technique?
2. Briefly describe the advantages and disadvantages of waveform coding.
3. Briefly describe the advantages and disadvantages of source coding.
4. Briefly describe the advantages and disadvantages of hybrid coding.
5. Briefly describe Pulse Code Modulation.
6. Diagram the steps involved with PCM.

7. How is ADPCM different from PCM?
8. Briefly describe Code Excited Linear Prediction.
9. Diagram CELP.
10. Explain the advantage of using Conjugate-Structure Algebraic Code Linear Prediction (CS-ACELP) over ADPCM.

Part C:

1. Briefly describe how the PBX functions.
2. Diagram the PBX network.

Part D:

Identify each statement as either True or False.

1. ________	Implementing VoIP is one step to unifying networks.
2. ________	Toll charges are collected for Internet calls not directed via a VoIP gateway.
3. ________	Local telephone charges are always reduced using VoIP.
4. ________	Implementing VoIP usually reduces the expense of placing calls to overseas locations.
5. ________	VoIP allows multiplexing several telephone calls on one physical link.
6. ________	VoIP can be used to make phone-to-phone calls over the Internet.
7. ________	PC-to-PC telephone calls are not possible using VoIP.
8. ________	PC-to-Telephone calls are possible using and ITSP voice over IP gateway.
9. ________	Businesses must stop using their PBX systems in order to implement VoIP.
10. ________	Transmitting calls over the Internet bypasses the PSTN.

Part E:

1. Briefly describe RTP.
2. Briefly describe RTCP.
3. H.323 defines four major components for IP network communications. List the four components and briefly describe the purpose of each.
4. Diagram the relationship of the major components of H.323.
5. Briefly describe IGMP and how it is used for VoIP.

6. Diagram and describe the steps in RSVP.

Scoring

Criteria	%	Your Score
Part A: Identify the three components of human speech.	20	
Part B: Compare waveform, source, and hybrid coding methods used to digitize and compress analog signals.	20	
Part C: Understand the basic function of the PBX.	10	
Part D: Identify the advantages of implementing VoIP.	20	
Part E: Identify the services provided by H.323, IGMP, RTP, RTCP, and RSVP.	30	
TOTAL	**100**	
Try It Out:	**100**	
Stretch Yourself:	**100**	
Network Wizards: Diagram and defend a Core WAN Network Design.	**100**	
FINAL TOTAL	**400**	

Resources:

Black, U. (2000). Voice Over IP. Upper Saddle River, New Jersey: Prentice Hall PTR.

Cray, A. (1998). Voice Over IP: Hear's How. Available Online: http://www.data.com/roundups/voiceip.html.

Goncalves, M. (1999). Voice Over IP Networks. New York: McGraw-Hill.

Held, G. (1998). Voice Over Data Networks. New York: McGraw-Hill.

Nortel Networks. (1999). Voice Fundamentals. Santa Clara, California: Nortel Networks.

Roberts, E. (1997). RSVP: A Priority Problem? Available Online: http://www.data.com/roundups/rsvp.html.

Stardust Technologies. (1997). Higher Level Protocols used with IP Multicast: An IP Multicast Initiative White Paper. Available Online: http://www.ipmulticast.com/community/whitepapers/highprot.html.