Voice over IP

Abdus Salam ICTP, February 2004

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VoIP

Takes advantage of data networks for transmitting voice in packets, providing potential BW saving and integration between the telecommunication and Data communications world.

It requires telephony, networking and traffic engineering skills

The basic functions to be performed by a VoIP system are:

1) Voice digitizing
2) Voice Packetizing
3) Packet Routing

Additional Functions

- Conversion from telephone numbers to IP addresses and vice versa
- Generation of telephone system signaling
- Admission Control and invoicing
- Fax handling

Advantages of VoIP

- Bandwidth saving by using more efficient codification techniques and making a more efficient use of the channel
- Using the same network for data and voice allows for considerable saving in maintenance
- Easy integration with web based services

VoIP Limitations

- Ip networks latency can be quite large and variable
- Packet retransmission in the event of errors is useless for voice
- Overhead added to each packet can negate any BW savings
- Quality of voice can suffer when conversions are required

Requirements for VoIP

- Use protocols that allow for QoS to give priority of Voice over Data
- Control number of hops and all delays to keep latency below 170 ms

VoIP Trends

- MCI is planning to migrate all its network to IP by 2005
- AT&T will offer VoIP services to big customers in 2004.
- Vonage from N.J. offers flat rate over U.S and Canada for $35/month
- Skypes offers free good quality phone calls over the Internet using peer to peer

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skype™

- Free unlimited worldwide phone calls to other Skype users
- Superior sound quality - better than your regular phone
- Works with all firewall, NAT and routers – nothing to configure!
- Friends list shows you when your Skype friends are online and ready to talk or chat
- Super-simple and easy to use
- Your calls are encrypted “end-to-end” for superior privacy
- Based on cutting edge peer-to-peer technology developed by the creators of Kazaa and Joltid

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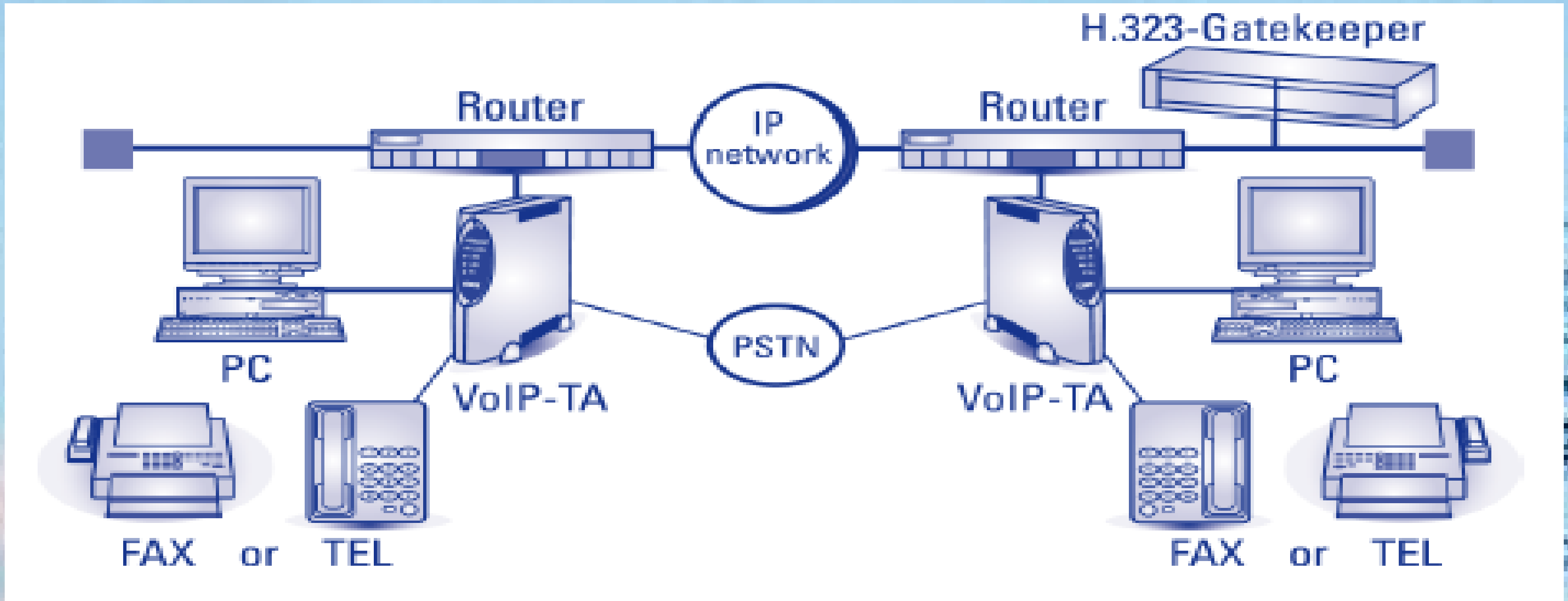
Skype vs. All The Rest

	skype	Net2Phone	MSN Messenger, ICQ, AIM, Yahoo Messenger	Other standard VoIP clients
Works with ANY firewall/NAT setup – nothing to configure	✓	⊘	⊘	⊘
Unlimited FREE calls to users of same application	✓	⊘	✓	Sometimes
Sound quality	★★★ Better than phones	★ Worse than phones	★ Worse than phones	★ Worse than phones
Secure and encrypted communications	✓	⊘	⊘	⊘
100% ad-free	✓	⊘	⊘	Sometimes

VoIP using normal phones and HW Gateways



VoIP Protocols

- H.323
- SIP
- MEGACO
- S/MGCP

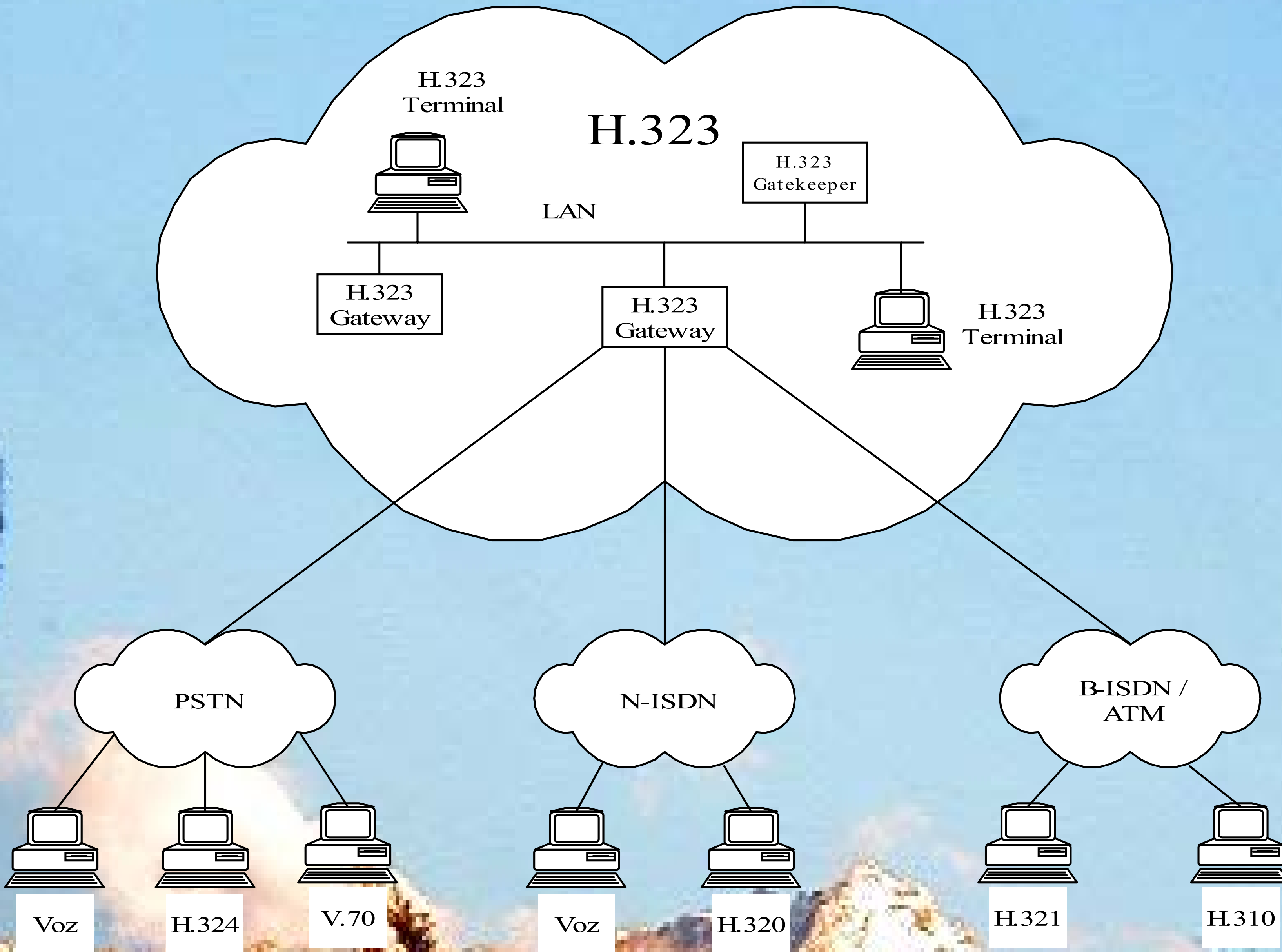
ITU-T H.323 Recommendation

Multimedia over networks that do not provide QoS like:

- Ethernet (IEEE 802.3)
- Fast Ethernet
- FDDI
- Token Ring (IEEE 802.5)

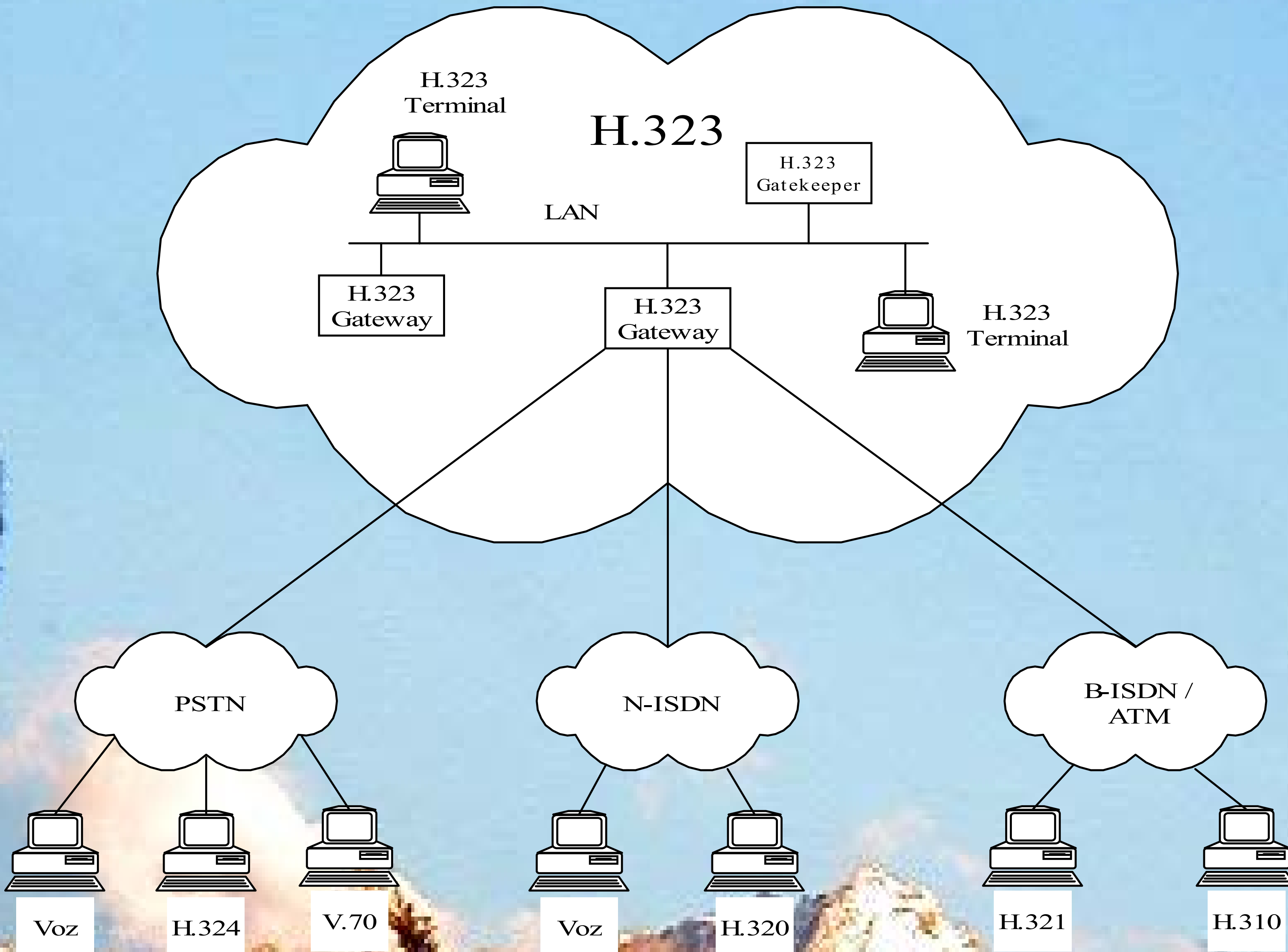
Interoperability

H.323 terminals can be used in multiple configurations and can communicate with completely different networks, like B-ISDN,Wireless networks, Cable TV based and so on.


H.323
H.323 Terminal
H.323 Gatekeeper
LAN
H.323 Gateway
H.323 Gateway
H.323 Terminal
PSTN
N-ISDN
B-ISDN / ATM
Voz
H.324
V.70
Voz
H.320
H.321
H.310

Components of H.323

- **Terminal: endpoint that can accomplish communication with another endpoint, GW or MCU.**
- **Gateway: enpoint that provides communication between LAN terminals and other ITU WAN terminals, including** H.320 (ISDN), H.321, ATM, .Asynchronous Transfer Mode), H.322 (GQOS, Garanteed Quality of Service), H.324 M (mobile).
- **Gatekeeper: entity that provides address translation and acces control for terminals, GW and MCU, when needed.**
- **Multipoint Control Unit (MCU):** end terminal which handles the coordination function ina multicast conference..


H.323
H.323
Terminal
H.323
Gatekeeper
LAN
H.323
Gateway
H.323
Gateway
H.323
Terminal
PSTN
N-ISDN
B-ISDN /
ATM
Voz
H.324
V.70
Voz
H.320
H.321
H.310

H.323 makes use of:

RTP (Real Time Protocol,) which adds sequence number and timing information that evidences loss of packets.

RTCP (RTP Control Protocol) keeps track of the quality of transmission

Sources of delay

- Compression Delay.
- Packetizing Delay
- Queing Delay at CPE.
- Serialization Delay in the WAN CPE
- WAN.transmission Delay
- Jitter buffer delay.
- Decompression Delay

Latency and Latency Jitter

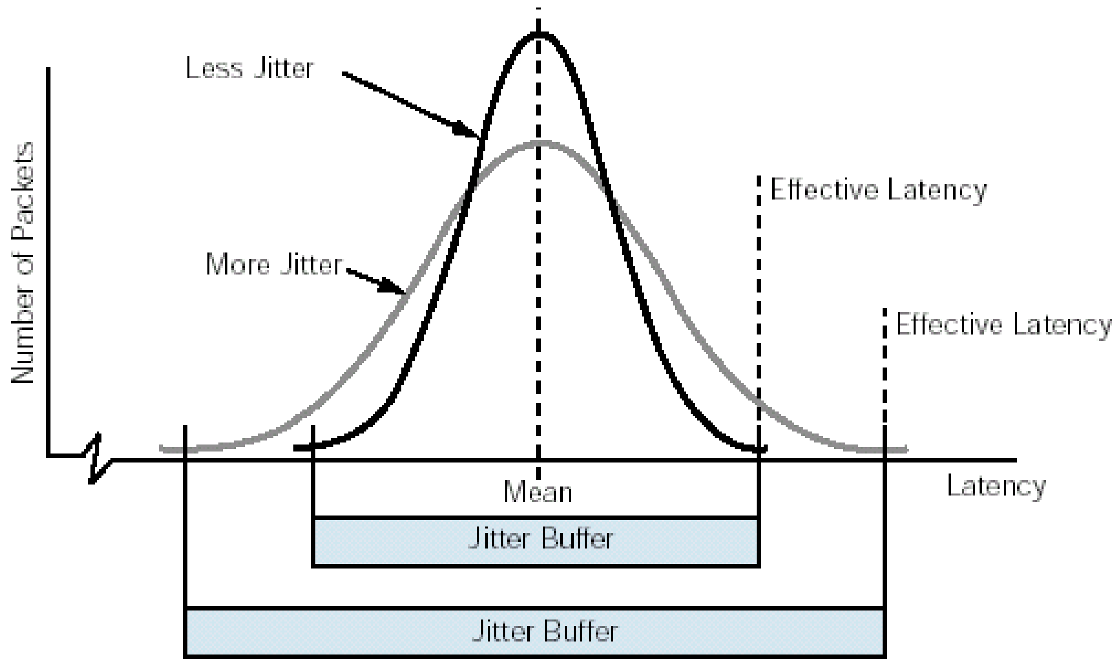
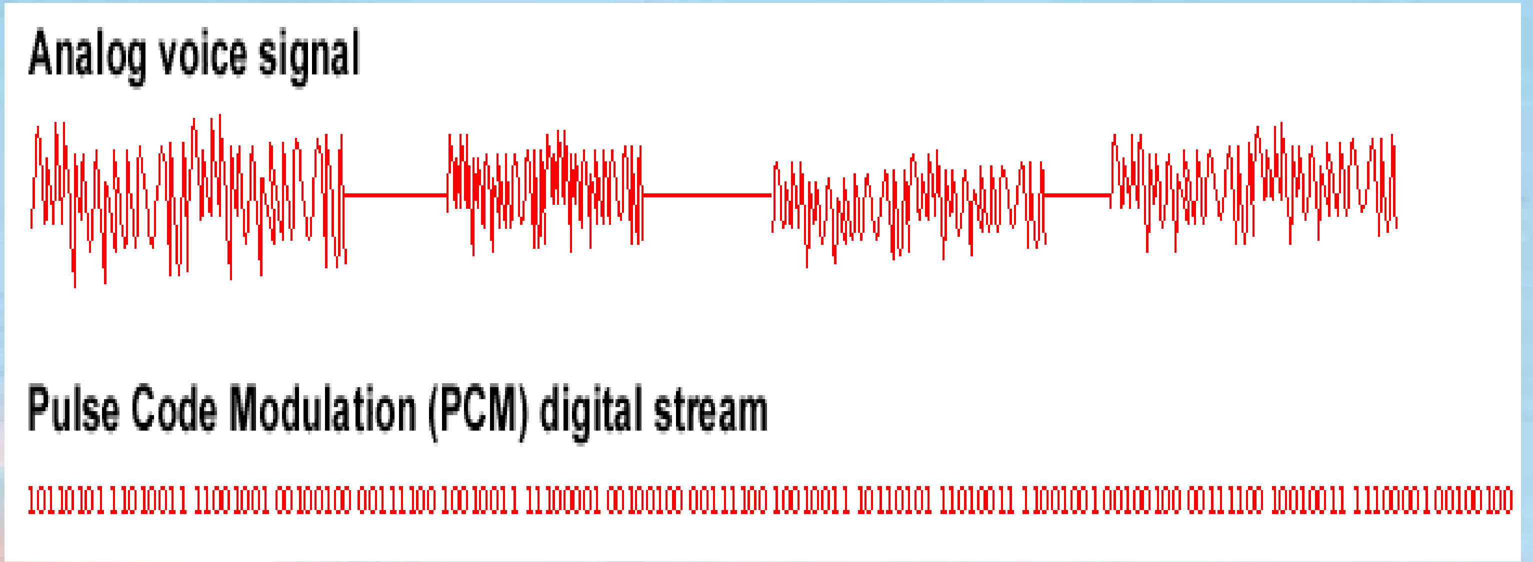


Figure 1. Jitter Increases Latency

Analog /Digital conversion



Echo and silence supression

Once PCM stream is analyzed:

10110101 11010011 11001001 00100100 00111100 10010011 11100001 00100100 00111100 10010011 10110101 11010011 11001001 00100100 00111100 10010011 1110000100100100

- Echo is removed

10110101 11010011 11001001 .0010011 11100001 00100100 00111100 10010011 10110101 11010011 11001001 00100100 00111100 10010011 1110000100100100

- Silence is removed by VAD

1011 11101 1 11001001 .0010011 .001001 .00 10010011 10110101 11010011 110010)0 00111100 1110000100100100

- Remaining PCM samples are forwarded to the CODEC

10110101 11010011 11001001 00100100 00111100 10010011 11100001 00100100 00111100 10010011 10110101 11010011

Factors that affect voice quality

- Delay
- Jitter
- Packet loss
  - Isolated losses
  - Burst losses
- Voice compression
- Echo
- Digitizing Distortion

PCM Technology

PCM (Pulse Code Modulation) technology is a technique based on scalar quantification of the voice stream. The analog voice signal is directly coded in binary format. Quantification may be uniform or non-uniform, depending on the application.

The PCM method was first defined in CCITT/ITU standard G.711. It is based on the modulation of coded pulses, and uses 64 kbps. After non-linear compression is applied, the amplitude of samples is quantified over 8 bits.

Standard G.721 defined a 32 kbps coding method called ADPCM (Adaptive Differential Pulse Code Modulation) . Rather than measuring the sampling amplitude, this method quantifies the difference between the amplitude and a predetermined value, using an adaptive filter.

ACELP Codec

ACELP (Algebraic Code Excited Linear Prediction) voice compression algorithm is a toll quality dual-rate codec that maintains high-quality sound with a compression rate of 8 kbps or 4.8 kbps. It is ideal for multiplexing applications, can handle DTMF (Dual Tone Multi-Frequency) codes and provides a low-cost solution to maintaining voice quality in high-traffic networks. The ACELP Comfort Noise version also offers bad/lost packet interpolation and reduced bandwidth during silence. The quality of ACELP voice has been extensively tested, with results indicating that it is equal to or better than the industry-standard 32 kbps ADPCM (CCITT/ITU standard G.721). ACELP has a MOS (Mean Opinion Score) of approximately 4.2, which is in the toll quality range.

MOS (Mean Opinion Score)

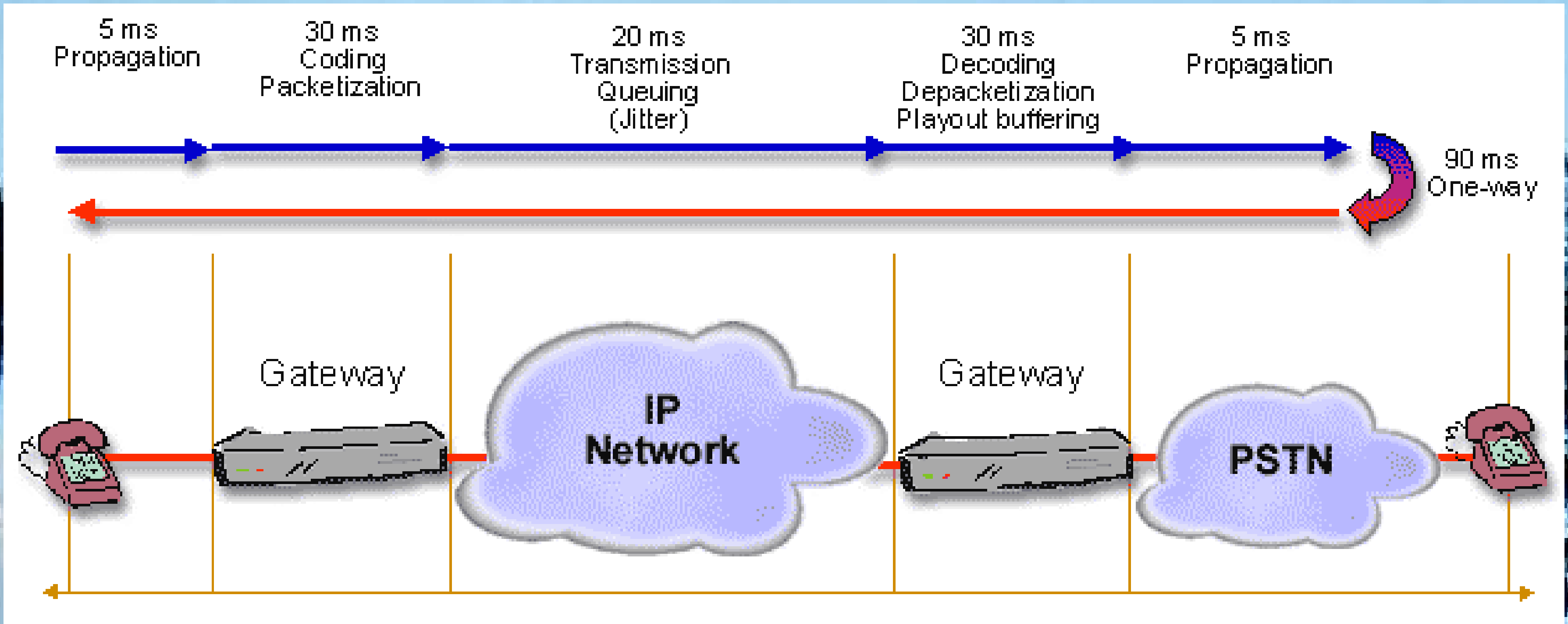
Compression Method	Required speed	MOS
PCM (G.711)	64 kbps	4.4
ADPCM (G.726)	32 kbps	4.2
LD-CELP (G.728)	16 kbps	4.2
CS-ACELP (G.729)	8 kbps	4.,2
MPMLQ (G.723.1)	6.3 kbps	3.98
ACELP (G.723.1) Recommended for IP	5.3 kbps	3.5

ITU-T G.107 presents a mathematical model, known as the E-Model, which attempts to predict QoS scores using more objective impairment factors. TIA/EIA TSB116 provides a comparison of E-Model Rating Values (R) and MOS scores. See Table 1 for details. An R-Value of 94 is equal to a MOS of 4.4[1]

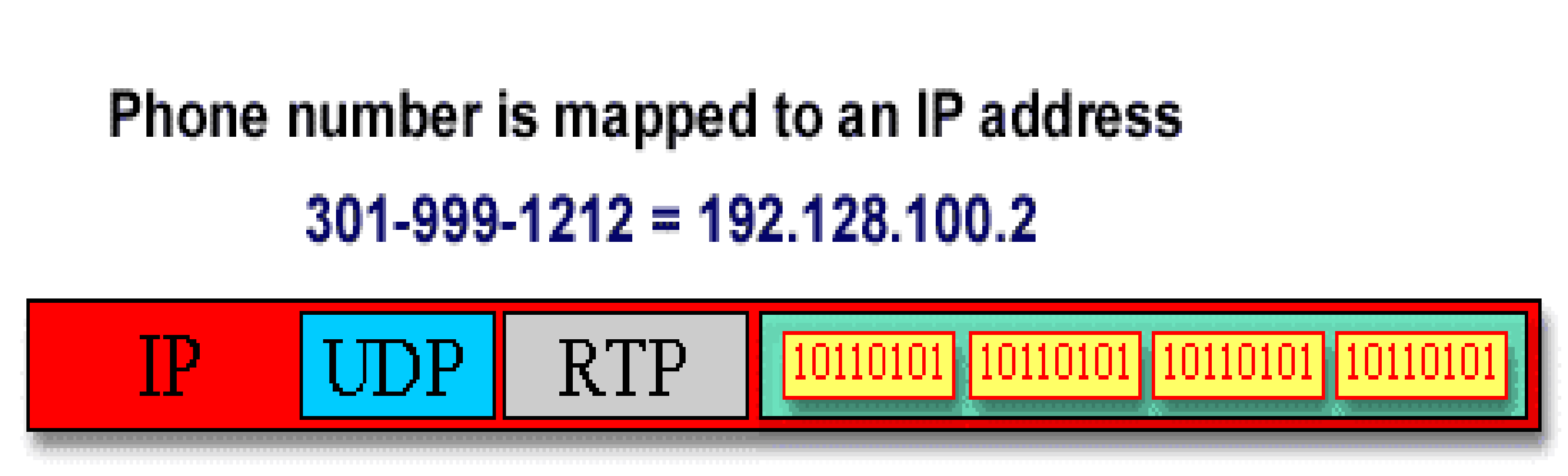
R-Value	Characterization	MOS
90-100	Very satisfied	4.3+
80-90	Satisfied	4.0-4.3
70-80	Some Users Dissatisfied	3.6-4.0
60-70	Many Users Dissatisfied	3.1-3.6
50-60	Nearly All Users Dissatisfied	2.6-3.1
0-60	Not Recommended	1.0-2.6

Table 1. Comparison of R-Values and MOS Scores

Delays

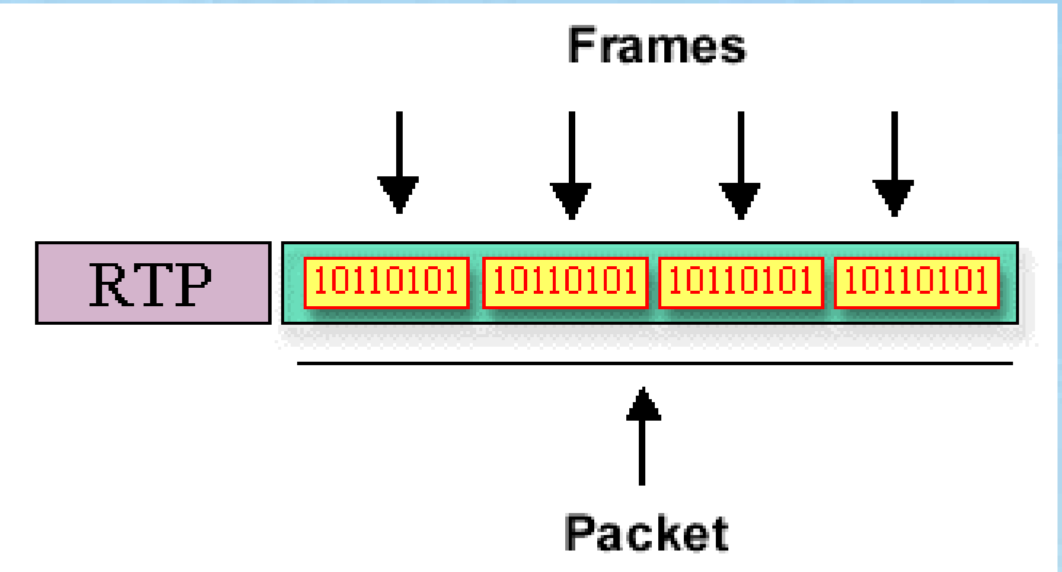

5 ms
Propagation
30 ms
Coding
Packetization
20 ms
Transmission
Queuing
(Jitter)
30 ms
Decoding
Depacketization
Playout buffering
5 ms
Propagation
90 ms
One-way
Gateway
IP
Network
Gateway
PSTN

Phone numbers to IP address mapping



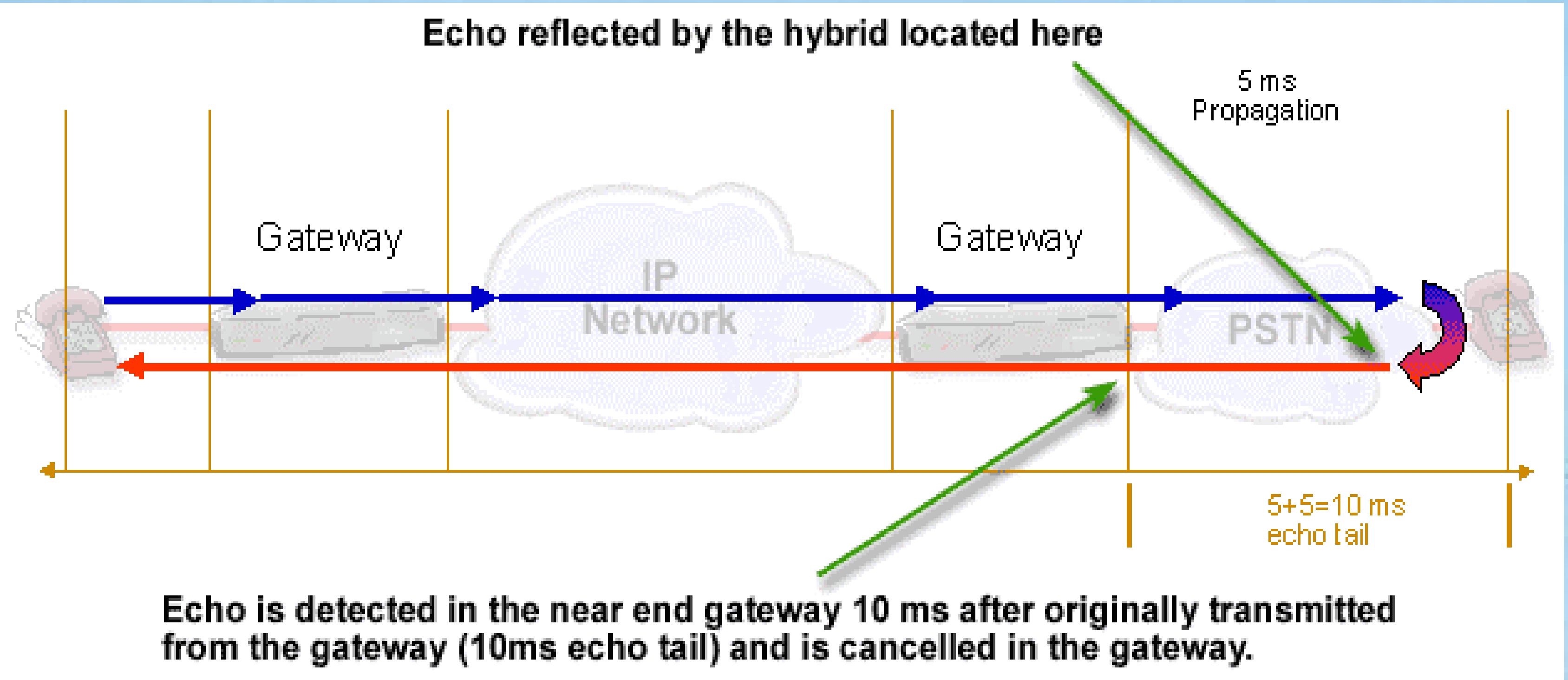
8 bytes UDP and 20 bytes of RTP plus source and destination addresses

Packing of several frames in a packet

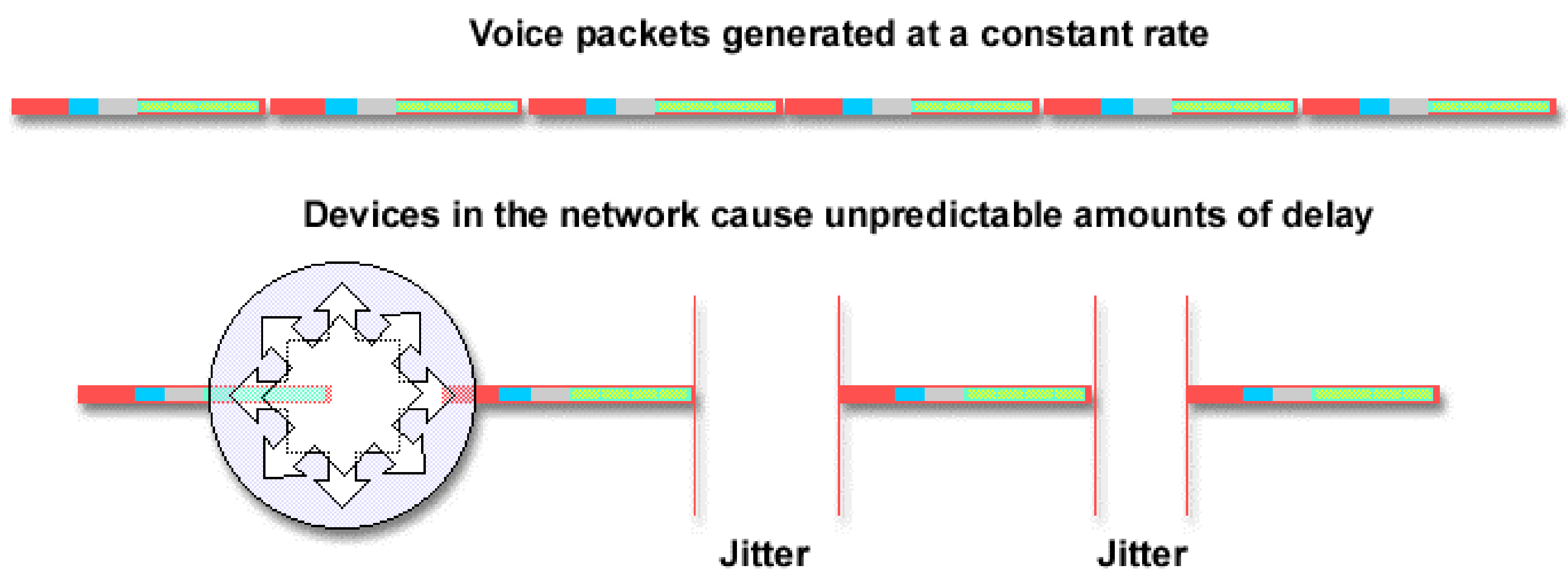


A 12 bytes Real Time Protocol (RTP) header allows for packet prioritization and ordering

Echo


Echo reflected by the hybrid located here
5 ms
Propagation
Gateway
IP
Network
Gateway
PSTN
5+5=10 ms
echo tail
Echo is detected in the near end gateway 10 ms after originally transmitted
from the gateway (10ms echo tail) and is cancelled in the gateway.

Jitter


Voice packets generated at a constant rate
Devices in the network cause unpredictable amounts of delay
Jitter
Jitter

Codec	Speed (kbps)	Segment (bits)	Segments/s	Duration (ms)	Delay (ms)
G.711 (PCM)	64	8	8000	0.125	0.125
G.721 (ADPCM)	32	4	8000	0.125	0,125
G.723 (ADPCM)	24 – 40	3 – 5	8000	0.125	0.125
G.726 (ADPCM)	16 – 40	2 – 5	8000	0.125	0.125
G.727 (ADPCM)	16 – 64	2 –8	8000	0.125	0.125
G.729 (CS-ACELP)	8	80	100	10	15
G.728 (LD-CELP)	16	10	1600	0,625	0.625
G.723.1	6.3	189	33.33	30	37.5
G.723.1	5.3	159	33.33	30	37.5

ITU-T reccomendations

- G.711 **PCM for voice frequencies** 3kHz audio at 48, 56 or 64 kbps.
- G.723 **Multimedia at 5.3 and 6.3 kbps.**
-
- G.728 **15 kbps with Low-Delay code Excited Linear Prediction.**
-
- G.729 **Multimedia at 8 or13 kbps.**
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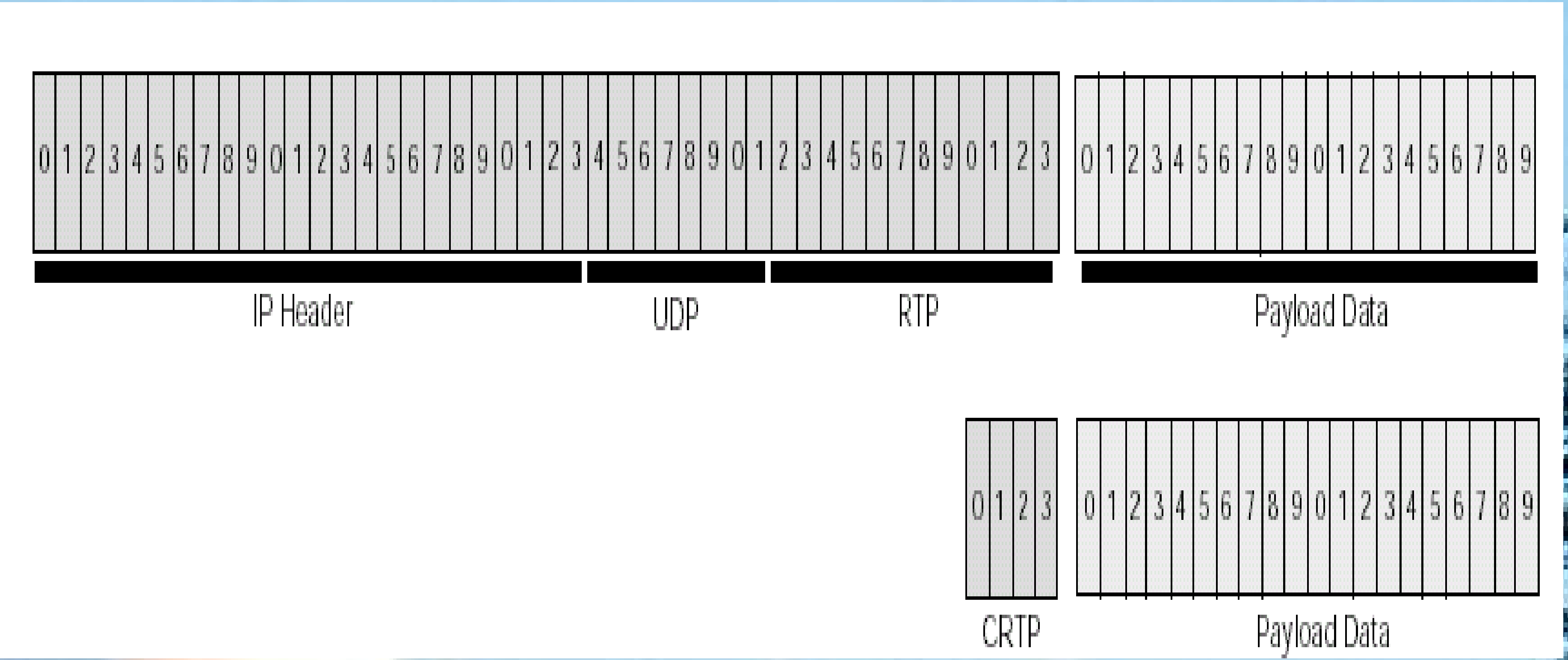
Latency components

- Packetization Latency
- Propagation Latency
- Transport Latency
- Jitter buffer Latency

BW saving

- Headers compression
- Silence Supression
- Frame packing
- Admission Control

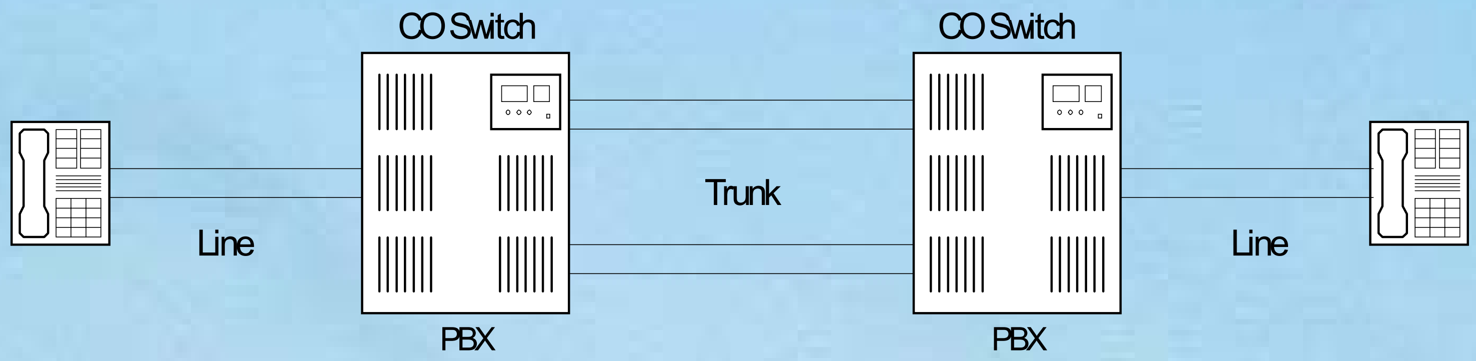
Headers compression



Voice Interfaces

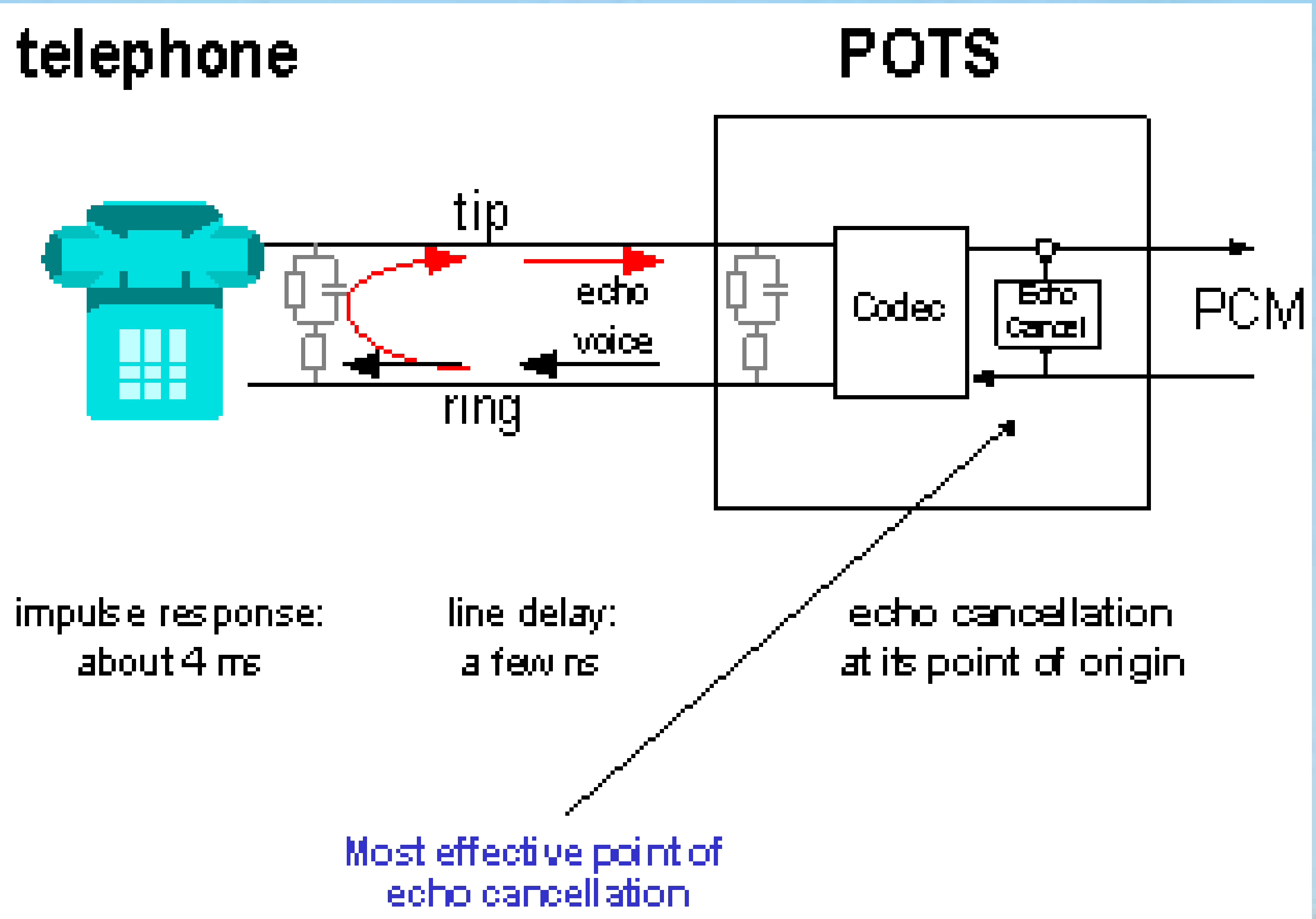
- E&M: “Ear and Mouth” for a trunk connection
- FXO: “Foreign Exchange Office” for Central Office to CO.
- FXS : “Foreign Exchange Station” to connect a telephone or a fax

Classical Telephony

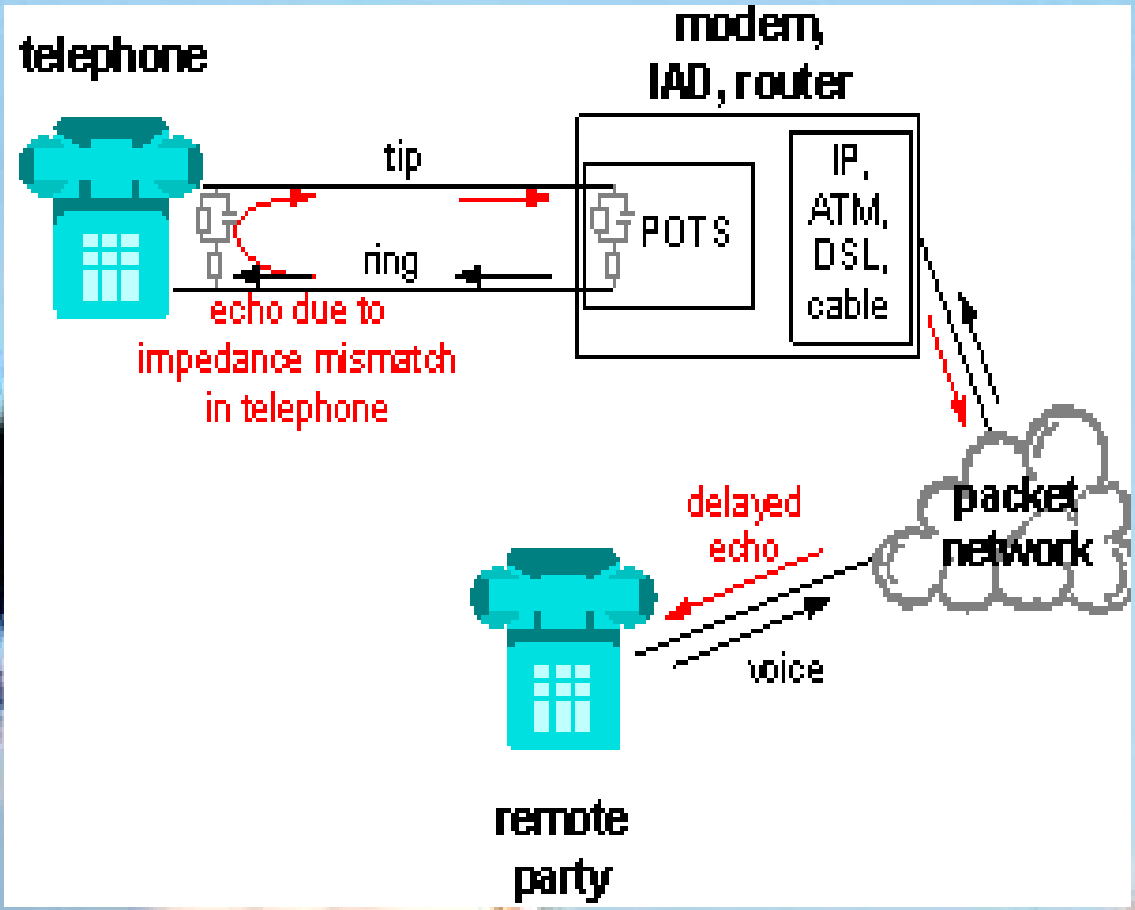


- Local Loop
  - 2 wires line (Tip & Ring)
- Central (CO)
  - Terminates Local Loop
  - Terminates trunk

Echo Cancellation

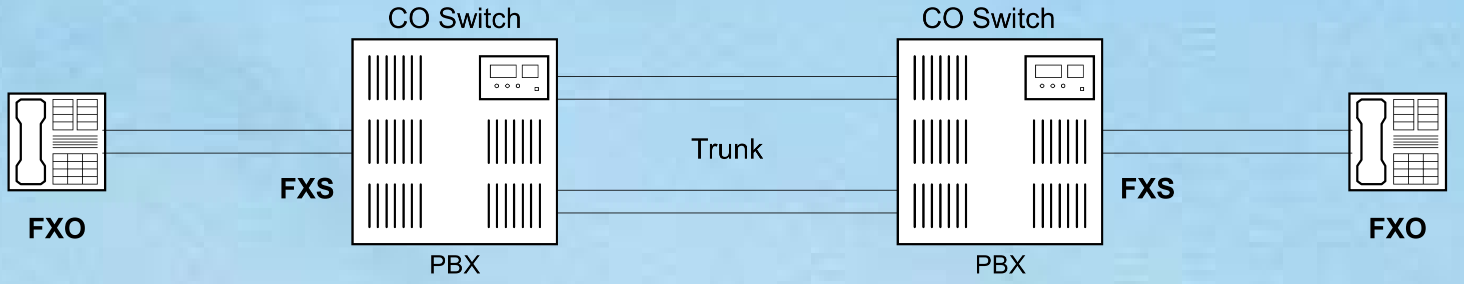


Echo and Delay



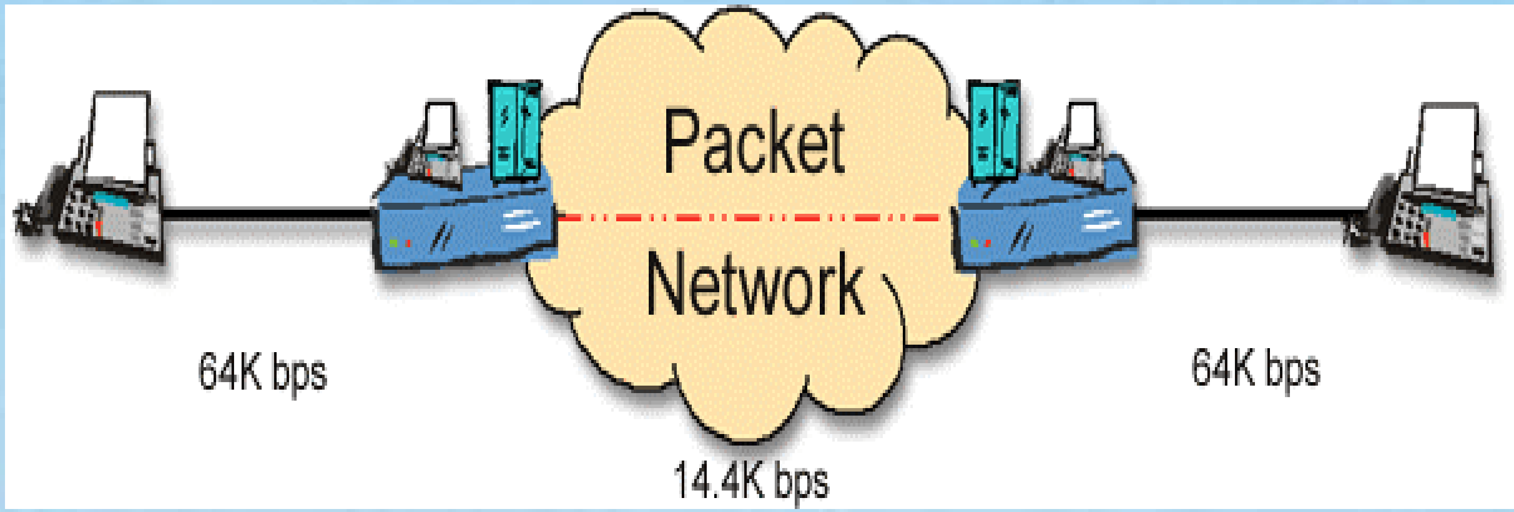
From 30ms. to 50 ms. roundtrip delay and higher, echoes are not automatically suppressed by the human brain, but rather are perceived as very annoying. The typical roundtrip delay in ATM- or IP-based systems is 150ms. to 300 ms.

FXO and FXS

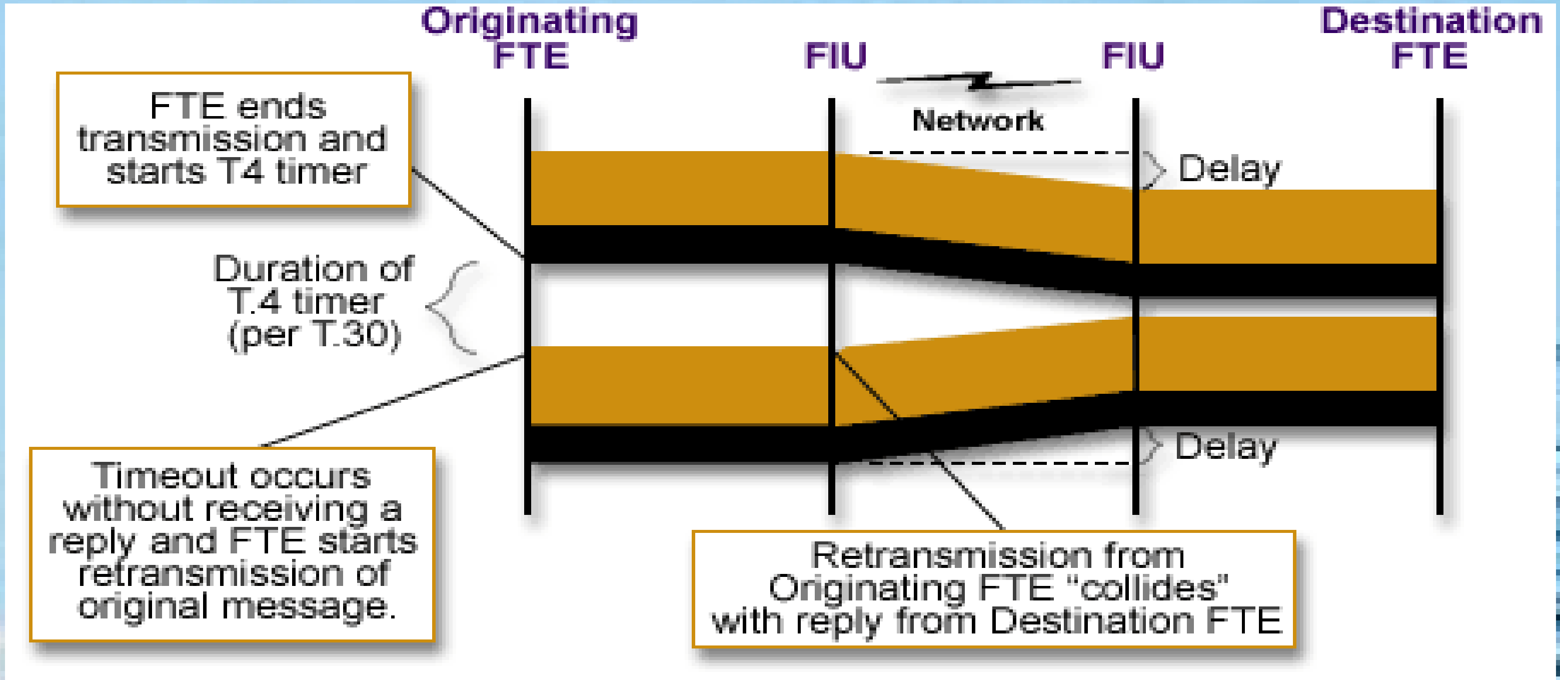


- FXO interface acts like a standard telephone set.
- An FXS port generates a ring so it must connect to a device that can detect a ring. In other words an FXS port must connect to an FXO port over a 2-wire circuit.

Fax over IP



Fax over IP



Fax is more affected by delays tham voice. If the remote station does not receive a response in less 3 s it will drop the call.. Spoofing is needed to avoid this

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Questions?

Ermanno Pietrosemoli

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ULA

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