Chapter 5

Voice Communication Concepts and Technology

By Masud-ul-Hasan

Objectives

- Investigate PSTN.
- Study and understand digital voice communication and digitization.
- Alternatives of PSTN.
- Understand PBXs (Private Branch eXchange).
- Understand CTI (Computer Telephony Integration) and voice services.
- Introduce wireless voice transmission services.

GOAL: Study the business behind voice communication.

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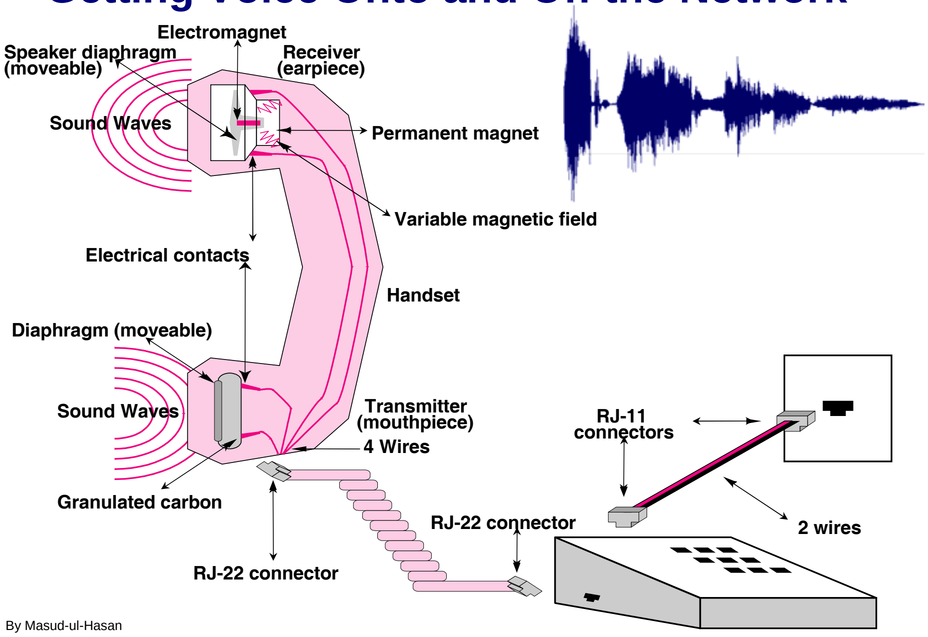
Voice Network Concepts

- Telephone calls are connected from source via circuit switching.
- Circuit switching originally meant that a physical electrical circuit was created from the source to the destination.
- The modern telephone system is commonly known as the **Public Switched Telephone Network** or **PSTN**.

Basic Concepts

- Voice consists of sound waves of varying frequency and amplitude.
- The *transmitter* (mouthpiece) part of phone handset converts voice into electrical signals to be transmitted onto the analog network.
- The *receiver* (earpiece) part of a handset works the opposite of the transmitter i.e., converts electrical signals into voice that received from the analog network.

Getting Voice Onto and Off the Network



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Basic Concepts

- POTS (Plain Old Telephone Service) employs analog transmissions to deliver voice signals from source to destination.
- POTS uses a bandwidth of 4000 Hz, but *guardbands* limit the useable range to 300-3400 Hz.
- Channels are separated by "guardbands" (empty spaces) to ensure that each channel will not interfere with its neighboring channels.
- Today, the local loop is still analog, but high-capacity digital circuits link the exchanges or Central Offices (COs) with each other.

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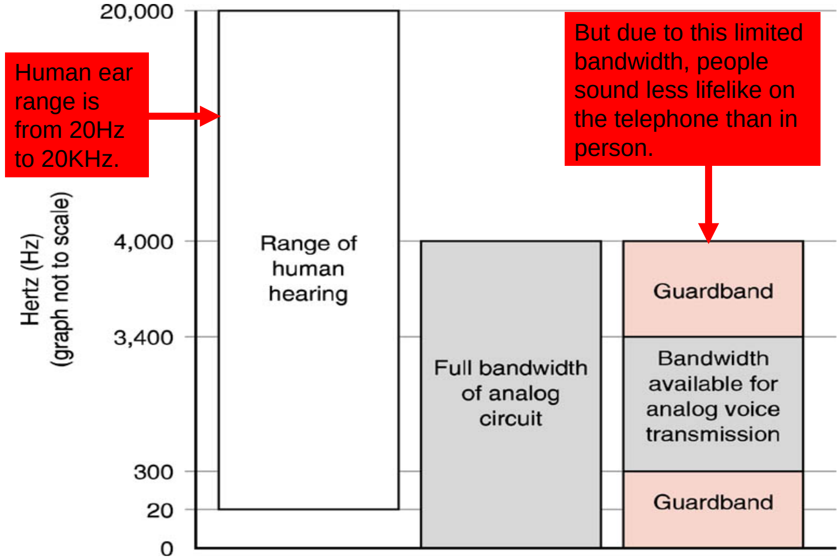
Voice Bandwidth



Voice Network Concepts

- PSTN
- Network hierarchy
- Signaling and dial tone
- Control and management

From History

- At the turn of the 20th century, Blake wall phone . *(Image courtesy of Nortel Networks.)*

From History

- In 1886, this 50-line magneto switchboard, made by Bell Telephone of Canada, was used to switch voice calls in small localities. These instruments were the beginning of the worldwide PSTN. *(Image courtesy of Nortel Networks.)*

Public Switched Telephone Network (PSTN)

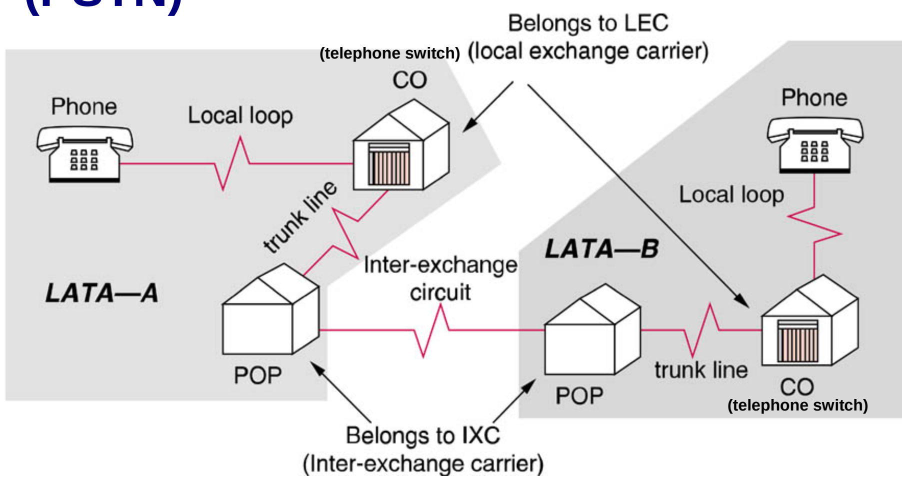


Figure 2-3 Basic Telecommunications Infrastructure

Public Switched Telephone Network (PSTN)

Telephone calls are established by a device located at CO known as telephone switch.

A central office (CO) is a facility belonging to local phone company in which calls are switched to their proper destination.

Long distance carrier doing business in a given LATA maintain a switching office in that LATA known as POP or point of presence. POP handles billing information & routes the call over long distance carrier's switched network to its POP in the destination LATA.

The circuits between a residence or business and Central Office (CO) are known as local loops.

A central office (CO) is a facility belonging to local phone company in which calls are switched to their proper destination.

Local loop: This is only remaining analog circuit in PSTN.

All voice traffic destined for outside the local LATA must be handed off to the long distance carrier or IXC.

Circuit between POPs may be via satellite, microwave, fiber optic cable, traditional wiring, or some combination of these media.

The telephone switch routes calls to the destination telephone. Requested destinations are indicated by dialing a series of numbers. Which tell the switch whether the call is intra-LATA, or inter-LATA.

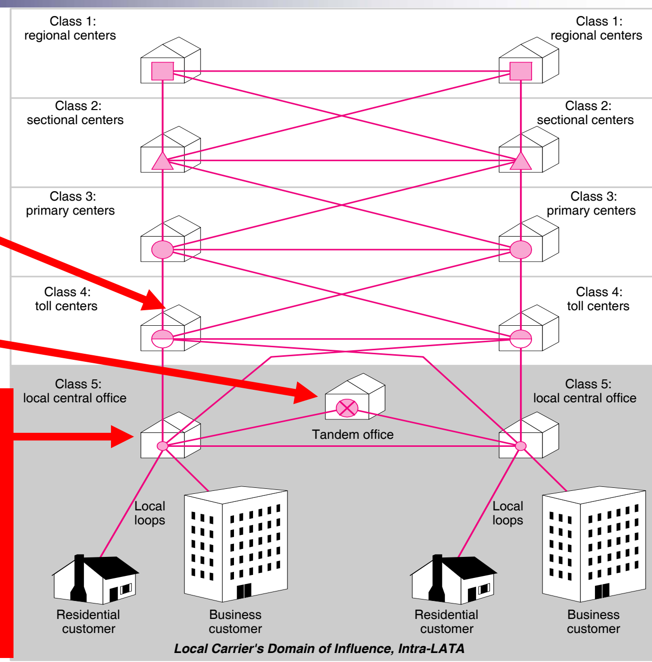
Figure 2-3 Basic Telecommunications Infrastructure

Representative Voice Network Hierarchy

This is POP, implies the long distance billing and switching activities.

This establishes the intra-LATA circuit & also handles billing procedures for long distance calls.

This is an end office (CO) in hierarchy contains a switch that processes incoming calls, determines the best path to call destination, & establishes the circuit connection.



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Representative Voice Network Hierarchy

- Circuit redundancy offers multiple alternatives paths for call routing which is a basic idea in voice network hierarchy.
- If no paths are directly available, then the call is escalated up to the network hierarchy to the next level of switching office.
- The overall desire is to keep the call as low as possible in the hierarchy for quicker call completion and maximization of the cost-effective use of switching offices (i.e. trying to use the least expensive and less number of switching offices).
- Higher levels on network hierarchy imply greater switching and transmission capacity as well as greater expense. When calls cannot be completed directly, ***Class 4 toll centers*** turn to ***Class 3 primary centers*** that subsequently turn to ***Class 2 sectional centers*** that turn finally to ***Class1 regional centers***.

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Telephone Number Plans

- Telephone numbers are built using a hierarchical address method. Numbers tell whether the call is local, intra-LATA, or inter-LATA.
- Divided into 3 basic parts: a 2-digit area code starting with 0, a 3-digit exchange, & a 4-digit subscriber number.
- To make a call, at a minimum the exchange plus the subscriber number must be dialed. But if the call is within the PBX then only 4(or less)-digit subscriber number will be dialed.
- If the call is to a destination outside the source phone's code, destination area code must be dialed as well.

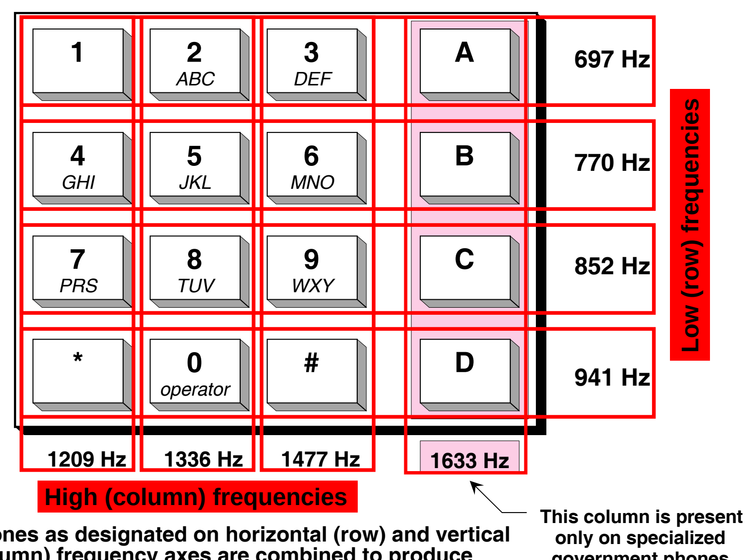
Signaling and dial tone

- Numbers are dialed by:
  - Rotary type phones: **pulses**
    - Generate electrical pulses, 1 pulse for digit 1, 2 pulses for digit 2, and so on, 10 pulses for digit 0.
  - Push Button type phones: **tones**
    - Dual-Tone Multi-Frequency tones (DTMF).
    - Tones are used for much more than merely dialing destination phone numbers. Also used to enable specialized services from PBX's, carriers, banks, information services, and etc.

Pulse Dialing

- Pulse dialing sends digit information to the CO by momentarily opening and closing (or breaking) the local loop from the calling party to the CO.
- This local loop is broken once for the digit 1, twice for 2, etc., and 10 times for the digit 0. As each number is dialed, the loop current is switched on and off, resulting in a number of pulses being sent to the local CO.

Tone Dialing with DTMF



Two tones as designated on horizontal (row) and vertical (column) frequency axes are combined to produce unique tones for each button on the keypad

Tone Dialing with DTMF

High Freq. / Low Freq.	1209Hz	1336Hz	1447Hz	1633Hz
697Hz	1	2	3	A
770Hz	4	5	6	B
852Hz	7	8	9	C
941Hz	*	0	#	D

- Pressing a key on a phone's keypad generates two simultaneous tones, one for the row and one for the column.
- These are decoded by the CO to determine which key was pressed.

System Signaling

- In addition to carrying the actual voice signals, the telephone system must also carry information about the call itself.
- This is referred to as **system signaling** or **inter-office signaling**.
- There are two approaches to system signaling: **in band** and **out of band**.

In-band Signaling

- In this system, the signals are sent on the same channels as the voice data itself.
- Dial tone makes sure that telephone switch at CO is ready to serve.
- Dialing the number sends the phone number across in the voice bandwidth.
- If the called party answers the phone, the remote phone switch comes off the hook and the connection is established.

Out-of-Band Signaling

- In this system, the signals are sent on a separate channel as from the voice.
- Monitoring of circuit status notification and re-routing in the case of alarms or circuit problems.
- The worldwide approved standard for out-of-band signaling is Signaling System 7 (SS7).

Signaling System 7 (SS7)

- It controls the structure and transmission of both circuit-related and non-circuit related information via out-of-band signaling between central office switches.
- It delivers the out-of-band signaling via a packet switched network physically separate from the circuit switched network that carries the actual voice traffic.
- It is nothing more than a packet-switched network.

Signaling System 7 (SS7)

- **Alternate Billing System (ABS)** allows a long-distance call to be billed to a calling party, to the receiver (call collect), or to a third party.
- **Custom Local Area Signaling Service (CLASS)** is a group of services that allows many services local access to the customer's telephone. E.g., call waiting, call forwarding, call blocking, etc.
- **Enhanced 800 services** allows 800-number portability. Originally, 800 numbers were tied to a specific area code and long-distance provider.
- **Intelligent Call Processing (ICP)** enables the customers to reroute incoming 800 calls among multiple customer service centers, geographically dispersed, in seconds. This is transparent to the caller.

Analog vs. Digital Transmission

- Transmissions can be either analog or digital.
  - **Analog transmissions**, like analog data, vary continuously. Examples of analog data being sent using analog transmissions are voice on phone, broadcast TV and radio.
  - **Digital transmissions** are made of square waves with a clear beginning and ending. Computer networks send digital data using digital transmissions.
- Data can be converted between analog and digital formats.
  - When digital data is sent as an analog transmission **modem** (modulator/demodulator) is used.
  - When analog data is sent as a digital transmission, a **codec** (coder/decoder) is used.

Voice Digitization

- The analog POTS system has been supplanted in the modern telephone system by a combination of analog and digital transmission technologies.
- Converting a voice conversation to digital format and back to analog form before it reaches its destination is completely transparent to phone network users.
- There are a limited ways the electrical pulses can be varied to represent an analog signal.

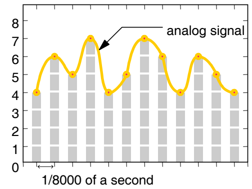
Voice Digitization Techniques

- Pulse Amplitude Modulation: (PAM)
  - Varies the amplitude of the electrical pulses.
  - Used in earlier PBX's.
- Pulse Duration Modulation: (PDM/PWM)
  - Varies the duration of electrical pulses.
- Pulse Position Modulation: (PPM)
  - Varies the duration between electrical pulses.

Voice Digitization:

PAM

PAM: Pulse Amplitude Modulation



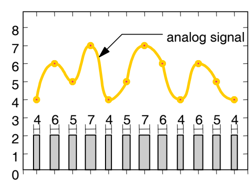
Variable: Pulse amplitude

Constants: Pulse duration, pulse position

Sampling rate = 8,000 times/second

PDM

PDM: Pulse Duration Modulation

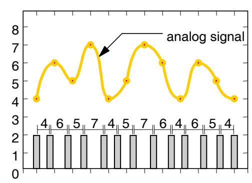


Variable: Pulse duration

Constants: Pulse amplitude, pulse position

PPM

PPM: Pulse Position Modulation



Variable: Pulse position

Constants: Pulse amplitude, pulse duration

Pulse Code Modulation

- The most common method used to digitize voice is Pulse Code Modulation (PCM).
- No matter how complex the analog waveform happens to be, it is possible to digitize all forms of analog data, including full-motion video, voices, music, telemetry, and virtual reality (VR) using PCM. Native of .wav
- The analog signal amplitude is sampled (measured) at regular time intervals. The sampling rate, or number of samples per second, is several times the maximum frequency of the analog waveform in cycles per second or hertz.

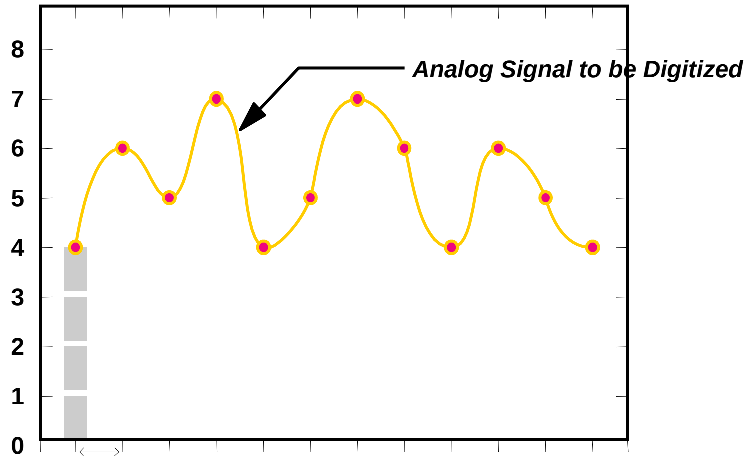
How to obtain Pulse Code Modulation?

- The instantaneous amplitude of the analog signal at each sampling is rounded off to the nearest of several specific, predetermined levels (called quantization).
- The number of levels is always a power of 2, e.g., 4, 8, 16, 32, 64, or 128. These can be represented by bits.
- The output of a pulse coder is thus a series of binary numbers, each represented by some power of 2 bits.
- At the destination (receiver end) of the communications circuit, a pulse decoder converts the binary numbers back into pulses having the same quantum levels as those before the coder.

Step 1: Sample Amplitude of Analog Signal

Amplitude in example, at first sample position, is 4

8 possible amplitudes are actually 256 (2^8) amplitudes in PCM



1/8000 of a second

Sampling rate = 8,000 times/second

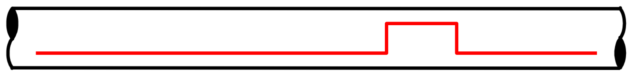
Step 2: Represent Measured Amplitude in Binary Notation

$(0000\ 0100)_2 = (4)_{10}$

Power of 2	2^7	2^6	2^5	2^4	2^3	2^2	2^1	2^0		
Value	**128**	**64**	**32**	**16**	**8**	**4**	**2**	**1**		
Binary notation	**0**	**0**	**0**	**0**	**0**	**1**	**0**	**0**	**=**	**4**

8 bits = 1 byte

Step 3: Transmit Coded Digital Pulses Representing Measured Amplitude



0	0	0	0	0	1	0	0

8 transmitted bits = 1 transmitted byte = 1 transmitted sampled amplitude

In this way next few samples will be:

$(0000\ 0110)_2 = (6)_{10}$
$(0000\ 0101)_2 = (5)_{10}$
$(0000\ 0111)_2 = (7)_{10}$

T-1 and E-1

- PCM uses:
  - 8000 samples/sec and 8 bits/sample, so for 1 digitized voice: 8000 x 8 = 64,000 bps is the required bandwidth.
  - This is known as a DS-0 (basic unit of voice data trans.)
  - 24 DS-0s = 24 x 64 Kbps = 1,536 Kbps = 1.536 Mbps
  - 1 framing bit/sample x 8000 samples/sec = 8000 framing bps = 8 Kbps
  - 8 Kbps + 1,536 Kbps = 1,544 Kbps = Trans. cap. of T-1
  - T-1 (1.544 Mbps) can carry 24 simultaneous voice conversations digitized via PCM.
- European equivalent standard is E-1 (2.048Mbps)

Adaptive Differential PCM (ADPCM)

- Each voice channel uses 4 bits instead of 8 bits.
- So, for 1 digitized voice: 8000 x 4 = 32,000 bps is the required bandwidth. The standard for 32-Kbps is known G.721
- ADPCM supports 48 simultaneous conversations over a T1 circuit.
- The G.721 is used as a quality reference point for voice transmissions (Toll Quality).
- ADPCM is used to send sound on fiber-optic long-distance lines as well as to store sound along with text, images, and code on a CD-ROM.

Voice Compression

- ADPCM is also known as voice compression technique because of its ability to transmit 24 digitized voice conversations in half the bandwidth required by PCM.
- Other more advanced techniques employ DSPs (Digital Signal Processors) that take the PCM code & further manipulate and compress it.
- DSPs are able to compress voice as little as 4800 bps.
- Efficiency: 13 times more than PCM.
- Voice compression may be accomplished by stand alone units, or by integral modules within other equipment.

Voice Transmission Alternatives to PSTN

- Although the PSTN is the cheapest and most effective way to transmit voice, alternative methods are do exist.
- Some of them are:
  - Voice over the Internet (VoIP)
  - Voice over Frame relay (VoFR)
  - Voice over ATM (VoATM)

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Voice over the Internet (VOIP)

- VOIP refers to any technology used to transmit voice over any network running the IP protocol (in packets).
- It is not confined to use on the Internet only, can be used in any of the following:
  - Modem based point-to-point connections
  - Local area networks (LANs)
  - Private Internets (Intranets)
- It can be successfully deployed with:
  - VOIP client software
  - using a PC with sound card, microphone, and speakers
  - gateways are being established to allow Internet voice callers to reach regular telephone users as well.

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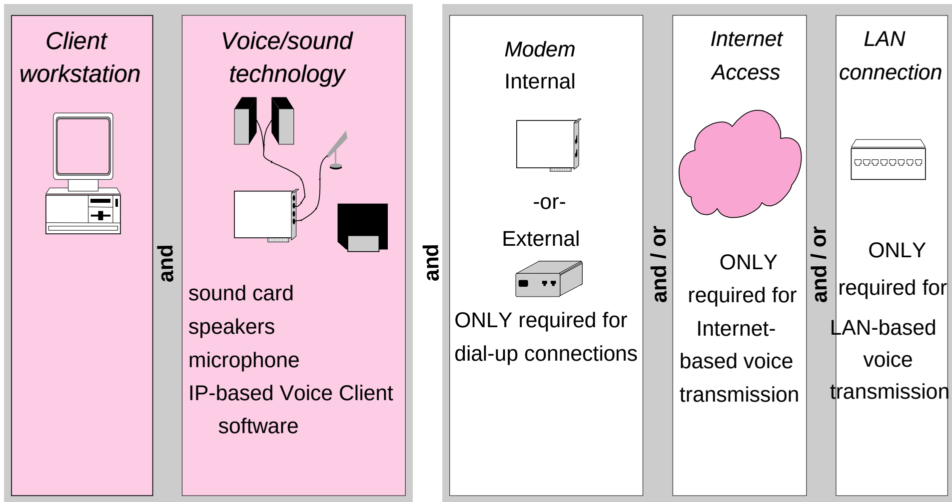
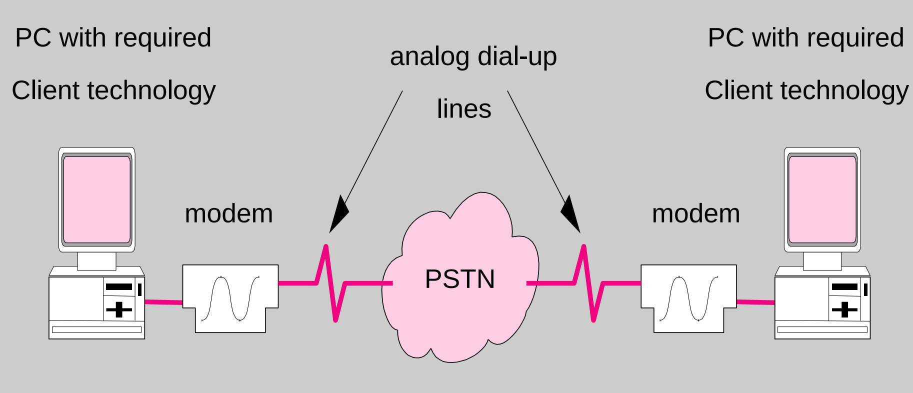
VOIP Transmission Technology

REQUIRED CLIENT TECHNOLOGY



VOIP Transmission Topologies

POINT-TO-POINT/MODEM-TO-MODEM

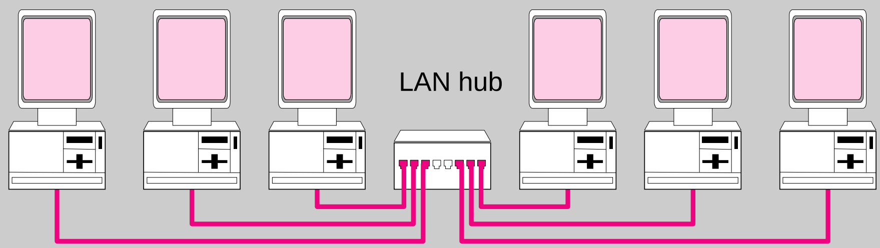


VOIP Transmission Topologies

LOCAL AREA NETWORK

LAN attached PCs with required Client technology.

IP protocols REQUIRED



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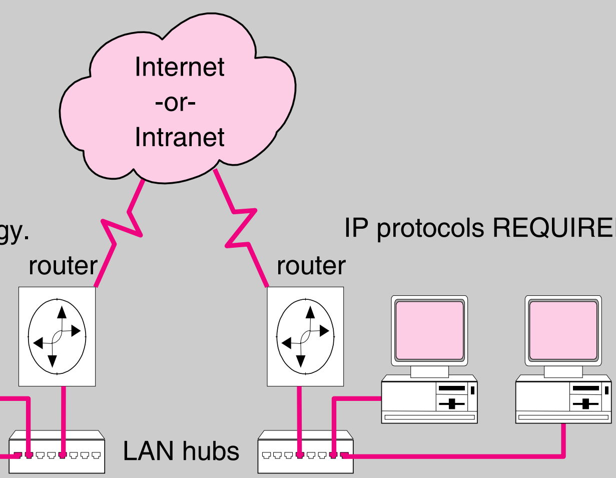
VOIP Transmission Topologies

INTERNET/INTRANET



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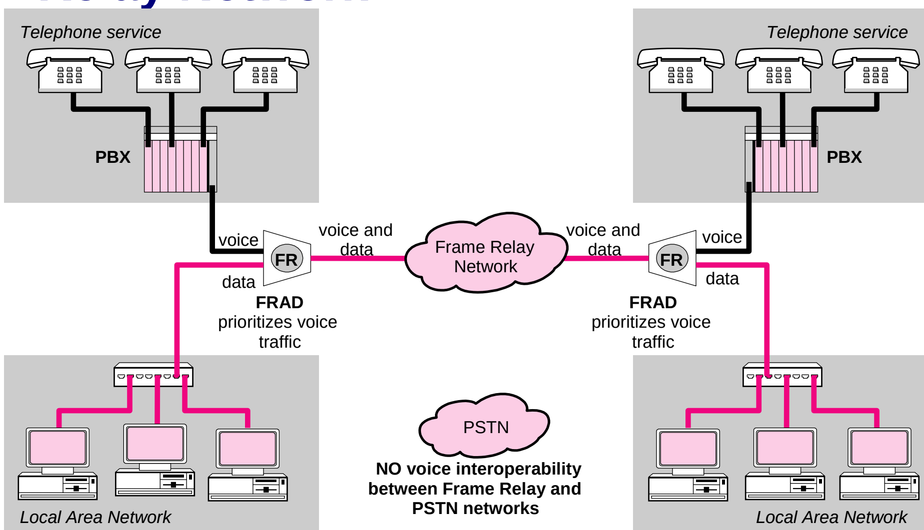
Voice over Frame relay

- Initially deployed for data transmission but is now capable of delivering voice transmissions as well.
- Frame relay encapsulates segments of a data transfer session into variable length frames.
- For longer data transfers, longer frames and for shorter data transfers, shorter frames are used.
- These variable length frames introduce varying amounts of delay resulting from processing by intermediate switches on the frame relay network.
- This variable length delay works well with data transmission but is not acceptable in voice transmission because it is sensitive to delay.

Voice over Frame relay

- Frame relay access device (FRAD) accommodates both voice and data:
  - Voice prioritization: FRAD distinguish between voice and data traffic (because of tagging), priority given to voice over data
  - Data frame size limitation: long data frames must be segmented into multiple smaller frames to limit delays
  - Separate voice and data queues: within the FRAD
- Voice conversations require 4 – 16 Kbps of bandwidth.
- This dedicated bandwidth is reserved as an end-to-end connection through frame relay network called Permanent Virtual Circuit (PVC).
- Voice conversation can take place only between locations directly connected to a frame relay network.
- No current standards defined between frame- relay networks and the voice based PSTN.

Voice Transmission over a Frame Relay Network



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Voice over ATM

- ATM (Asynchronous Transfer Mode) is a switched-based WAN service using fixed-length frames (called cells).
- Fixed length cells assures fixed time processing by ATM switches enabling predictable delay and delivery time.
- Voice transmitted using Constant Bit Rate (CBR) bandwidth reservation scheme.
- CBR does not make optimal use of bandwidth because of moments of silence.
- Most common method: reserve a CBR of 64Kbps for one conversation digitized via PCM.

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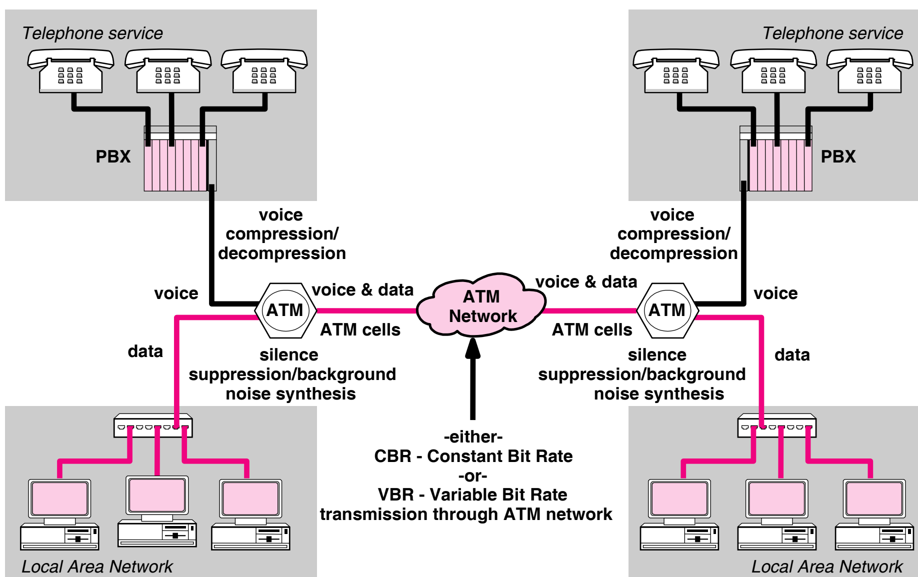
Optimizing voice over ATM

- Voice Compression: Achieved via ITU, G series of standards, algorithms vary in amount of bandwidth required to transmit toll quality voice:
  - G.726: 48, 32, 24 or 16 Kbps
  - G.728: 16 Kbps
  - G.729: 8 Kbps
- Silence suppression: Cells containing silence are not allowed and replaced at the receiver with synthesized background noise. It reduces the amount of cells transmitted for a given voice conversation by 50%.
- Use of VBR (Variable bit rate): Combines positive attributes of both voice compression and silence suppression. By using bandwidth only when someone is talking, remaining bandwidth is available for data transmission or other voice conversations.

Voice Transmission over an ATM Network



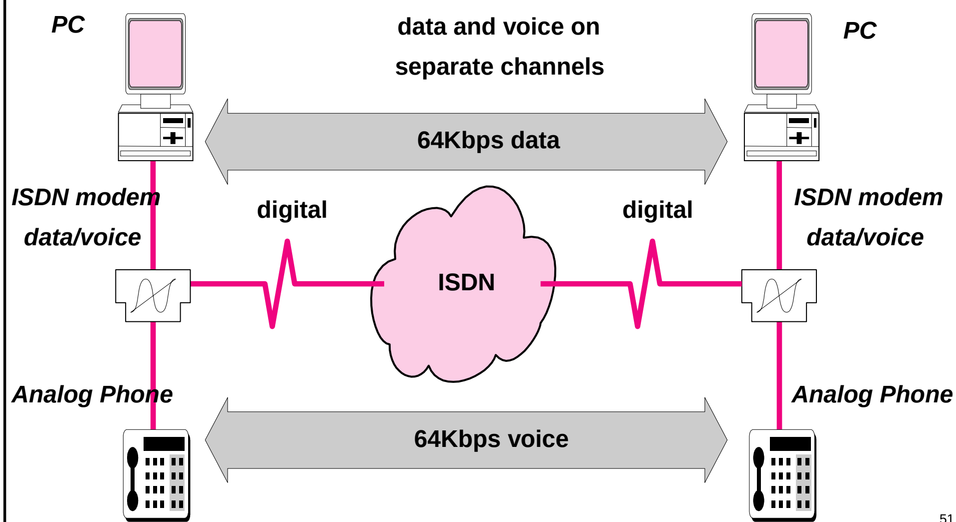
Voice/Data Multiplexers

- Organizations have traditionally chosen to link voice and data transmission over long distances via leased digital transmission services such as T-1/E-1.
- From a business perspective, switched services (frame relay, ATM) are charged according to usage and leased lines are charged according to flat monthly rate whether they are used or not.
- Many businesses found that usage based pricing can produce significant savings.
- A voice/data multiplexer simultaneously transmits digitized voice and data over a single digital transmission service.

Integrated Services Digital Network (ISDN)

- A newer switched digital service used for small business and residential users.
- ISDN BRI (Basic Rate Interface) service offers two 64Kbps channels.
- It offers two 64 Kbps channels, one for voice while the other for data. Both can be used simultaneously.

Simultaneous Voice/Data Transmission with ISDN



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Wireless Voice Transmission

- Modern wireless telephones are based on a cellular model.
- A wireless telephone system consists of a series of cells that surround a central base station, or tower.
- The term “cellular phone” or “cell phone” comes from the cellular nature of all wireless networks.

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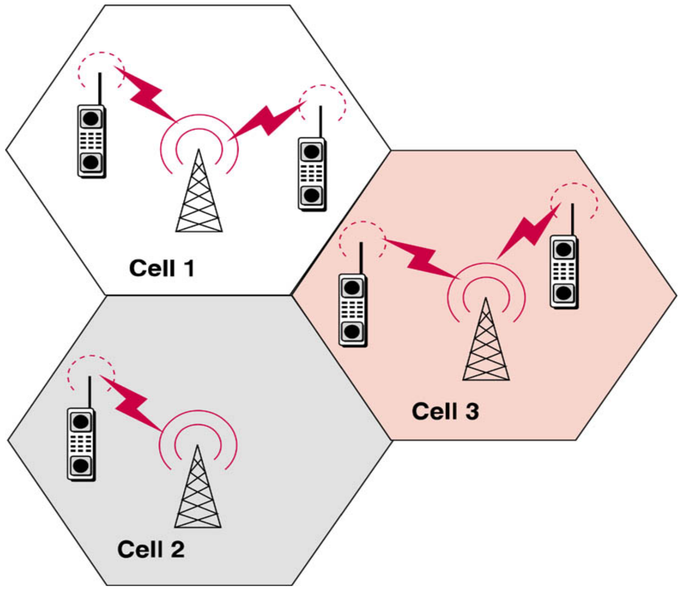
Wireless Voice Transmission



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Analog Cellular (1G)

- **Advanced Mobile Phone Service** (**AMPS**)
  - operate in the 800MHz frequency range.
  - carried just voice traffic.
  - have significant limitations.
  - offer relatively poor signal quality.
  - static and interference are inherent with the system.
  - can handle relatively few concurrent calls per cell.

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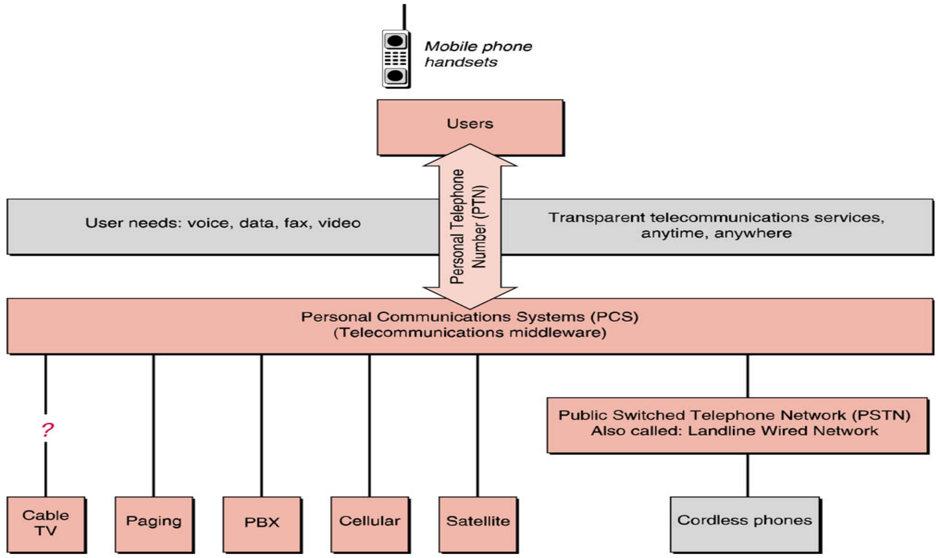
Wireless Voice Transmission



- Elements of digital cellular

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Digital Cellular (2G)

- carriers have steadily moved to digital cellular systems.
- the call is digitized at the telephone handset and sent in a digital format to the tower.
- quality is greatly improved.
- more calls to share the common bandwidth in a cell concurrently.
- better equipped to support wireless data transmission.

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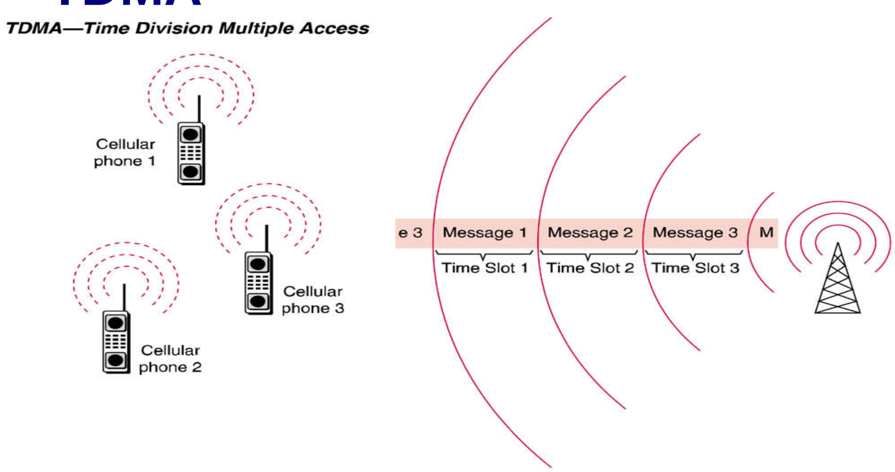
Digital Cellular Standards

- **TDMA** and **CDMA** are the two access methodologies used in digital cellular systems.
- Both offer significant capacity increases compared to AMPS analog cellular systems.

TDMA



- TDMA achieves more than one conversation per frequency by assigning timeslots to individual conversations.

Global System for Mobile Communication (GSM)

- A new service layer overlies TDMA.
- It provides a standardized billing interface (consumer can roam seamlessly between the GSM network of different companies), offers enhanced data services.
- In GSM, SIM card store the user's information, his phone number, contacts, and so on. So easy to change the phone set, no need of programming of new phone set.

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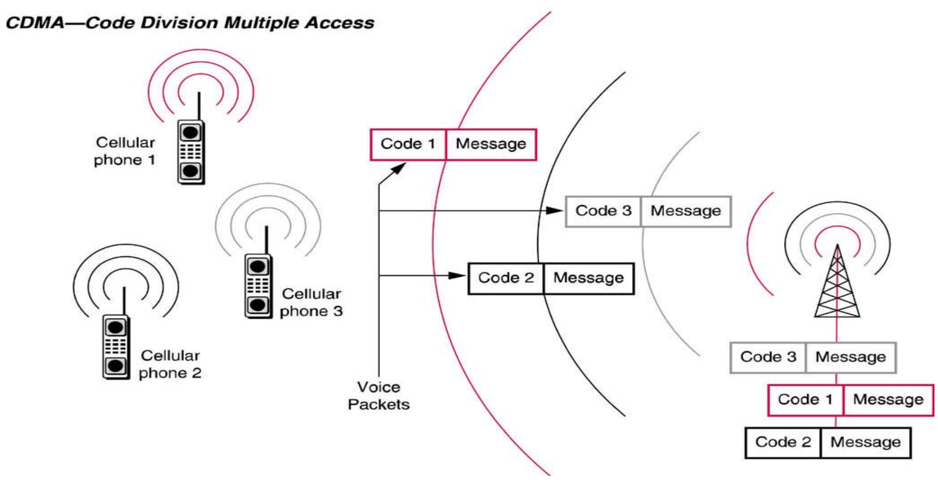
CDMA



- CDMA attempts to maximize the number of calls transmitted within a limited bandwidth by using a spread spectrum transmission technique.

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CDMA

- Spread spectrum transmission technique is like datagram connectionless service.
- In a CDMA system, encoded voice is digitized and divided into packets.
- These packets are tagged with “codes”.
- The packets then mix with all of the other packets of traffic in the local CDMA network as they are routed towards their destination.
- The receiving system only accepts the packets with the codes destined for it.

Different Generations

- AMPS → **1G** (1st Generation) max. 14.4Kbps
- TDMA & CDMA → **2G** (2nd Generation) 9.6-14.4Kbps
- GPRS (General Packet Radio Service) → **2.5G** (Advanced 2nd Generation) 56Kbps-115Kbps
- EDGE (Enhanced Data for GSM Evolution) & EV-DO (Evolution Data Only) → **3G** (3rd Generation) 128Kbps for moving car and 2Mbps for fixed.
- High-Speed Downlink Packet Access (HSDPA) is a → **3.5G,** 6 times faster than 3G. ***Currently available.***
- Commercially available in 2010 → **4G** (4th Generation) 100 Mbps

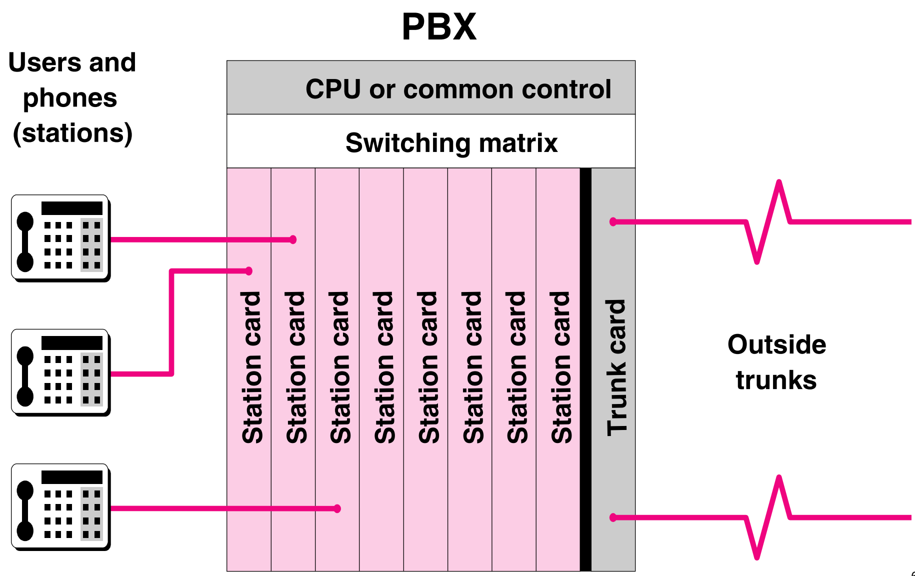
Private Branch Exchanges

- A PBX is just a privately owned, smaller version but similar in function to a public exchange.
- It is exclusively used by the organization and physically located on the organization's premises.
- Provides an interface between users and the shared network (PSTN).
- Additional services offered by a PBX allow users to use their phones more efficiently and effectively.
- Medium to large organizations can save a lot of money by using a PBX.

PBX Architecture

- PBX overall functionality and added features are controlled by software programs running on specialized computers within the PBX area sometimes referred to as the ***PBX CPU, stored program control***, or ***common control area.***
- User phones are connected to PBX via slide-in modules or cards known as ***line cards, port cards,*** or ***station cards***.
- Connection of PBX to outside world is accomplished via ***Trunk cards***.
- Starting with an open chassis or cabinet with power supply and backbone, cards can be added to increase PBX capacity either for the user extensions or outside connections.
- Additional cabinets can be cascaded for expandability.

PBX Physical Architecture



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PBX Technology Analysis

- PBX features and services tend to fall into three categories:
  1. provide users with flexible usage of PBX resources.
  2. provide for data/ voice integration.
  3. control and monitor the use of those PBX resources.

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1. Flexible Usage - Voice Based Features and Services

- Common features: Conference calling, Call forwarding /divert, Redialing, Call transfer, Speed dialing, Call hold, Hunting, etc.
- Least Cost Routing: Selecting lowest price long distance carriers.
- Automatic Call distribution: Incoming calls are routed directly to certain extensions without going through a central switchboard.
- Call pickup: Allows a user to pickup or answer another user's phone without forwarding.
- Paging: Ability to use paging speakers in a building.

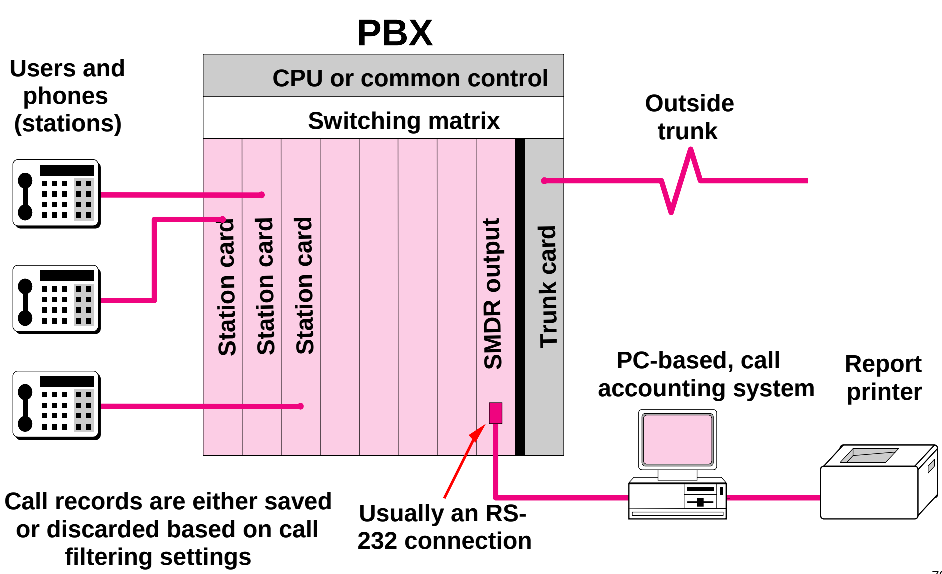
2. Data/Voice Integration – Features and Services

- Data is transmitted either:
  - through the PBX via a dedicated connection OR
  - a hybrid voice/data phone is used to transmit both voice and data simultaneously over a single connection.
- Features:
  - ISDN (Integrated Services Digital Network) support, T-1 / E-1 interfaces support (codecs included or not), Data interfaces, modem pooling, printer sharing, file sharing, video conferencing, etc.

3. Control and Monitoring – Features and Services

- Basic: (e.g.)
  - Limiting access to outside lines from certain extensions.
- Advanced:
  - Call accounting system: program run on a separate PC directly connected to the PBX.
  - Process within the PBX known as Station Message Detail Recording (**SMDR**) where an individual detail record is generated for each call.
  - Used for spotting abuse, both incoming and outgoing calls can be tracked.
  - Allocating phone usage on a departmental basis.

Call Accounting Systems Installation



Auxiliary Voice Related Services

- Auxiliary add-on device that provides the following services:
  - Automated attendant
  - Voice mail
  - Voice response units (VRU), e.g., Interactive voice response (IVR).
  - Voice processor: e.g. speech recognition
  - Voice server: a LAN based server that stores, and delivers digitized voice messages. Used with voice mail system.
  - Music / ads on hold

Computer Telephony Integration (CTI)

- CTI seeks to integrate the computer and the telephone to enable increased productivity not otherwise possible by using the two devices in a non-integrated fashion.
- CTI is not a single application, but an ever-widening array of possibilities spawned by the integration of telephony and computing.

Computer Telephony Integration (CTI)

- CTI attempts to integrate the two most common productivity devices, the phone and the computer to increase productivity.
- Examples of the integration:
  - *Call control:* allows users to control their telephone functions through their computer, on-line phone books, on-line display and processing of voice mail.
  - *Interactive Voice Response:* E.g., IVR systems used by banks, carriers, etc.
  - *Unified massages:* Voice mail, e-mail, faxes, pager messages to be displayed on a single graphical screen. Then can be forwarded, replied, deleted, etc.

CTI Architectures

- CTI is commonly implemented in one of the following three architectures:
  - PBX-to-host interfaces (Integration of PBX with mainframe, minicomputers, etc. for call center and office automation applications)
  - Desktop CTI
  - Client/server CTI

C T I Architecture

1 - PBX to host interfaces

- Before the arrival of open systems Computer Telephony Integration APIs such as TAPI (Telephony Application Program Interface), TSAPI, each PBX vendor had its own PBX-to-host interface specifications.
- In PBX-to-host interface CTI was achieved by linking mainframes to PBXs via PBX-to-host-interface.
- Compatible applications with computer and PBX.
- Systems linked to an automatic call distribution unit (ACD)
- All phones are controlled by CTI application running on mainframe computer.
- Expensive systems.

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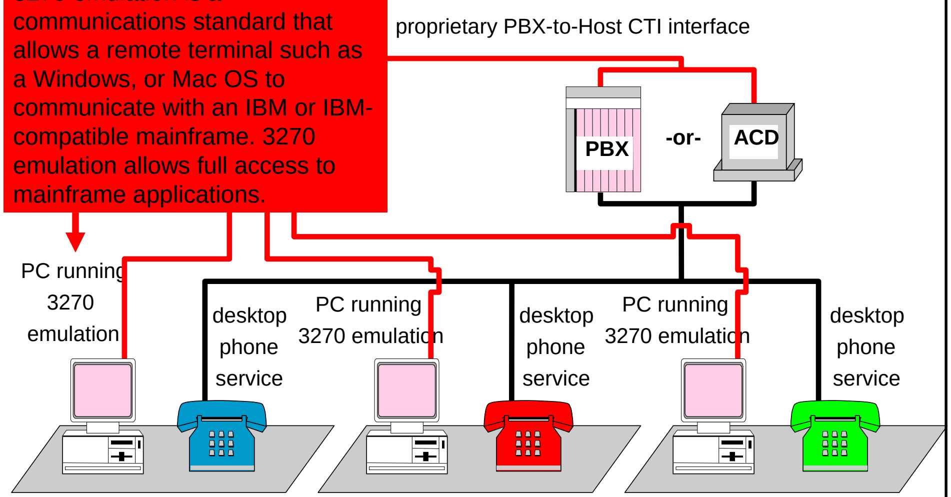
C T I Architecture

1 - PBX to host interfaces

3270 emulation is a communications standard that allows a remote terminal such as a Windows, or Mac OS to communicate with an IBM or IBM-compatible mainframe. 3270 emulation allows full access to mainframe applications.



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C T I Architecture

2 - Desktop CTI

- Also known as, first party call control
- Less expensive alternative to PBX-to-host architecture.
- PC's are equipped with telephony boards and associated call control software.
- Each PC controls only the telephone to which it is attached.
- No overall automatic call distribution across multiple agents and their phones.
- No sharing of call related data among the desktop CTI PC's.

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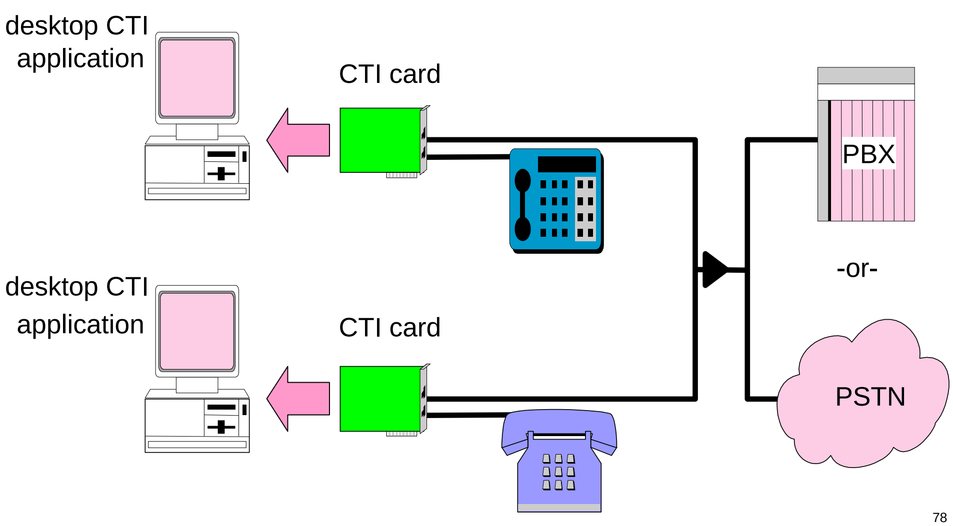
C T I Architecture

2 - Desktop CTI



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3 - Client/Server C T I

- CTI server computer interfaces to the PBX or ACD to provide overall system management.
- Individual client based CTI applications execute on multiple client PCs.
- Multiple CTI applications on multiple client PCs can share the information supplied by the single CTI Server.
- Offers overall shared control of the PBX-to-host CTI architecture at a cost closer to that of the desktop architecture.

C T I Architecture

3 - Client/Server C T I