



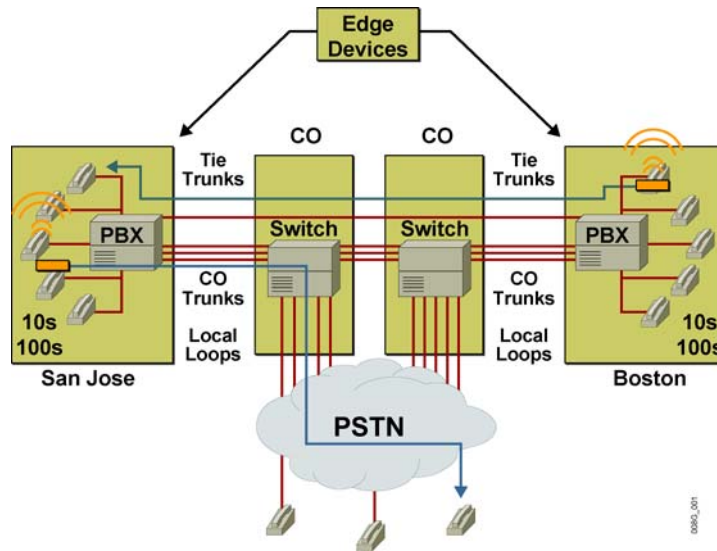
Introduction to Packet Voice Technologies and VoIP

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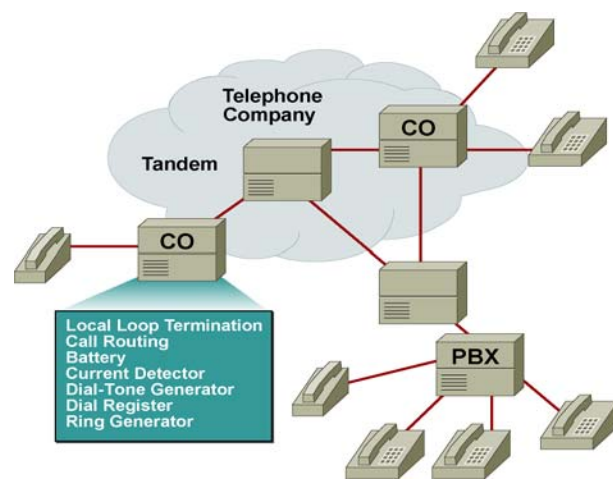
Traditional Telephony



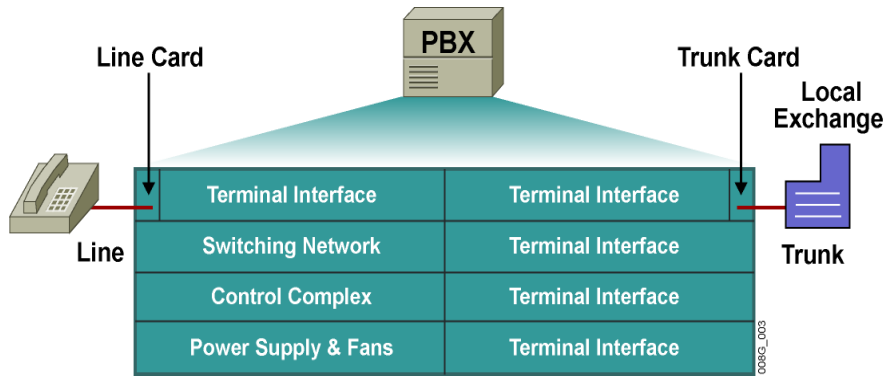
Basic Components of a Telephony Network



Central Office Switches

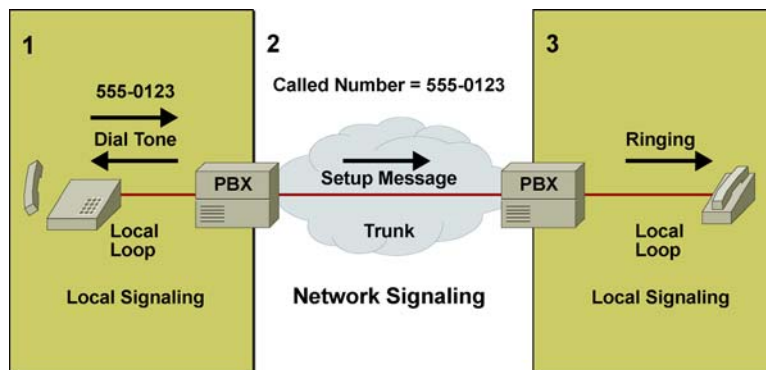


What Is a PBX? (Private Branch Exchanges)



The equivalent to a switch , the PBX, supports from from five to several thousand local loops.

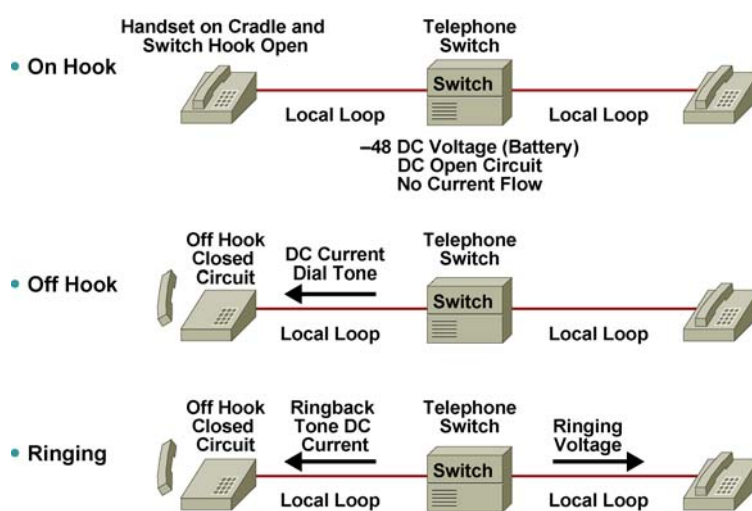
Basic Call Setup



Types of Local-Loop Signaling

- Supervisory signaling
- Address signaling
- Informational Signaling

Supervisory Signaling



Address Signaling



Tone telephone
DTMF dialing

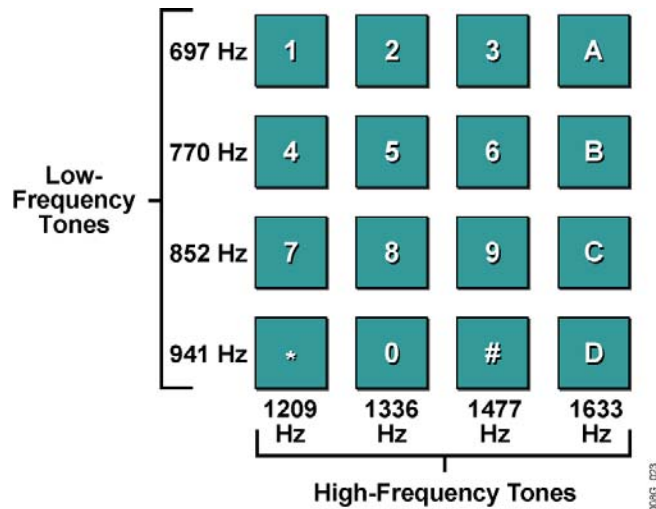


- **Rotary telephone**
 - Pulse dialing

DTMF address signaling

- **Dual tone multifrequency (DTMF)**
- **With DTMF , two simultaneous frequencies are generated, and a phone switch interprets this combination of frequencies as a dialed digit.**
- **Dual tones are used instead of just a single tone to reduce affect of background noise.**

Dual Tone Multifrequency



DTMF_023

Informational Signaling

Tone	Frequency (Hz)	On Time (Sec)	Off Time (Sec)
Dial	350 + 440	Continuous	Continuous
Busy	480 + 620	0.5	0.5
Ringback, line	440 + 480	2	4
Ringback, PBX	440 + 480	1	3
Congestion (toll)	480 + 620	0.2	0.3
Reorder (local)	480 + 620	0.3	0.2
Receiver off hook	(1400 + 2060 + 2450 + 2600)	0.1	0.1
No such number	200 to 400	Continuous	Continuous
Confirmation tone		Freq. Mod. 1 kHz	Freq. Mod. 1 kHz

DTMF_024

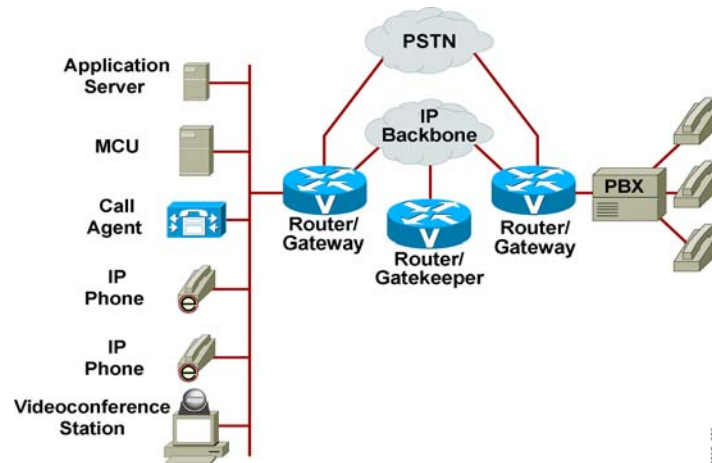
Packetized Telephony Networks



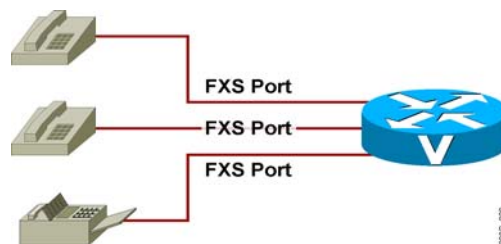
Packet Telephony vs. Circuit-Switched Telephony

- **More efficient use of bandwidth and equipment**
- **Lower transmission costs**
- **Consolidated network expenses**
- **Increased revenue from new services**
- **Service innovation**
- **Access to new communications devices**
- **Flexible new pricing structures**

Packet Telephony Components



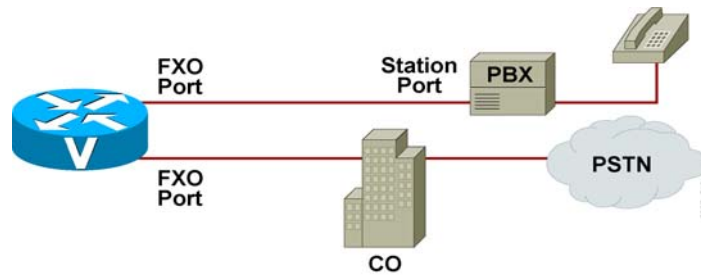
Foreign Exchange Station Interface



Foreign Exchange Station

- analog connection on a gateway that connects to a station, such as a analog phone , fax machine or speaker phone
- connects directly to station equipment
 - Used to provision local service

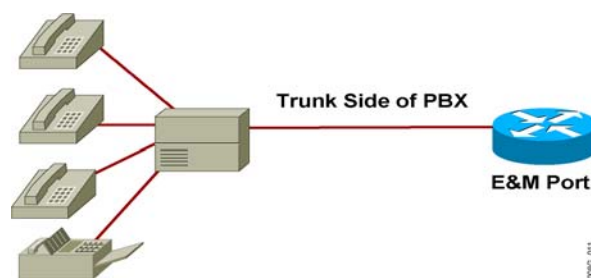
Foreign Exchange Office Interface



Foreign Exchange Office

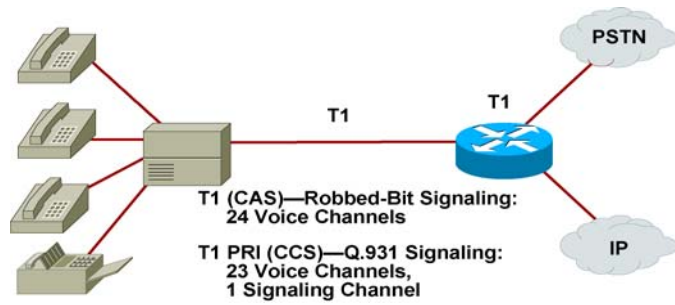
- analog port on a gateway that to an office (that is phone switch, such as a PBX)
- connects directly to office equipment
- used to extend connections to another location

E&M Interface



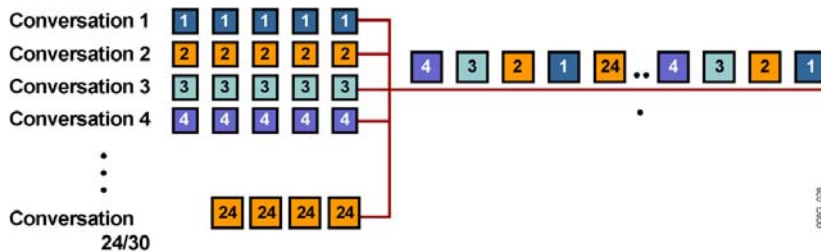
E&M (Ear and Mouth) is an analog interface present in many of today's PBX

T1 Interface



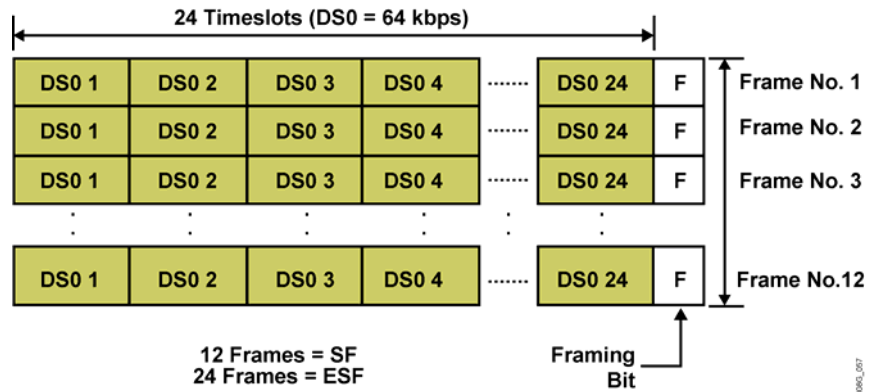
T1 is a 1.544-Mbps digital transmission link normally used in North America

Time-Division Multiplexing (using with T1)



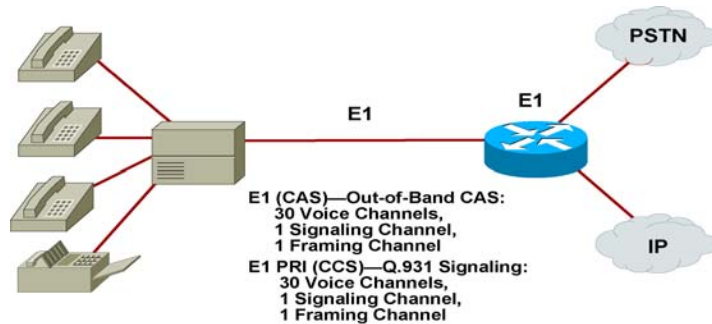
Time-division multiplexing is a process of sending multiple conversations on a single connection by giving different time slices to different conversations.

T1 Digital Signal Format



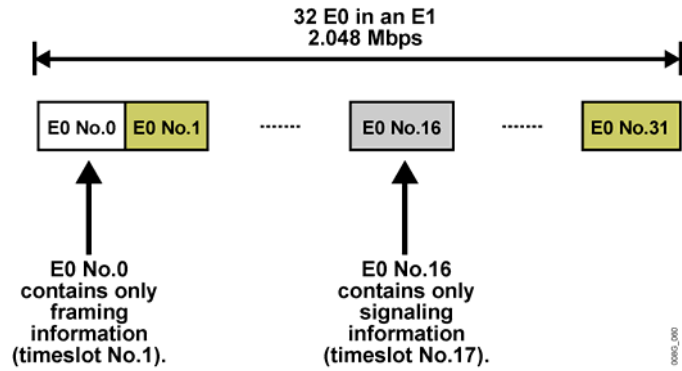
T1 link has 24 full duplex channels or digital level 0 (DS0 1-DS0 24). The 24 channel, 8 bit values are multiplexed into a serial bit stream using TDM to generate a 192-bit frame.

E1 Interface



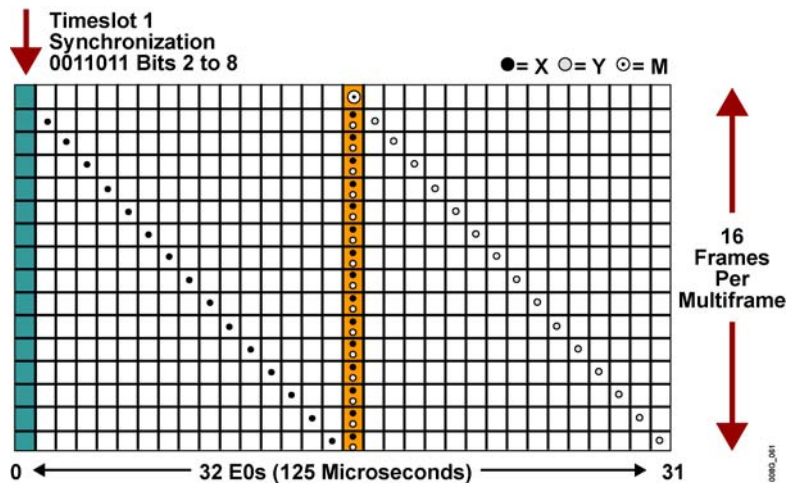
E1 is a 2.048-Mbps digital transmission link normally used in Europe

E1 Framing and Signaling



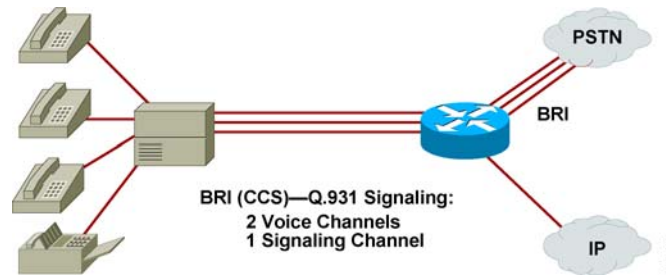
E1 link has 32 full duplex channels.
E1 combines 16 frame together in a multiframe.

Channel Associated Signaling—E1



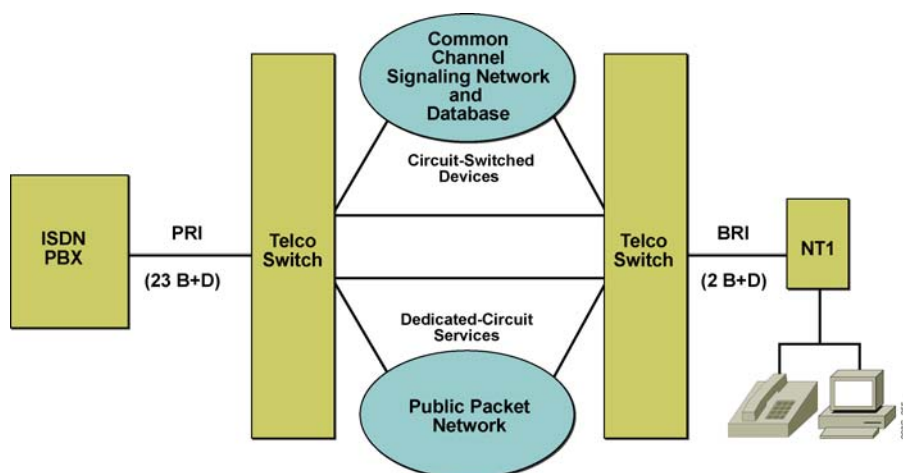
E1 made up 30 DS-0 for voice and data plus one channel for framing and one for signaling..

ISDN – Integrated Services Digital network BRI

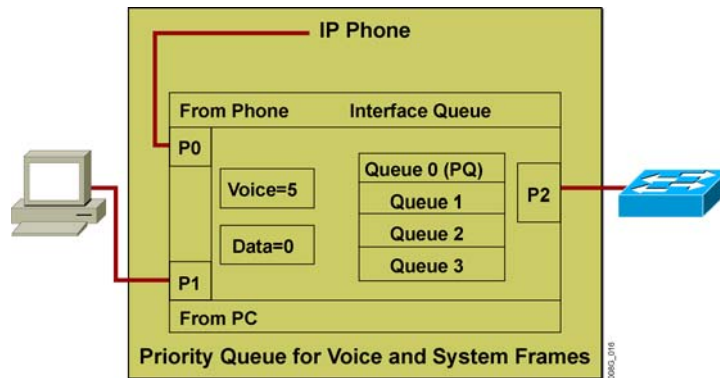


BRI – Basic rate interface service is the entry level and offers two 64 kbps voice channels and one 16 kbps signaling channel

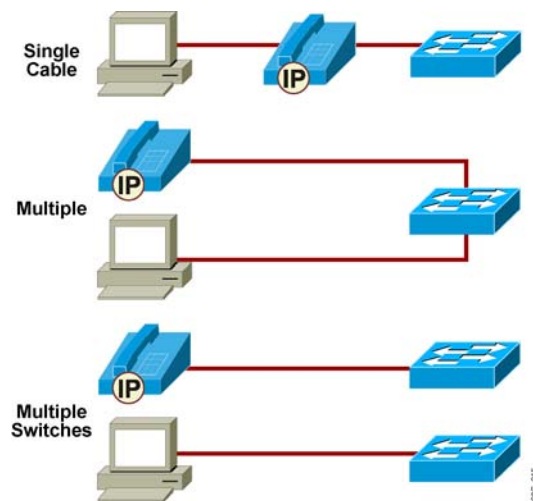
ISDN Network Architecture



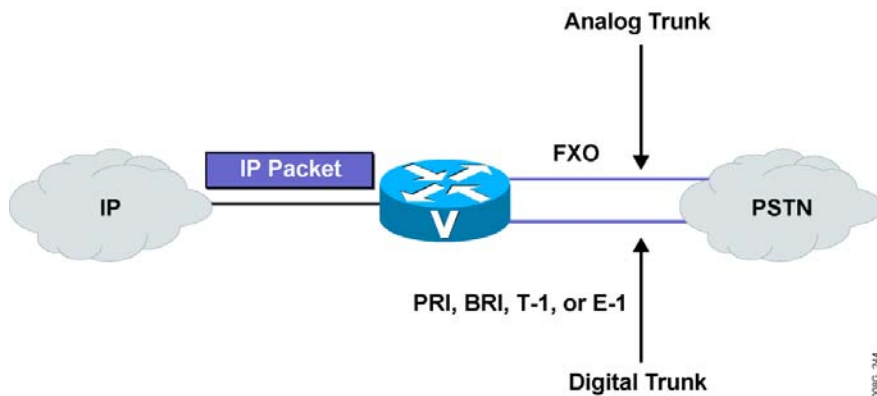
IP Phone



Physical Connectivity Options



Gateways and Their Roles Analog vs. Digital



Gathering the Requirements

- Is an analog or digital gateway required?
- What is the required capacity of the gateway?
- What type of connection is the gateway going to use? Is Foreign Exchange Office (FXO), FXS, E&M, T1, E1, PRI, or BRI signaling required?
- What signaling protocol is used? H.323, Media Gateway Control Protocol (MGCP), or session initiation protocol (SIP)?
- Is voice compression a part of the design? If so, which type?
- Are direct inward dialing (DID), calling line identification (CLID), modem relay, or fax relay required?
- Is the device acting only as gateway or as gateway and router/LAN switch? Is inline power for IP Phones required?
- Is remote site survivability required?
- To which country is the hardware shipped?

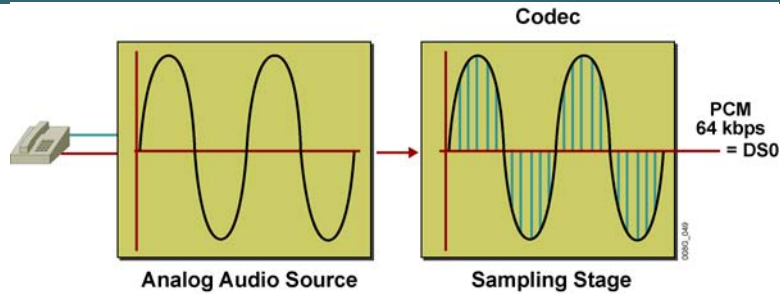
Analog-to-Digital Voice Encoding



Digital & Analog

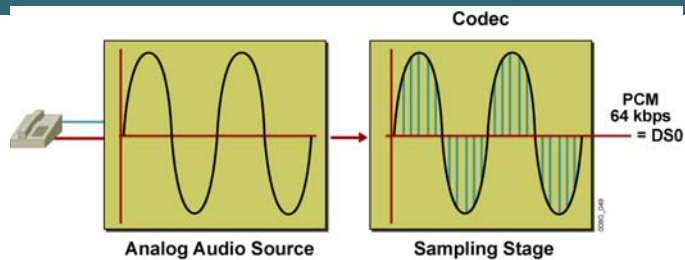
- **Analog communication is a mix of time and amplitude. Our voices are analog, meaning a continuously varying waveform.**
- **VoIP networks transmit our voices digitally using binary encoding, meaning a series of 1s and 0s**
- **PCM (pulse code modulation) is the most common method of encoding an analog voice signal into a digital stream of 1s and 0s**

Nyquist Theorem



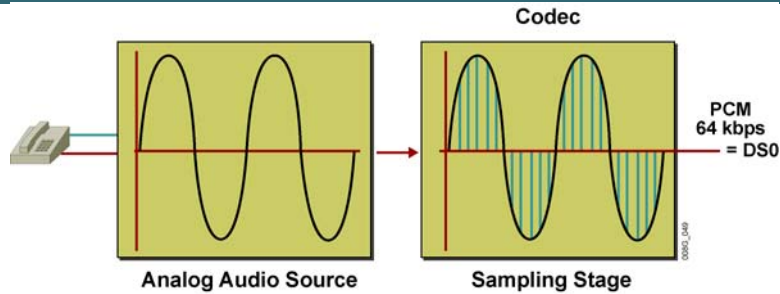
The sample rate needs to be at least twice as high as the highest frequency being sampled

Nyquist Theorem



**Highest frequency for voice 4 kHz
8000 (4000*2) samples per second
Each sample 8 Bits
8000*8=64 000 bits per second**

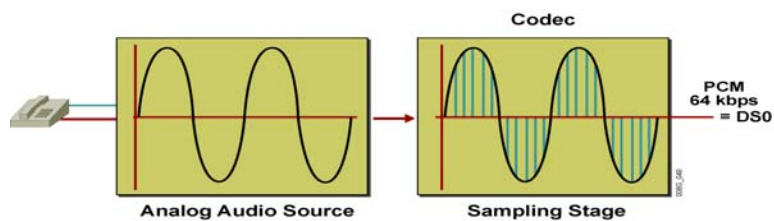
Nyquist Theorem



Once analog waveforms have been digitized, we might want to save WAN bandwidth by compressing these digitized waveforms by encoding them. The process of encoding and decoding these waveforms are defined by coder decoders (CODEC)

Digitizing Analog Signals

1. Sample the analog signal regularly.
2. Quantize the sample.
3. Encode the value into a binary expression.
4. Compress the samples to reduce bandwidth, optional step.



Basic Voice Encoding: Converting Digital to Analog

1. Decompress the samples, if compressed.
2. Decode the samples into voltage amplitudes, rebuilding the PAM signal.
3. Filter the signal to remove any noise.

Voice Compression Techniques

Waveform algorithms

PCM-pulse code modulation. Doesn't actually compress the analog waveform. G711 is the CODEC that uses PCM.

Waveform coding scheme

ADPCM- Adaptive differentiated PCM. Uses a difference signal, instead of encoding an entire sample can send the difference in the current sample. G726 is the CODEC that uses ADPCM.

Waveform coding scheme

Adaptive: automatic companding

Differential: encode changes between samples only

Voice Compression Techniques

Source algorithms

CS-ACELP- Conjugate Algebraic Code Excited Linear Predication
– Dynamically builds a codebook based on speech pattern. It then uses a look ahead buffer to see whether the next sample matches a pattern already in the codebook. If it does, then the codebook location can be send instead of the actual sample.

Example. We have conversation across a digital circuit. I make the sound “ing”(word routing, reading). Instead of digitizing the “ing” sound, you make an entry in a codebook that describes what “ing” sounds like. The advantage is that instead of sending the actual sound, I’m only sending you location of that sound in your codebook, which takes up far less bandwidth than sending the actual sound.

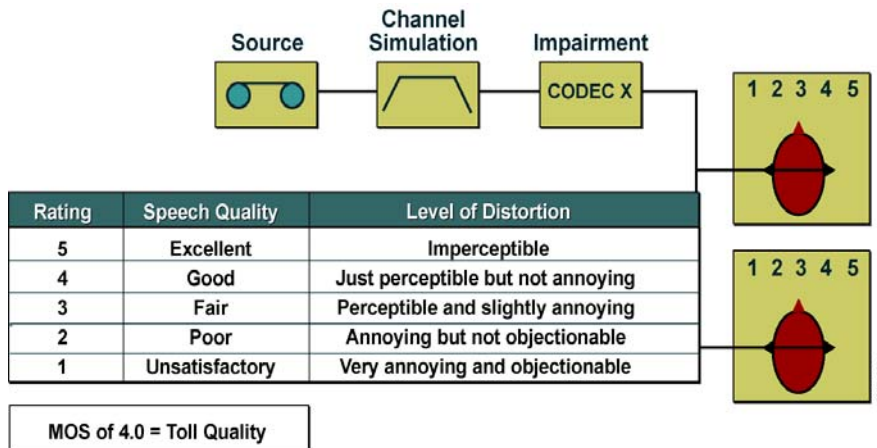
LDCELP- Low -Delay Conjugate Excited Linear Predication is very similar to CS-ACELP , but uses smaller codebook, resulting in less delay, but it requires more bandwidth.

Compression Bandwidth Requirements

Standard	Bit Rate (kbps)
G.711, PCM	64
G.726, ADPCM	16, 24, 32
G.728, LDCELP	16
G.729, CS-ACELP	8
G.729A, CS-ACELP	8

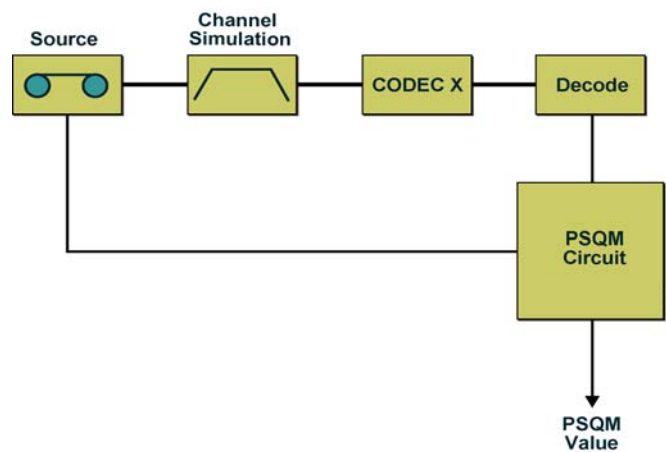
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Mean Opinion Score MOS



Uses a trained ear to judge the quality of voice after passing through the CODEC being tested. Range from 1 to 5.

Perceptual Speech Quality Measurement (PSQM)

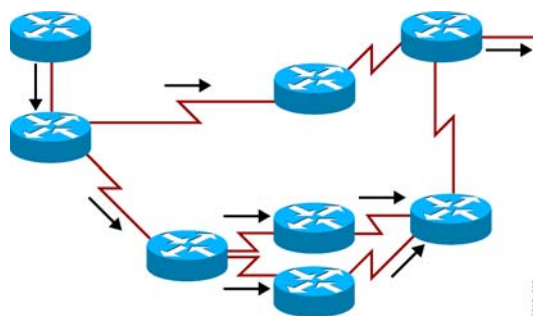


Digitally measures the difference in the original signal and the signal after it passed through a CODEC.

Part2 Requirements of Voice in an IP Internetwork



IP Internetwork

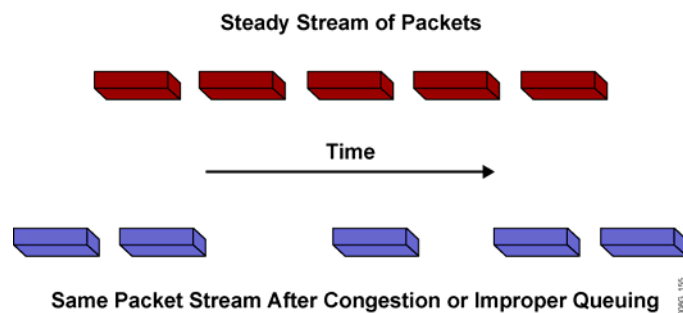


- IP is connectionless.
- IP provides multiple paths from source to destination.

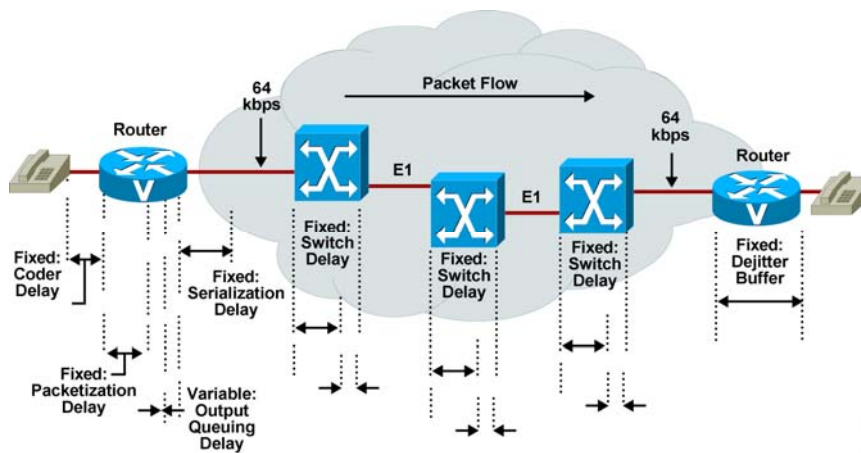
Packet Loss, Delay, and Jitter

- **Packet loss**
Loss of packets severely degrades the voice application.
- **Delay**
VoIP typically tolerates delays up to 150 ms before the quality of the call degrades.
- **Jitter is the variation of packet interarrival time**
Instantaneous buffer use causes delay variation in the same voice stream.

Jitter in IP Networks



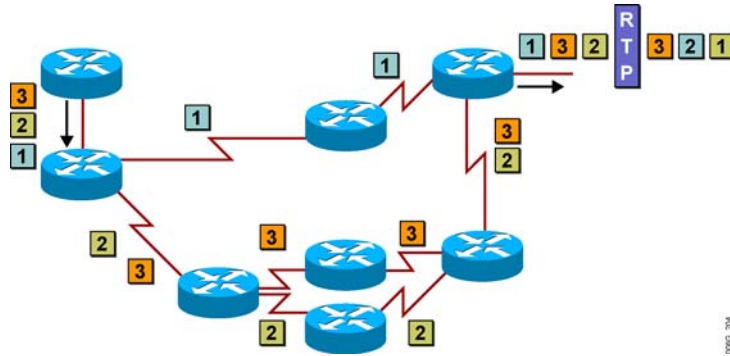
Sources of Delay



Consistent Throughput

- Throughput is the amount of data transmitted between two nodes in a given period.
- Throughput is a function of bandwidth, error performance, congestion, and other factors.
- Tools for enhanced voice throughput include:
 - Queuing
 - Congestion avoidance
 - Header compression
 - RSVP- Resource reservation protocol
 - Fragmentation

Reordering of Packets



- IP assumes packet-ordering problems.
- RTP reorders packets.

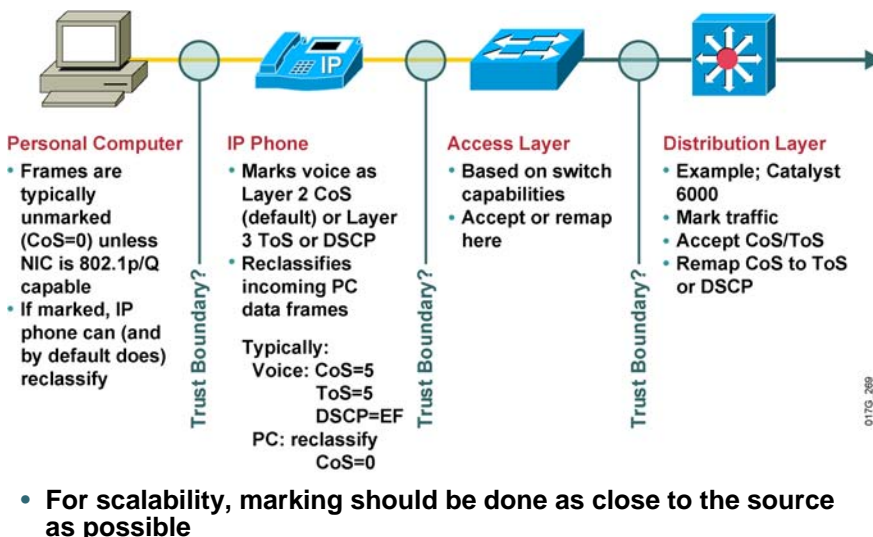
Reliability and Availability

- Traditional telephony networks claim 99.999% uptime.
- Data networks must consider reliability and availability requirements when incorporating voice.
- Methods to improve reliability and availability include:
 - Redundant hardware
 - Redundant links
 - UPS
 - Proactive network management

QoS Mechanisms

- **Classification:** Each class-oriented QoS mechanism has to support some type of classification
- **Marking:** Used to mark packets based on classification and/or metering
- **Congestion Management:** Each interface must have a queuing mechanism to prioritize transmission of packets
- **Traffic Shaping:** Used to enforce a rate limit based on the metering by delaying excess traffic
- **Compression:** Reduces serialization delay and bandwidth required to transmit data by reducing the size of packet headers or payloads
- **Link Efficiency:** Used to improve bandwidth efficiency through compression and link fragmentation and interleaving

Trust Boundaries Mark Where?



Encapsulating Voice in IP Packets



Major VoIP Protocols

VoIP Protocol	Description
H.323	ITU standard protocol for interactive conferencing. Evolved from H.320 ISDN standard. Flexible, complex.
MGCP	Emerging Internet Engineering Task Force (IETF) standard for PSTN gateway control, thin device control.
SIP	IETF protocol for interactive and noninteractive conferencing. Simpler, but less mature, than H.323.
RTP	IETF standard media streaming protocol.
RTCP	IETF protocol that provides out-of-band control information for an RTP flow.

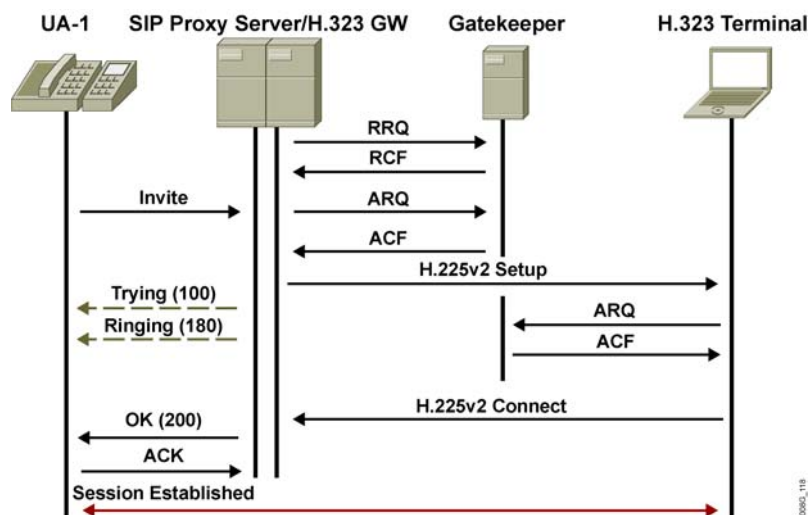
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VoIP Protocols and the OSI Model

Application	Softphone/CallManager/Human Speech
Presentation	Codecs
Session	H.323/SIP/MGCP
Transport	RTP/UDP (media); TCP/UDP (signal)
Network	IP
Data Link	Frame Relay (FR), ATM, Ethernet, Multilink Point-to-Point Protocol (MLPPP), Point-to-Point Protocol (PPP), High-Level Data Link Control (HDLC)...
Physical	...

Constant—Voice media packets use RTP/UDP
Variable—Several signaling methods and link layer protocols

Translation Between Signaling and Call Control



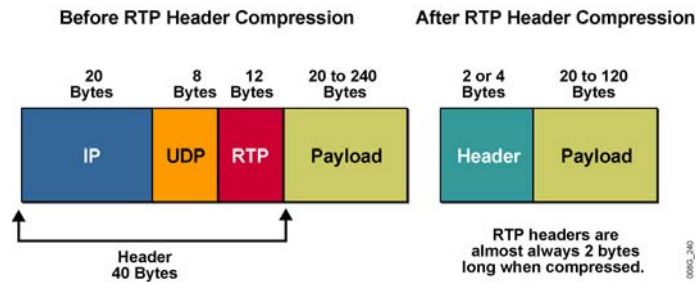
Real-Time Transport Protocol

- Provides end-to-end network functions and delivery services for delay-sensitive, real-time data, such as voice and video
- Works with queuing to prioritize voice traffic over other traffic
- Services include:
 - Payload-type identification
 - Sequence numbering
 - Time stamping
 - Delivery monitoring

Real-Time Transport Control Protocol

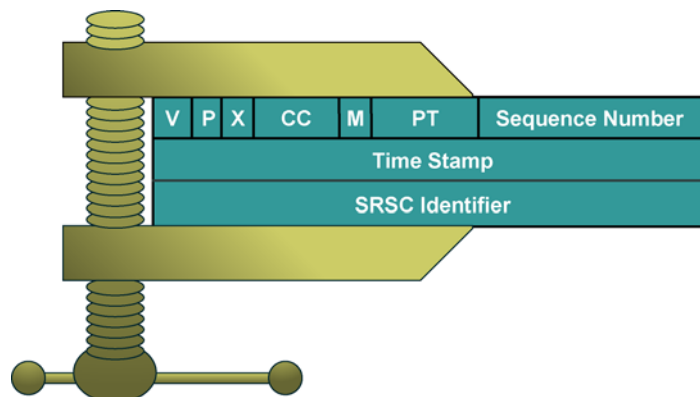
- Monitors the quality of the data distribution and provides control information
- Provides feedback on current network conditions
- Allows hosts involved in an RTP session to exchange information about monitoring and controlling the session
- Provides a separate flow from RTP for UDP transport use

RTP Header Compression



- RTP header compression saves bandwidth by compressing packet headers across WAN links.

When to Use RTP Header Compression



- Narrowband links (wireless communication)
- Slow links (less than 2 Mbps)
- Need to conserve bandwidth on a WAN interface

Calculating Bandwidth Requirements



Bandwidth Implications of Codec

Codec	G.711	G.726 r32	G.726 r24	G.726 r16	G.728	G.729	G.723 r63	G.723 r53
Bandwidth	64 kbps	32 kbps	24 kbps	16 kbps	16 kbps	8 kbps	6.3 kbps	5.3 kbps

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Data Link Overhead

- **Ethernet**
18 bytes overhead
- **MLP**
6 bytes overhead
- **Frame Relay**
6 bytes overhead

Security and Tunneling Overhead

- **IPSec**
50 to 57 bytes
- **L2TP/GRE**
24 bytes
- **MLPPP**
6 bytes
- **MPLS**
4 bytes

Total Bandwidth Required

Codec	Codec Speed	Sample Size	Frame Relay	Frame Relay with CRTP	Ethernet	Ethernet with CRTP
G.711	64000	240	76267	66133	78933	68800
G.711	64000	160	82400	67200	86400	71200
G.726r32	32000	120	44267	34133	46933	36800
G.726r32	32000	80	50400	35200	54400	39200
G.726r24	24000	80	37800	26400	40800	29400
G.726r24	24000	60	42400	27200	46400	31200
G.726r16	16000	80	25200	17600	27200	19600
G.726r16	16000	40	34400	19200	38400	23200
G.728	16000	80	25200	17600	27200	19600
G.728	16000	40	34400	19200	38400	23200
G.729	8000	40	17200	9600	19200	11600
G.729	8000	20	26400	11200	30400	15200
G.723r63	6300	48	12338	7350	13650	8663
G.723r63	6300	24	18375	8400	21000	11025
G.723r53	5300	40	11395	6360	12720	7685
G.723r53	5300	20	17490	7420	20140	10070

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Effect of VAD- *voice activity detection*

- In normal voice conversation, someone speaks and someone else listens.
- Networks contain bi-directional channels. This means that bandwidth is wasted.
- We can utilize this wasted bandwidth with voice activity detection.
- VAD works by detecting the magnitude of speech in decibels and deciding when to cut off the voice from being framed.

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Effect of VAD(voice activity detection)

Codec	Codec Speed	Sample Size	Frame Relay	Frame Relay with VAD
G.711	64000	240	76267	49573
G.711	64000	160	82400	53560
G.726r32	32000	120	44267	28773
G.726r32	32000	80	50400	32760
G.726r24	24000	80	37800	24570
G.726r24	24000	60	42400	27560
G.726r16	16000	80	25200	16380
G.726r16	16000	40	34400	22360
G.728	16000	80	25200	16380
G.728	16000	40	34400	22360
G.729	8000	40	17200	11180
G.729	8000	20	26400	17160
G.723r63	6300	48	12338	8019
G.723r63	6300	24	18375	11944
G.723r53	5300	40	11395	7407
G.723r53	5300	20	17490	11369

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