



Voice Over IP (VOIP) – An Overview

A presentation for the ECE Dept., Montana State University
<http://www.sonoma.edu/users/k/kujoory/> in “Technical Documents”

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Outline

Part 1 – Overview of VOIP

- What Is VOIP And Why
- VOIP Encapsulation
- Voice Compression / Encoding
- Voice Requirements
- Issues With VOIP
- VOIP Security

Part 2 - VOIP Signaling Protocols

- VOIP Signaling Protocols
 - H.323
 - SIP
 - MEGACO
- A Comparison Of Protocols
- Back up slides
 - Quick overview of TCP/IP stack

MEGACO = Media Gateway Control Protocol

SIP = Session Initiation Protocol

TCP = Transmission Control Protocol

UDP = User Datagram Protocol

How Does The Traditional Telephony Work?

- Pick up the phone
 - Wait for a dial-tone
- Dial the destination tel. #
- Remote phone starts ringing
 - Caller is alerted of the ringing at the other side
- Destination picks up and
 - A point-to-point circuit is established
- The circuit carries a digitized version of the voice samples
 - E.g., 8 bits/voice sample, one sample at a time, PCM
- The circuit and BW allocated to the call are devoted to
 - Only one conversation
- During the time between the digitized samples and the silent periods
 - The circuit is idle carrying no information
- **All this time is wasted**
 - **50-70% of the BW**

PCM = Pulse Code Modulation
BW = bandwidth

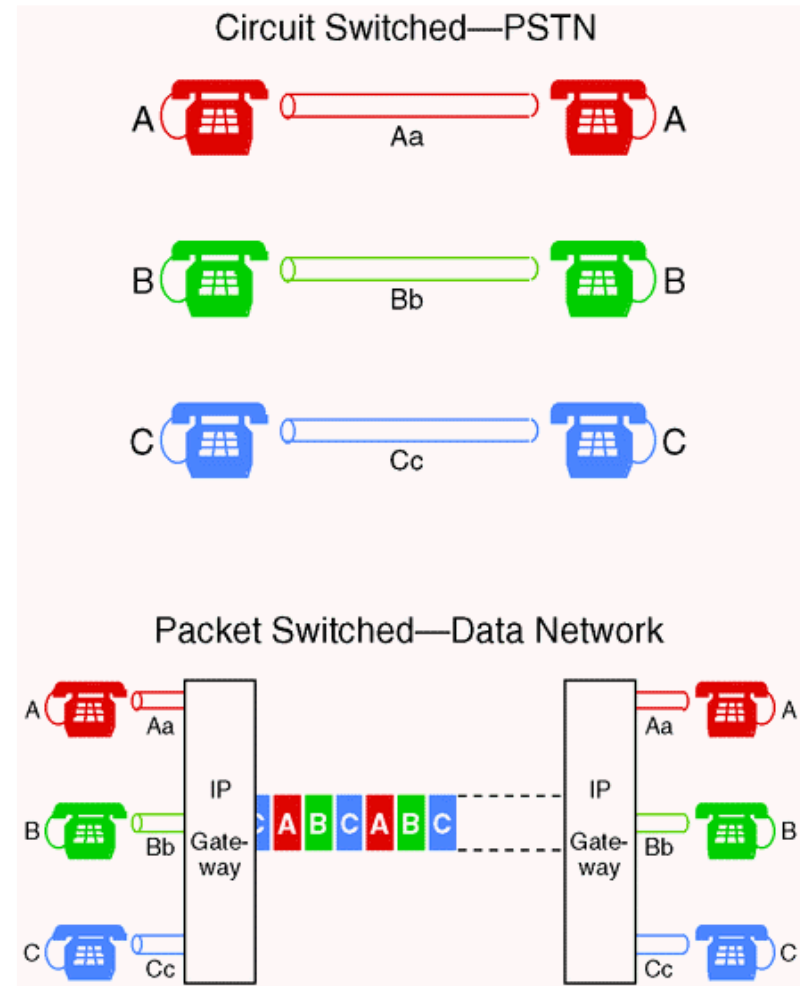
How Can We Reduce the BW Requirement?

- Two ways
 - Accumulate several digitized voice samples together and send them in a packet
 - Signal the silence periods instead of sending them
- Additionally, one can
 - Compress the digitized voice samples to save more BW
- What do you do with the BW that was saved over the network?
 - Use the BW for other calls or applications – an advantage
- But do we lose anything?
 - What are the issues?

BW = bandwidth

What Is VOIP?

- A technology for transmitting ordinary telephone calls over a packet-switched network, Internet
 - Also called IP telephony
- VOIP works through sending digitized voice samples in packets
- Advantages to user
 - Cheaper - avoids the tolls charged by ordinary telephone service
- Advantages to service provider
 - Shares the BW among many users
 - Shares the network among voice, data, and video applications
 - Makes telephony cheaper
- Any issues / impairments?



BW = bandwidth

PSTN = Public Switched Telephone Network

VOIP = Voice over Internet Protocol

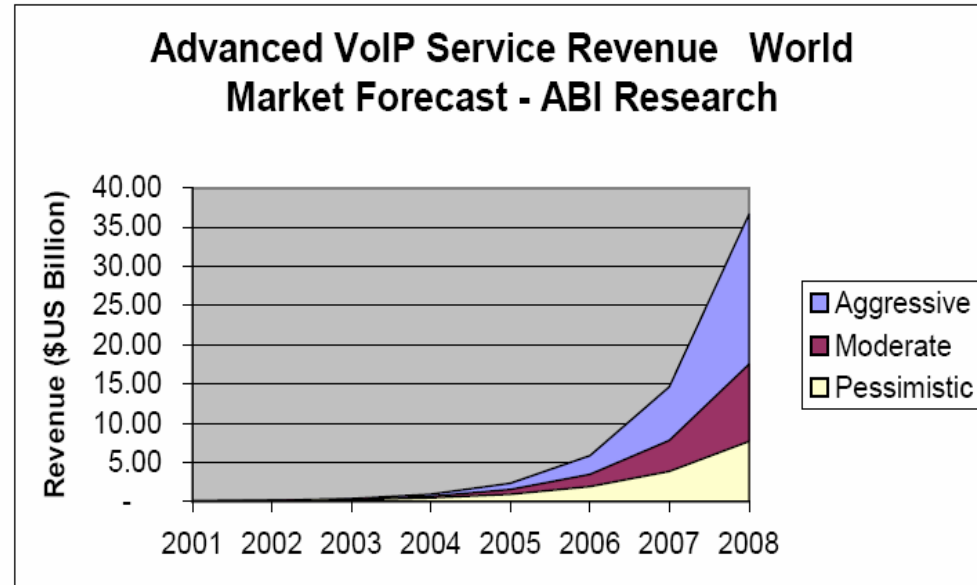
Why Network Operators Are Interested In VOIP?

- PSTN is based on TDM
 - Used primarily for voice traffic
 - Carrying little data directly
- Data and voice traffic have used separate networks
- Data traffic volume has passed voice volume since early 2000
- Makes sense to carry the voice over the data network
 - Amount of additional BW required for voice is minuscule with respect to data
- Using same network for voice, video, and data
 - Allows sharing resources
 - Reduces the CAPEX and OPEX
 - Allows service providers to quickly rollout new services
- VOIP is believed to be much cheaper for subscribers
- Issues
 - VOIP should offer the same quality, reliability, and security as PSTN
 - Including requirements for lifeline, emergency 911, legal call tapping (CALEA)

BW = Bandwidth, CALEA = Commission on Accreditation for Law Enforcement Agencies, CAPEX = Capital Expense, OPEX = Operation Expense, PSTN = Public Switched Telephone Network, TDM = Time Division Multiplexing

VOIP Service Revenue

- VOIP has been picking up in the past few years
 - All service providers offer it
 - Commonly used for international calls
- Market forecasts for VOIP through 2008 vary sharply depending upon confidence in consumer uptake*
- The 'Moderate' case estimates, which exclude consumer uptake
 - Estimate the worldwide market at roughly \$US 8 Billion
- If assumptions of consumer uptake are included
 - Then the market roughly doubles in the "Aggressive" case to \$US 15 Billion*

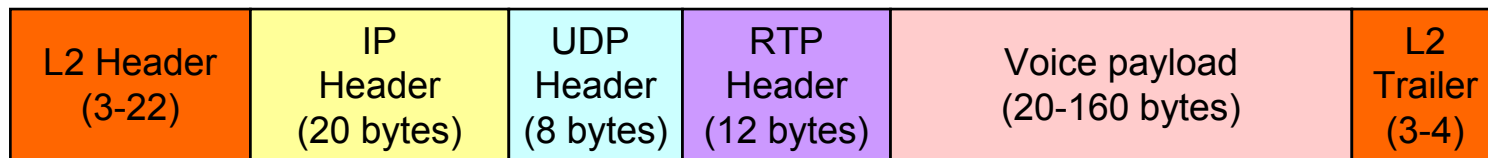


Source: ABI Research

* "VOIP Market Forecast", Publisher: ABI Research, Pub Time: 2003/09, <https://www.abiresearch.com/reports/IPS.html>

Voice Packet Encapsulation

- How are the voice samples carried over the data network?
 - Voice is treated as data
 - Sent as data packets with priority with respect to other data
 - Can end-to-end priority be guaranteed over the Internet?



Notes

- Voice payload comprises one or more frames, each carrying a few encoded voice samples
- Some headers can be compressed
- Layer 2 header and trailer may include flags
 - Typical Layer 2 protocols are PPP, Frame relay, Ethernet
 - PPP = Point-to-Point Protocol, provides framing and link functions
- IP, UDP, RTP are layers 3, 4, and application protocols, respectively
 - UDP = User Data Protocol, provides transport across networks
 - RTP = Real-Time Protocol, provides timing and sequencing

Voice Compression And Encoding

- Legacy telephone channel for voice is
 - 64Kbps as specified by Nyquist's Theorem
- Can we compress the voice, to reduce the BW? **YES**
- However compressing/decompressing introduce additional delay
- A number of standard algorithms offer
 - Acceptable voice quality and delay
- Trade-off between these standards is
 - Quality versus BW
 - The higher the voice quality, the more BW is required
- How good voice sounds is
 - A subjective opinion
 - Difficult for people to describe what sounds good to them
 - Specially when delay and echo are present

Common ITU-T Recommendations For Voice Encoding	BW (Kbps)	Codec Delay (msec)
G.711 PCM (Pulse Code Modulation)	64	0.125
G.726 ADPCM (Adaptive Differential Pulse Code Modulation)	32	0.125
G.728 LD-CELP (Low-Delay Code Exited Linear Prediction)	16	0.625
G.729-CS-ACELP (Conjugate Structure ACELP)	8	15
G.723.1 ACELP (Algebraic CELP)	5.3	37.5

Codec = Voice Coder-Decoder

ITU-T = International Telecommunications Union Telecommunication Standardization Sector

Voice Characteristics And Requirements

Characteristics	Voice	Data
Bit rate	Continuous <ul style="list-style-type: none"> • Constant Bit Rate (CBR) • Queues must be small 	Bursty <ul style="list-style-type: none"> • Variable Bit Rate (VBR) • Queues can be large
Tolerance to packet loss/errors	High <ul style="list-style-type: none"> • < 3 % acceptable packet loss • Uses UDP 	Low <ul style="list-style-type: none"> • Value can change • Usually uses TCP
Tolerance to delay	Low (ITU-T Recomm. G.114) <ul style="list-style-type: none"> • < 100 ms desirable • 150-200 ms acceptable • 400-500 ms unacceptable by many • Meaning can change 	High <ul style="list-style-type: none"> • May be acceptable • Can be asynchronous • Unless for real-time data
Tolerance to delay variation	Low <ul style="list-style-type: none"> • Otherwise not acceptable • Cause distortion 	High <ul style="list-style-type: none"> • Some delay variation acceptable

- **Packetized voice over IP is VBR**
 - **Challenge is how to smoothen VBR to CBR behavior**

Green = Desirable characteristics

Red = Undesirable characteristics

TCP = Transmission Control Protocol

UDP = User Datagram Protocol

Main Sources Delay

- Voice encoding/decoding
 - Voice compression/decompression
 - E.g., 0.125 msec for G.711 encoding
- Accumulation
 - To collect several voice samples in a voice frame before encoding
 - In turn several speech frames may be collected in a voice packet
 - Size depends upon format
- Processing
 - Time to process the packets, RTP/UDP/IP encapsulations
 - Negligible for a fast processor
- Queuing
 - Buffering time in the gateways and routers along the path
 - Can vary
- Transmission
 - Time to transmit the packets over the links along the path
 - The lower the link capacity, the larger the transmission delay, e.g.
 - 100 bytes at 10 Mbps => 0.08 ms
 - 100 bytes at 2 Mbps => 0.4 ms
 - 100 bytes at 64 Kbps => 12.5 ms
 - The larger the packet size, the larger the transmission delay
- Propagation
 - Propagation of voice packets across the media - air, fiber, wire
 - 1000 Km fiber at $2/3 c$ => 5 ms
 - The longer the distance, the larger the propagation delay

c = Speed of light

RTP = Real Time Protocol

How to Deal With Delay Variation

Problems

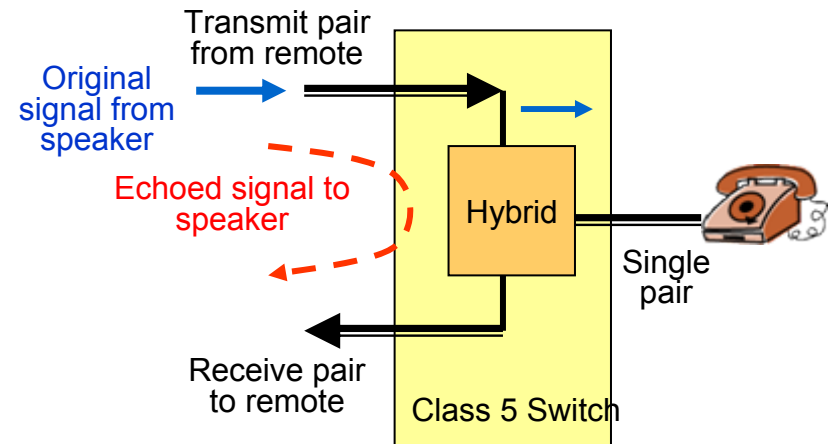
- Some delay components are variable
 - E.g., queuing delay
- IP network offers a best-effort service
 - I.e., packet loss
- Voice quality may suffer from the delay variation and packet loss in IP networks
- How can we handle these impairments?

Solutions

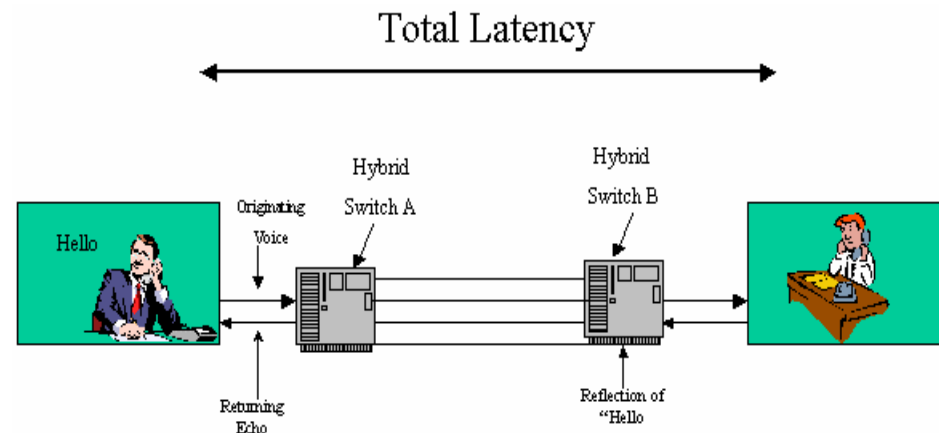
- For delay variation, the receiver buffers incoming voice packets for a period of T
 - T should be $>$ expected delay variation
 - T adds to the total delay, however
 - Increases the total delay that may not be acceptable
 - Then, voice samples are played out at a fixed rate
- For voice packet loss, the receiver may
 - Insert comfort noise, or
 - Play previous packet, or
 - Interpolate

Echo

- When voice signal passes from the 4-wire to the 2-wire
 - Some of the energy in 4-wire circuit is reflected back towards the speaker
 - Due to impedance mis-match
 - 4-wire is used since long-distance calls require amplification/direction
- The echo is acceptable
 - As long as round-trip delay is < 50 msec and is not too loud
 - As in majority circuit switched calls
 - Some echo is desired for talker
- Round-trip delays > 50 ms
 - Require echo cancellation
- In IP networks end-to-end delay > 50 msec due to additional delays
 - Echo cancellation is generally needed
 - An echo canceller is used in the gateway at each end



Speaker voice is echoed back toward speaker in the hybrid



Hello is echoed back from the hybrid in switch B

VOIP Signaling Protocols

- Allow establishing a session between the users to participate in a call
- Three major VOIP Signaling Protocols:
 - ITU-T H.323 – a collection of protocols
 - Specifies “Packet Based Multimedia Communications System”
 - Currently most mature and is popular in enterprise networks
 - Supported by many vendors
 - Microsoft Netmeeting (MM conferencing) is based on H.323
 - IETF MEGACO = ITU-T H.248
 - Telephony signaling protocol based on existing PSTN
 - Upgrade of earlier MGCP
 - Currently supported by many vendors
 - Used by VOIP telephony service providers
 - IETF SIP
 - Client-Server protocol for telephony applications over IP networks
 - Moves application control to the endpoints
 - Supported by some vendors
 - Microsoft Windows Messenger (Instant Messaging) is based on SIP
 - Used by some ISPs
 - May be more common in future

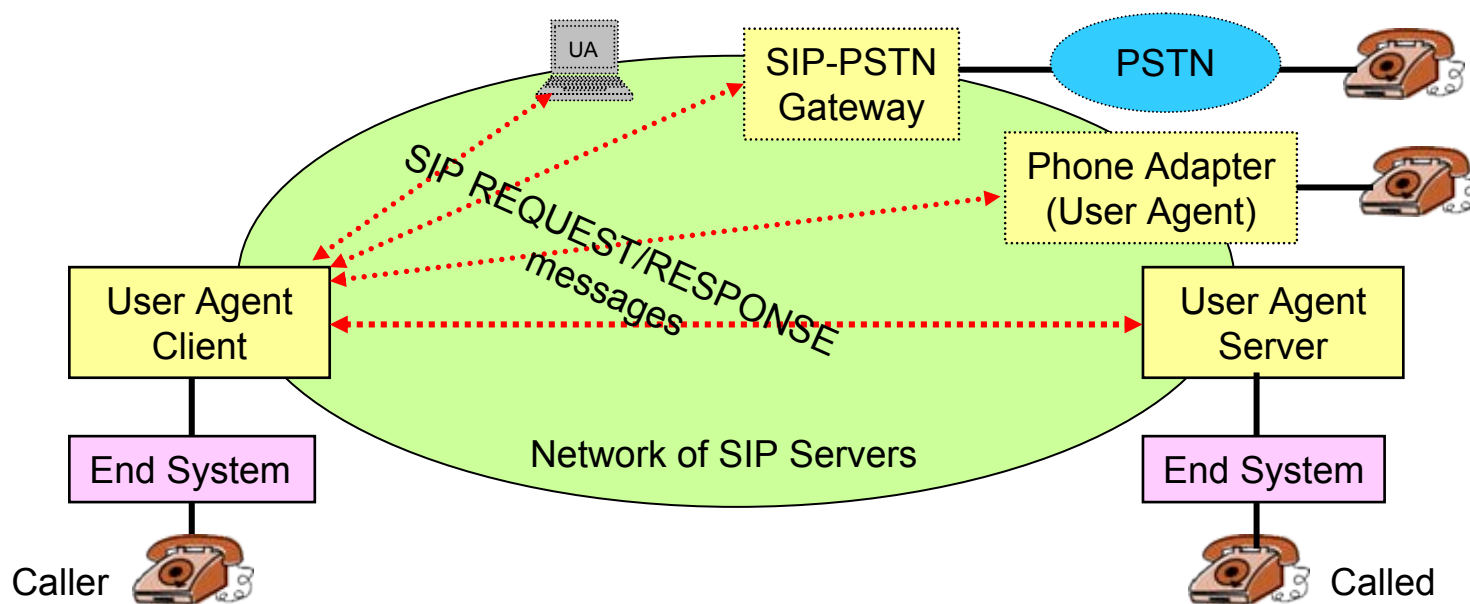
IETF = Internet Engineering Task Force
ISP = Internet Service Provider
MEGACO = MEdia GATeway Control Protocol
MGCP = Media Gateway Controller Protocol
SIP = Session Initiation Protocol

A Comparison of VOIP Signaling Protocols

Item	H.323	H.248/MEGACO (RFC 3015)	SIP(RFC 3265)
Designed by	ITU-T	ITU-T and IETF	IETF
Compatibility with PSTN	Yes	Yes	Largely
Compatibility with Internet	No	Yes	Yes
Call signaling	Q.931 over TCP	Signaling over TCP or UDP	SIP over TCP or UDP
Media Transport	RTP/RTCP	RTP/RTCP	RTP/RTCP
Multiparty call	Yes	Yes	Yes
Multimedia conference	Yes	In future?	Yes w/ other protocols
Addressing	Host or telephone #	Telephone # or IP address	URL
Implementation	Large and complex	Moderate	Moderate
Used by	Enterprise networks	Some telephony providers	Some ISPs and telcos

RTP = Real Time Protocol
 RTCP = Real Time Control Protocol
 URL = Uniform Resource Locator

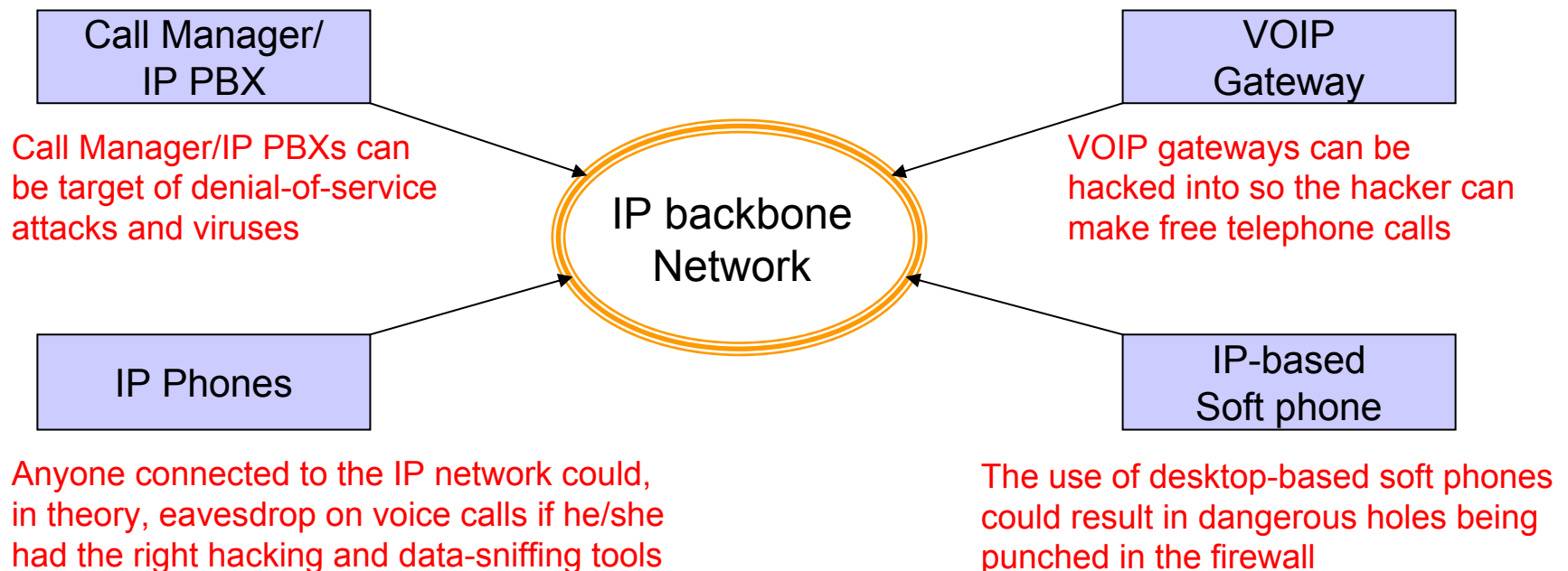
SIP Architecture Simplified



- User Agent Client (UAC)
 - Application which originates SIP requests
- User Agent Server (UAS)
 - Application which contacts user upon receiving SIP request, and
 - Returns user's response on his behalf
 - Accepts, rejects or redirects
- User Agent (UA)
 - Application which contains both UAC & UAS and exchange request/response messages
- UA is a piece of software that can be placed in a computer or a laptop
- Therefore, SIP can offer
 - Various telephony services, e.g.,
 - Internet phones-to-Internet phones
 - Internet phones-to-PSTN phones
 - PC phones-to-PC phones
 - Mobility option

VOIP Security And Vulnerabilities

- In addition to QoS, latency, redundancy, and interoperability issues
 - Security is a fundamental issue when we deal with the Internet
- VOIP is treated as data in the packet-switched network
 - Same precautions for privacy, authentication, service denial are necessary
- **What are the sources of vulnerabilities to security?**



PBX = Private Branch Exchange

Tips For Securing VOIP Traffic

- Encrypt VOIP traffic and run it over a VPN
- Make sure the firewalls are properly configured
- Check to see if your network and security vendors have
 - Support for SIP and H.323
- Consider segmenting voice and data traffic virtually by using a VLAN
 - To limit the threat posed by packet-sniffing tools, also
 - To minimize disruption in the event of an attack
- Think about using proxy servers in front of corporate firewalls
 - To process incoming and outgoing voice data
- Make sure that server-based IP PBXs are locked down and
 - Protected against viruses and denial-of-service attacks
- Make extra provisions for desktop soft phones
 - That initiate/receive VOIP-based calls
 - Do not allow punch holes in corporate firewall that hackers could exploit

VLAN = Virtual LAN

VPN = Virtual Private Network

Conclusion For Part 1

- Legacy voice networks are converting to VOIP
- Issues under discussion
 - QoS, reliability, security, legal tapping, emergency 911
- Voice quality affected by delay, delay jitter, packet loss, echo
 - Mechanisms are needed to control these impairments
- Delay requirements for voice
 - < 100 msec is desirable
 - 150-200 msec is acceptable, but lower quality is noticeable
 - > 400 is unacceptable
- Echo > 50 msec requires echo cancellation
- Several signaling protocols
 - These will coexist for sometimes
 - The market decides the winner
- Areas for further research
 - Quality of service
 - Traffic and capacity engineering for triple-play
 - Integrated voice, video, data
 - Multicast
 - Security
 - Internetworking

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VOIP Signaling Protocols

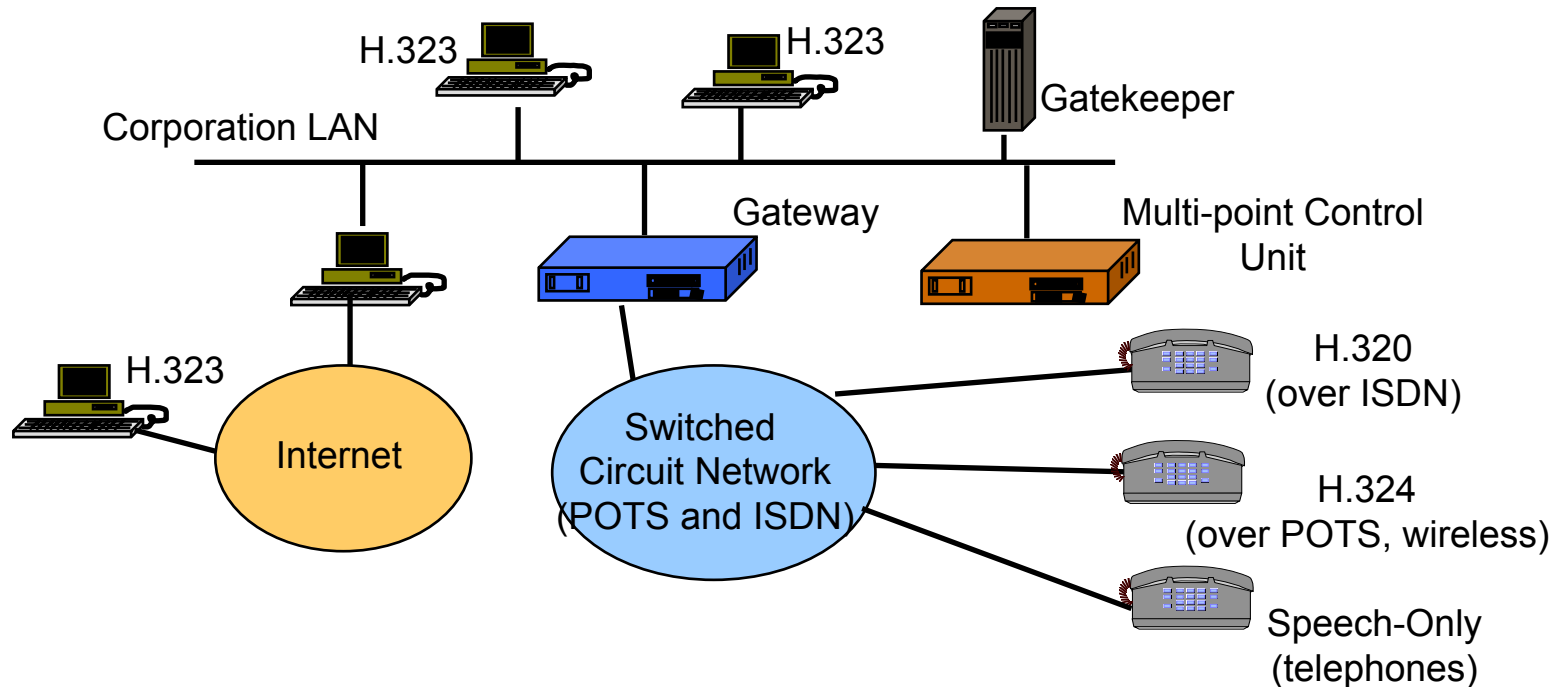
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ITU-T Recommendation H.323

- Standardized in 1996
 - Updated in 1998, 1999, 2000 (V. 4)
- Describes terminals and other entities that provide multimedia communication services over packet-based networks
 - Which may not provide a guaranteed Quality of Service
- H.323 entities may provide
 - Real-time audio, video and/or data communications
- Support for audio is mandatory
- Support for data and/or video is optional
 - But if supported, the ability to use a specified common mode of operation is required
 - This allows all terminals supporting that media type to interwork

H.323 Network Architecture and Components

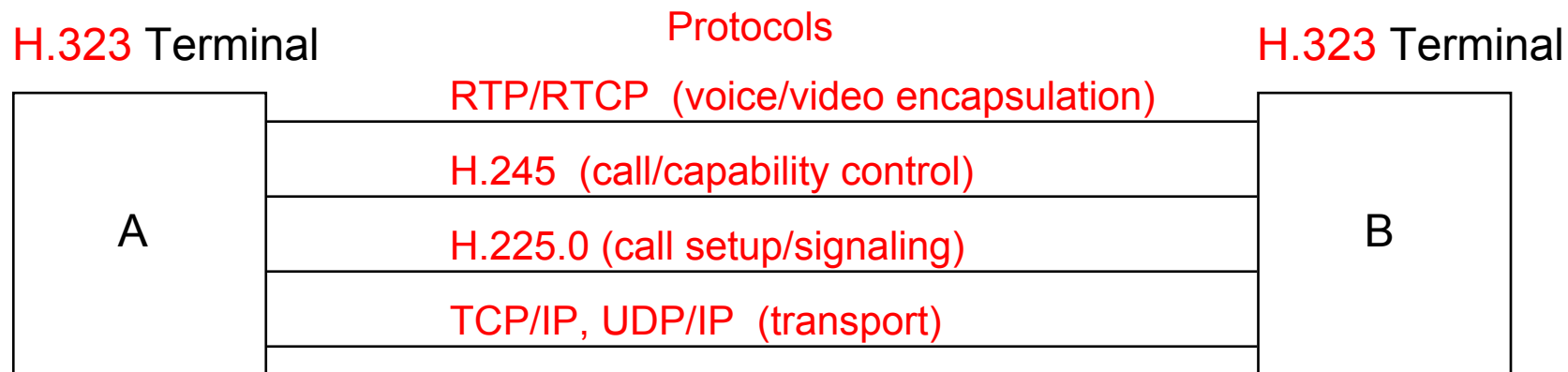


- H.323 terminal handles RTP for MM
- Gateway connects one or more PSTN, N-ISDN, B-ISDN to LAN and
 - Provides audio/video transcoding
- Gatekeeper does address translation and admission control
- Multi-point Control Unit controls multimedia conferencing

- H.323 Zone is the collection of all terminals, gateways, and MCU managed by a single Gatekeeper

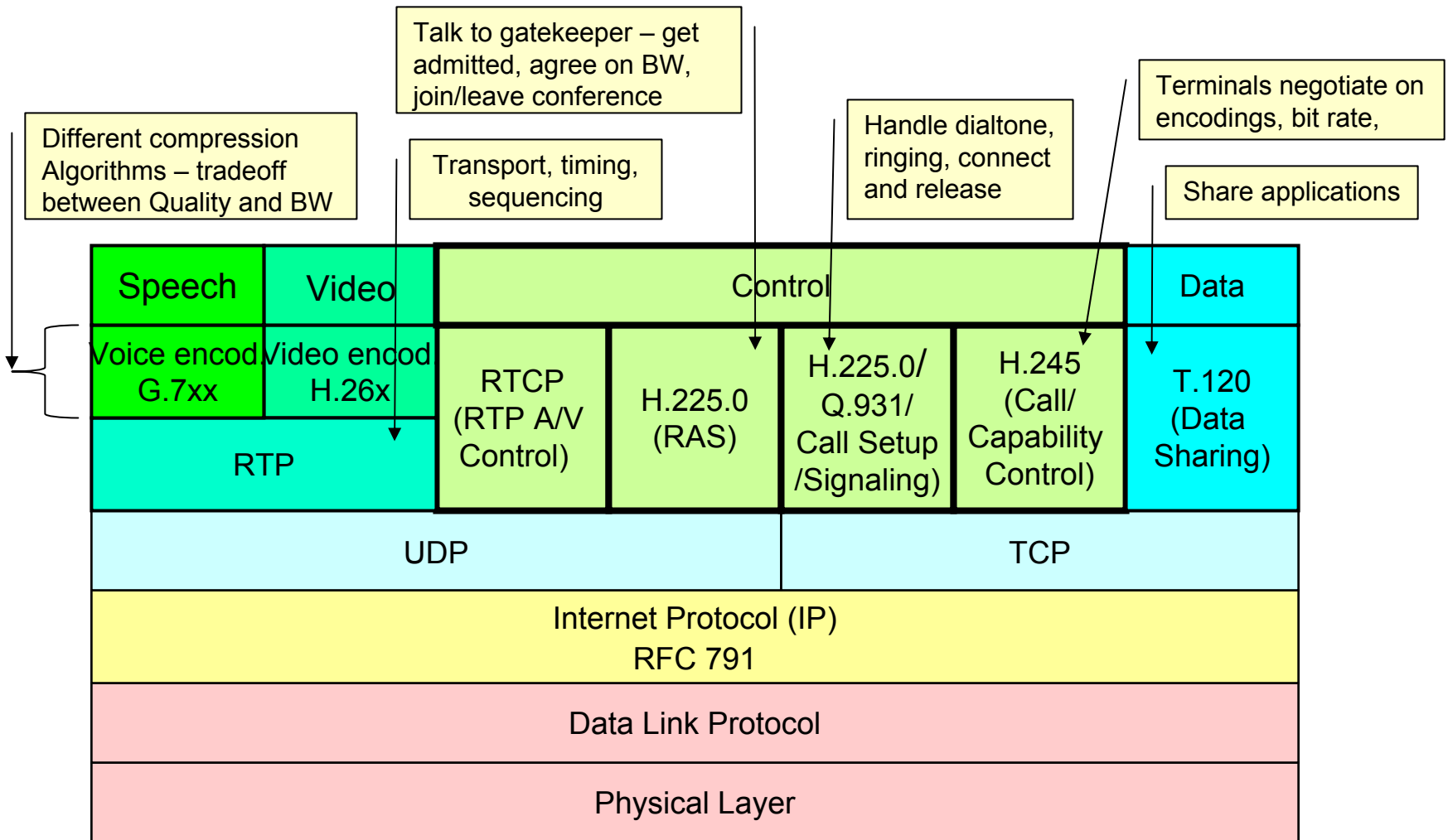
ISDN = Integrated Switched Digital Network
MCU = Multi-point Control Unit
MM= Multimedia
POTS = Plain Old Telephone Service
RTP = Real Time Protocol

H.323 Components And **Protocols**



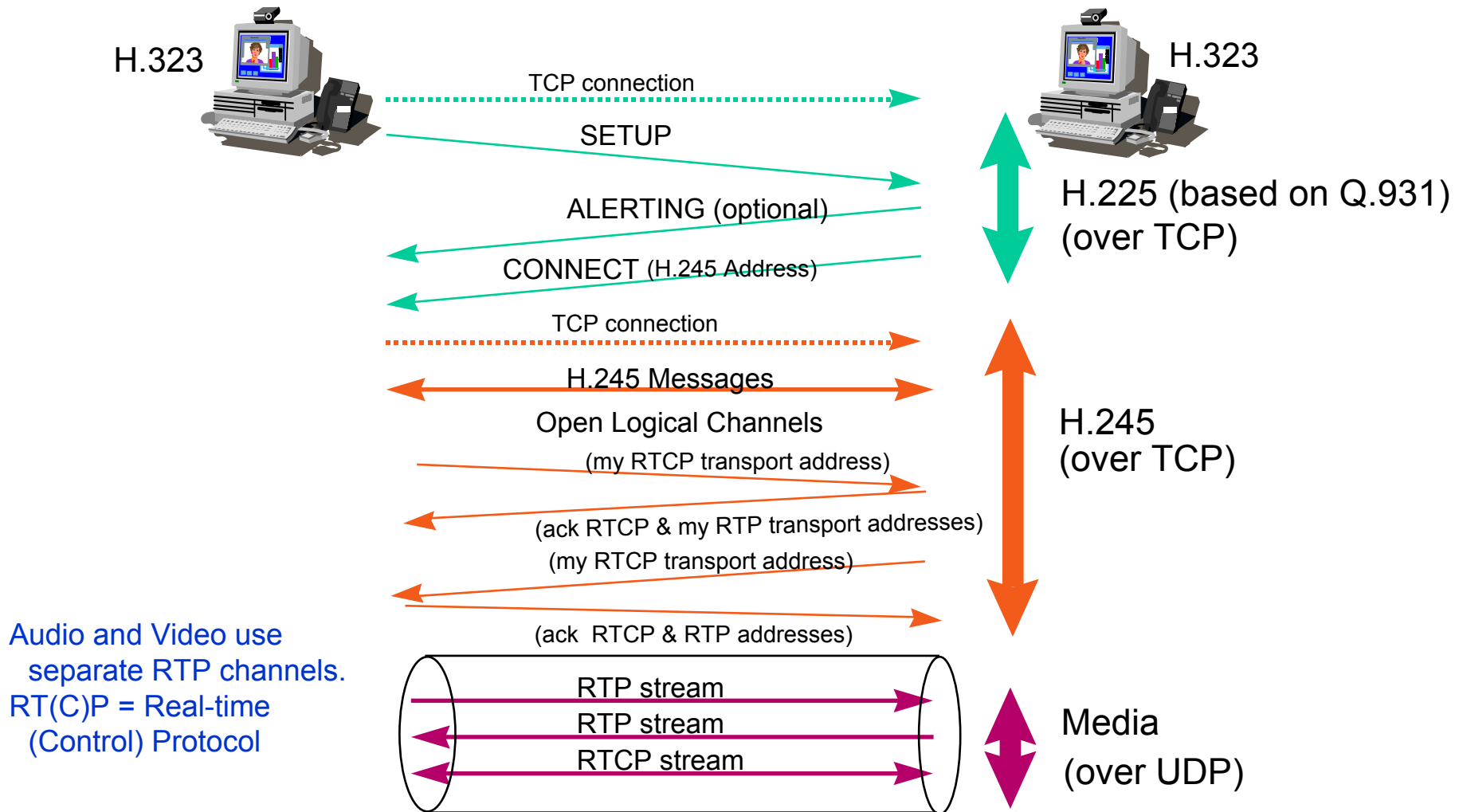
Component	Functions
Terminal	Endpoint, RTP for audio/video support
Gateway (GW)	Connects LAN to PSTN, audio/video transcoding
Gatekeeper (GK)	Address translation, admission control
Multi-point Control Unit (MCU)	Conference control

H.323 Protocol Stack



Bold rectangles represent H.323 scope. Other rectangles support H.323 and provide transport.
 RAS = Registration/Admission and Status RT(C)P = Real-time Transport (Control) Protocol

H.323 Call Exchange Scenario (Direct Call Model)



IETF Session Initiation Protocol (SIP)

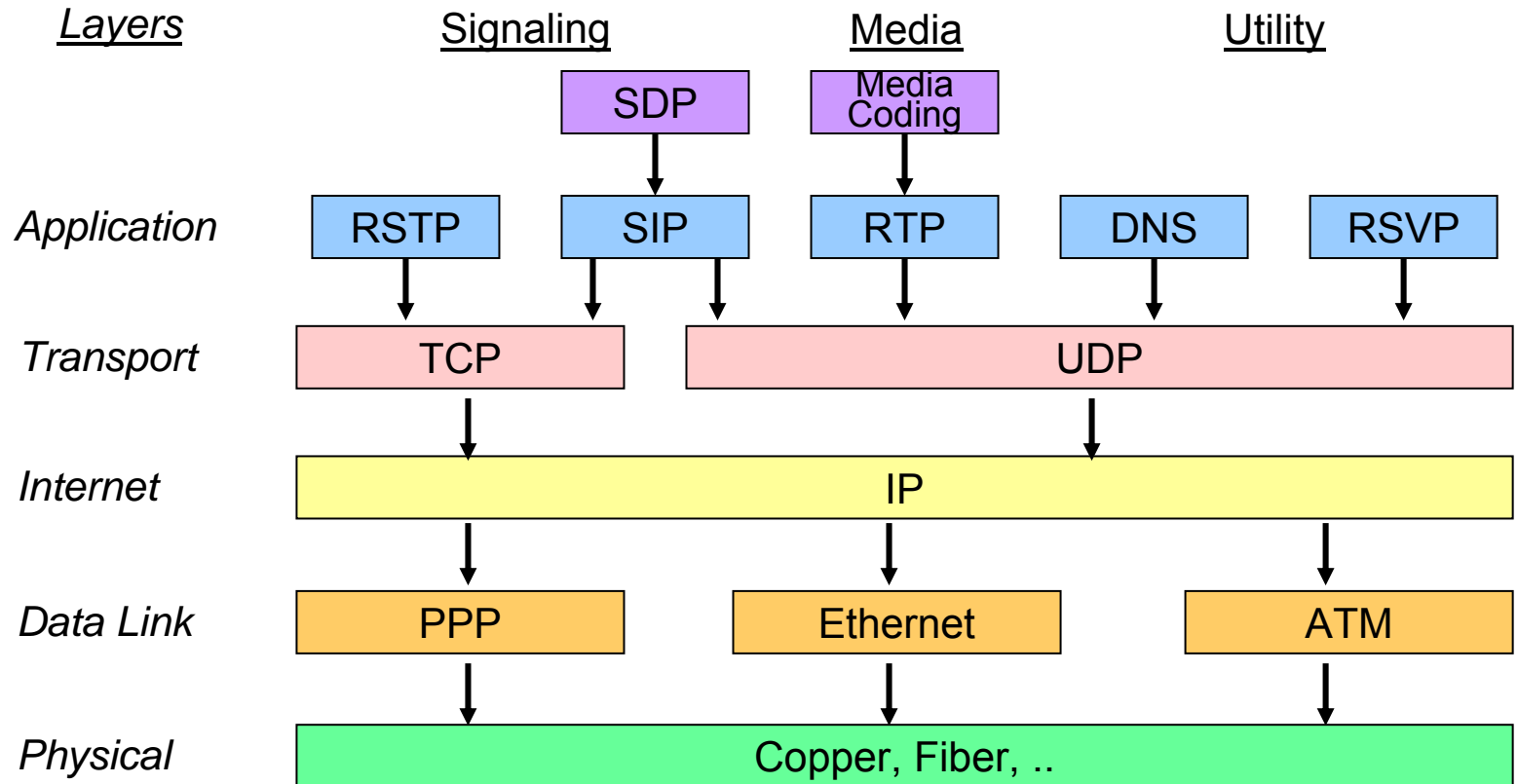
- An application-layer signaling protocol
- Used to establish, modify, and terminate multimedia sessions
 - With one or more participants
 - Sessions include
 - Internet conferences, Internet telephone calls, MM distribution
- Does not know about the details of a session
- Simple, scalable
- Part of Internet Multimedia Architecture
- Can use
 - TCP, UDP, AAL5/ATM
- Applications include
 - Voice and video communication
 - Instant messaging (IM)
 - Voice-enriched e-commerce
 - Web page click-to-dial
 - Call control

SIP (2)

- A request-response protocol
 - For telephony applications
 - Closely resembles HTTP (Web access) and SMTP (email)
- Members in a session can communicate via
 - Multicast or
 - Mesh of unicast relations, or
 - Combination of multicast & mesh
- Can invite both persons and robots, such as a media storage service
- Invitations used to create sessions carry session descriptions
 - To allow participants to agree on a set of compatible media types
- Moves application control to the endpoints, like HTTP or email
 - Eliminates the need for intelligence in the network core
 - A desire to put as much management control as possible at endpoints
 - Legacy telephony has intelligence and management in the network

HTTP = Hypertext Transfer Protocol
SMTP = Simple Mail Transfer Protocol

Some Internet Multimedia Protocols For SIP



DNS = Domain Name System

PPP = Point-to-Pont Protocol

RSTP = Real-time Streaming Protocol (controls video streams, like a VCR)

RSVP = Resource Reservation Protocol

RTP = Real-Time Transfer Protocol

SDP = Session Description Protocol

SIP Protocols

REQUESTS

- **INVITE**
 - Request initiation of a session
 - Most common and important
- **ACK**
 - Confirm that a session has been initiated
- **BYE**
 - Request termination of a session
- **OPTIONS**
 - Query a host about its capabilities
- **CANCEL**
 - Cancel a pending request
- **REGISTER**
 - Inform a redirection server about the user's current location

RESPONSES

- **1xx** - Provisional (Informational)
 - Request received, continued to process request, e.g., 180 = Ringing
- **2xx** – Success
 - Action was successfully received, understood, and accepted, e.g., 200 = OK
- **3xx** – Redirection
 - Further action must be taken to complete the request, e.g., 305 = Use Proxy
- **4xx** - Client Error
 - The request contains bad syntax or cannot be fulfilled at the server, e.g., 484 = Address Incomplete
- **5xx** - Server Error
 - The server failed to fulfill an apparently valid request, e.g., 500 = Internal Server error
- **6xx** - Global Failure
 - The request is invalid on any server, e.g., 600 = Busy

Simple Call Scenario (REQUEST)

Client -> Server:

INVITE sip:watson@petaluma.comcast.net SIP/2.0

Via: SIP/2.0/UDP california.comcast.net

From: A. Bell <sip:a.g.bell@comcast.net>

To: T. Watson <sip:watson@comcast.net>

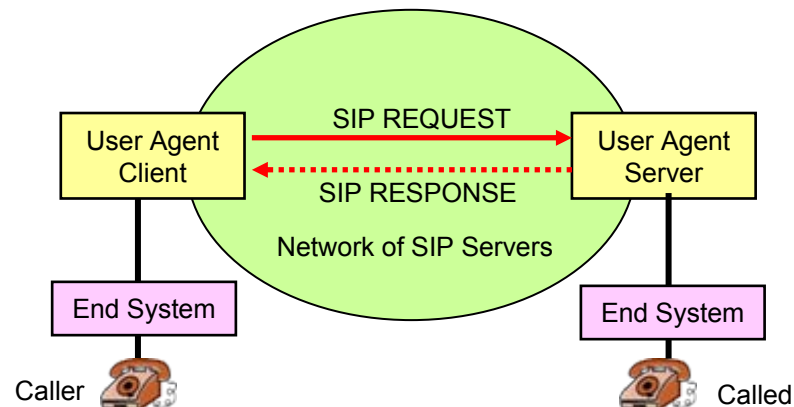
Call-ID: 7077661234@california.comcast.net

Cseq: 1 **INVITE**

Subject: Signaling Meeting

Content-Type: application/sdp

Content-Length: 120



Message Body:

V=0 /* SDP version*/

/* o=<usname><session-id><version><net-type><adr-type><addr>*/

o=bell 53655765 2 IN IP4 128.3.4.5

s=Bell's Call /* session name*/

i=Mr. Watson, Come here. /* session info*/

/* c=connection information=<net-type><adr-type><addr> */

c=IN IP4 petaluma.comcast.net

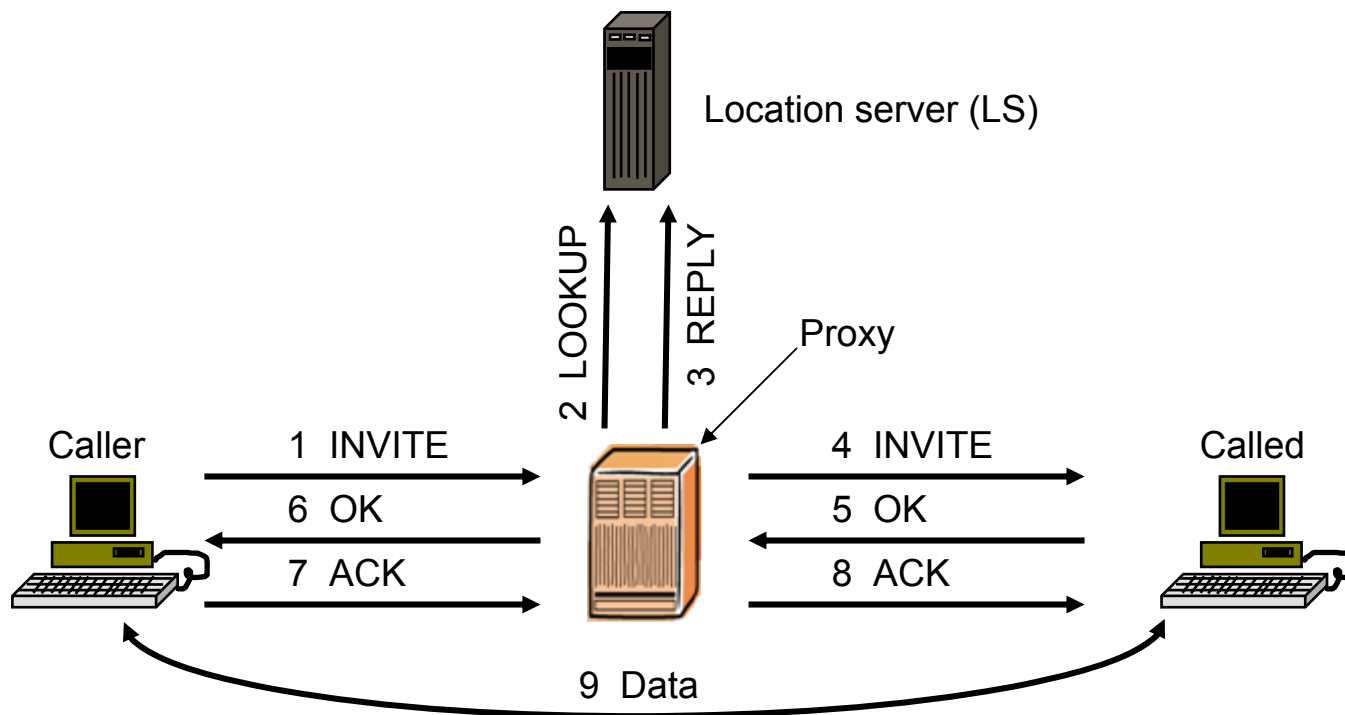
/* <media><port><transport><fmt-list> */

m=audio 5004 RTP/AVP 0 3 4 5

The "o=" line is not strictly necessary for two-party sessions, but **MUST** be present to allow re-use of SDP-based tools.

See "SIP session setup example" in RFC 3262, page 11.

SIP With Proxy And Redirection Server

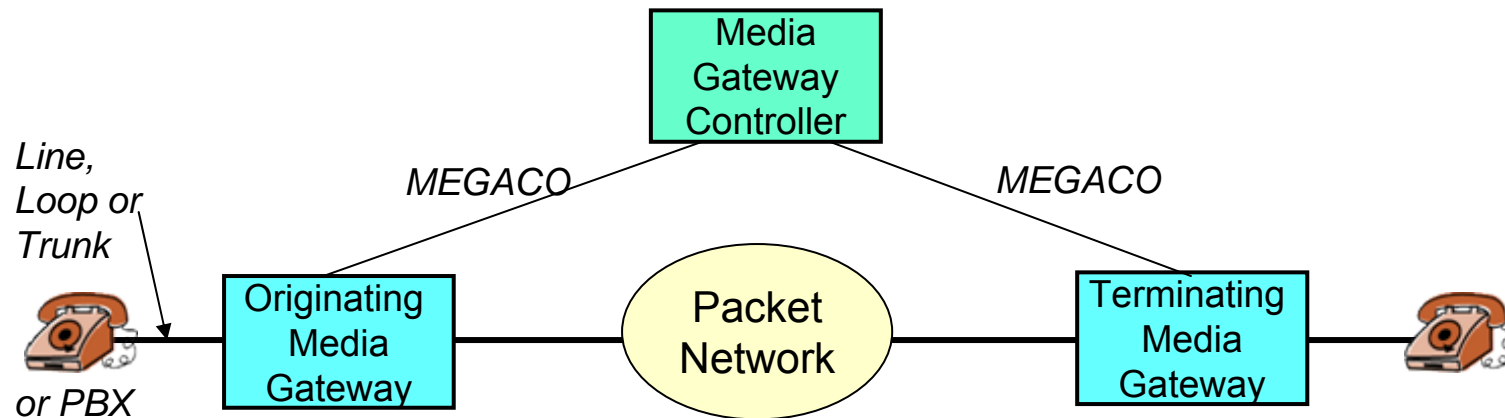


- The REGISTER method relates to SIP's ability to track down and connect to a user who is away (mobility)
 - This message is sent to a LS
 - LS keeps track of who is where
- Caller can send INVITE message to a Proxy server
 - To hide possible redirect operation
- Proxy consults LS to locate Called
- Proxy sends INVITE message to Called
- Proxy relays and performs 3-way handshake for the messages
- LOOKUP and REPLY are not part of SIP
 - Other protocols can be used

SIP Related Protocols

- SIP is specified by IETF RFC 3261
- SIP can use a number of other protocols, e.g.,
 - RTP (Real-Time Transfer Protocol, RFC 3550)
 - For transport
 - SDP (Session Description Protocol, RFC 3266)
 - For describing MM session parameters
 - RADIUS (Remote Authentication Dial In User Service, RFC 2865)
 - To authenticate users
 - LDAP (Lightweight Directory Access Protocol, RFC 2251)
 - To provide directories
 - RSVP (Resource ReSerVation Protocol, RFC 2205)
 - To guarantee voice quality

Media Gateway Control Protocol (MEGACO)

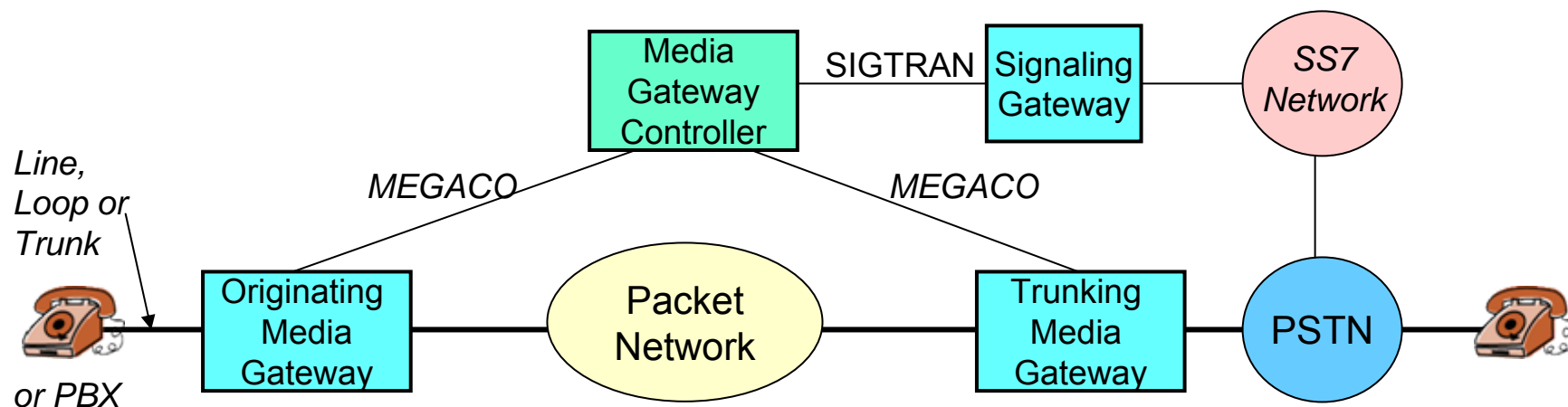


- Protocol used between
 - Media Gateway (MG) and Media Gateway Controller (MGC)
- Enables bearer-independent call setup and control in
 - Next generation network architecture
- Jointly defined by IETF MEGACO WG and ITU-T SG 16
 - IETF: RFC 2805 “Requirements”
 - IETF: RFC 3015 “MEGACO Protocol Version 1.0”
 - ITU-T: Recom. H.248 “Gateway Control Protocol”, plus H.248 Annexes F to K
- Inter-operates with multiple signaling architectures (SS7, CAS, ISDN)
- Supports network access services, multimedia, and multiple gateway for residential telephony

Signaling System No. 7 – out-of-band signaling

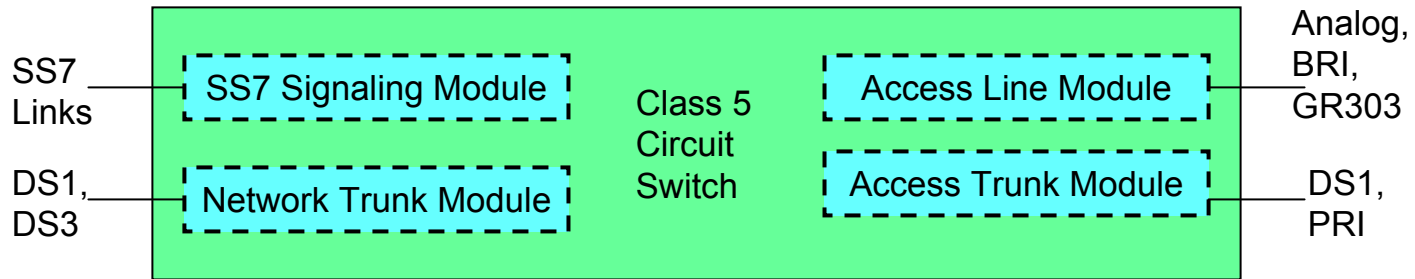
Channel Associated Signaling - Transmission of signaling info within the information band, or in-band signaling. I.e., voice signals travel on the same circuits as line status, address, and alerting signals.

VOIP With MEGACO Connected To PSTN

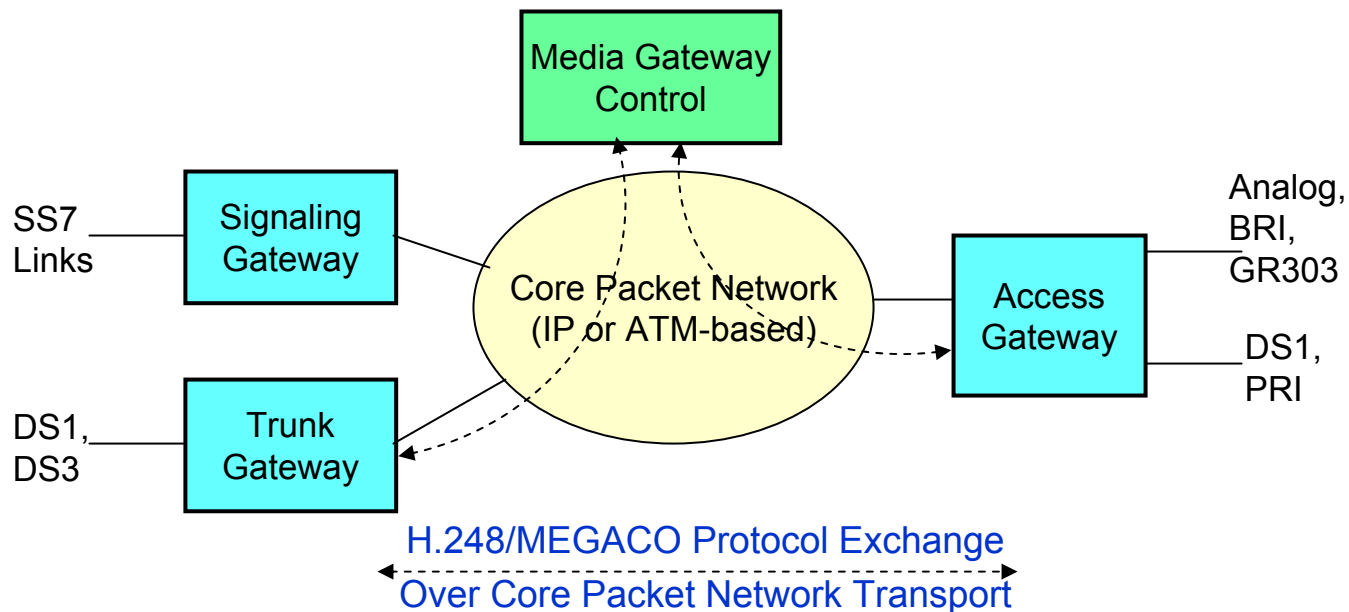


- **Line or Loop** - An analog or digital access connection from a user terminal which carries user media content and telephony access signaling.
- **Trunk** - Analog or digital connection from a circuit switch which carries user content and may carry telephony signaling.
- **Media Gateway** - provides media mapping and/or transcoding functions between user media streams and packet network (e.g., between public switched telephony network and IP or ATM network).
- **Media Gateway Controller** (or softswitch) - controls media gateways and coordinates functions involving multiple media gateways.
- **Signaling Gateway** - terminates switched circuit network signaling at the edge of a packet/cell network (e.g., termination of SS7 signaling in an SS7-internet gateway).

Class 5 Switch Replacement Using MEGACO



Historical Local Class 5 Circuit Switch



Future Distributed NGN Voice Switch

Class 5 Switch functions are separated between MGC and MGs

Examples of MEGACO / H.248 Commands

- **Add** – from MGC to MG
 - Adds a termination to a context
- **Modify** - from MGC to MG
 - Modifies the properties of a termination
- **Subtract** – from MGC to MG
 - Disconnects a termination from its context
- **Notify** - from MG to MGC
 - Allows the media MG to notify MGC of events occurring within the MG



Add (TerminationID
[,MediaDescriptor]
[,ModemDescriptor]
[,MuxDescriptor]
[,EventsDescriptor]
[,SignalsDescriptor]
[,DigitMapDescriptor]
[,AuditDescriptor]
)

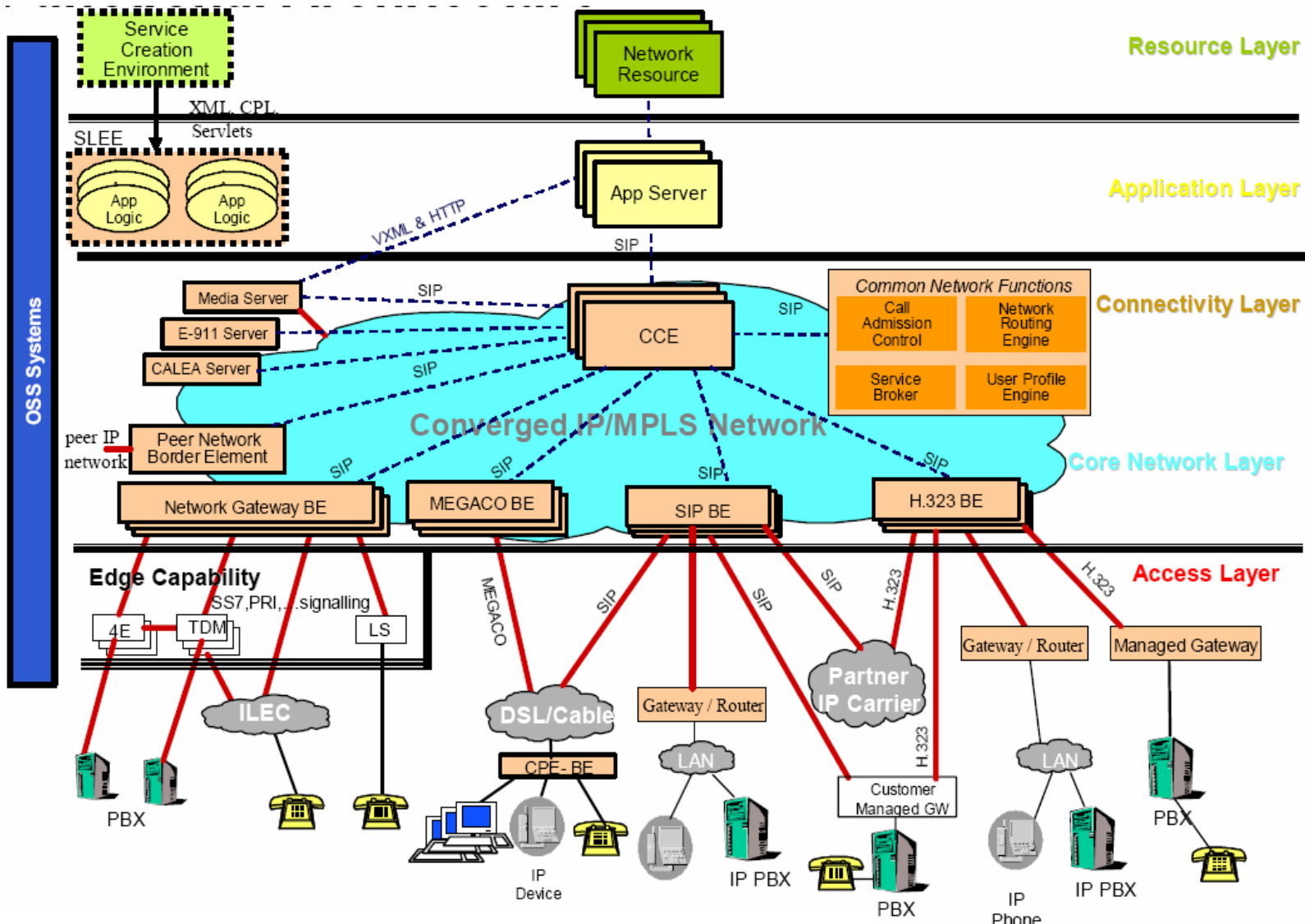
Notes

- Commands are responded by responses in each case
- Descriptors specify the properties of a specific termination
- Event example: on-hook, off-hook
- Signal example: ringing

A Comparison of VOIP Signaling Protocols

Item	H.323	H.248/MEGACO(RFC 3015)	SIP(RFC 3265)
Designed by	ITU-T	ITU-T and IETF	IETF
Compatibility with PSTN	Yes	Yes	Largely
Compatibility with Internet	No	Yes	Yes
Completeness	Full protocol stack	Signaling for voice calls	SIP just handles setup
Parameter negotiation	Yes	Yes for voice calls	Yes
Call signaling	Q.931 over TCP	Signaling over TCP or UDP	SIP over TCP or UDP
Message format	Binary	ASCII	ASCII
Media Transport	RTP/RTCP	RTP/RTCP	RTP/RTCP
Multiparty call	Yes	Yes	Yes
Multimedia conference	Yes	In future?	Yes w/ other protocols
Addressing	Host or telephone #	Telephone # or IP address	URL
Instant messaging	No	No	Yes
Encryption	Yes	Possible	Yes
Size of the main standard	~1400 pages	~200 pages	~270 pages
Implementation	Large and complex	Moderate	Moderate
Used by	Enterprise networks	Some telephony providers	Some ISPs and telcos

Functional Architecture Of A Service Provider*



BE = Border Element
 CALEA = Communications Assistance for Law Enforcement Act
 CCE = Call Control Element
 SLEE = Service Logic Execution Environment
 OSS = Operation Support System

*"Role of 3GPP/IMS in AT&T's RSOIP Network," AT&T whitepaper, V. 2.0, August 1, 2005.

Conclusion For Part 2

- TCP/IP stack a transport for VOIP
 - UDP/IP used for voice transfer
 - TCP/IP used for signaling
- Various VOIP signaling protocol in operation
 - These signaling protocols will coexist for sometimes
- There are products in the market today called call agents or soft switches that
 - Support H.323, SIP, and MGCP, and
 - Allow terminals supporting any of these protocols to place VOIP calls to other terminals
 - Regardless of the signaling protocol the terminal supports
- Eventually market forces will determine
 - Which VOIP signaling protocol(s) should be supported
- Additional advantages of SIP
 - It can be used between softswitches and internal components
 - It can be used as the internal protocol for the VOIP network
 - H.323 and MEGACO can not do this function readily
- With recent FCC rulings
 - Support of E.911 and legal tapping
 - Now required for VOIP applications that have an ability to connect to the PSTN

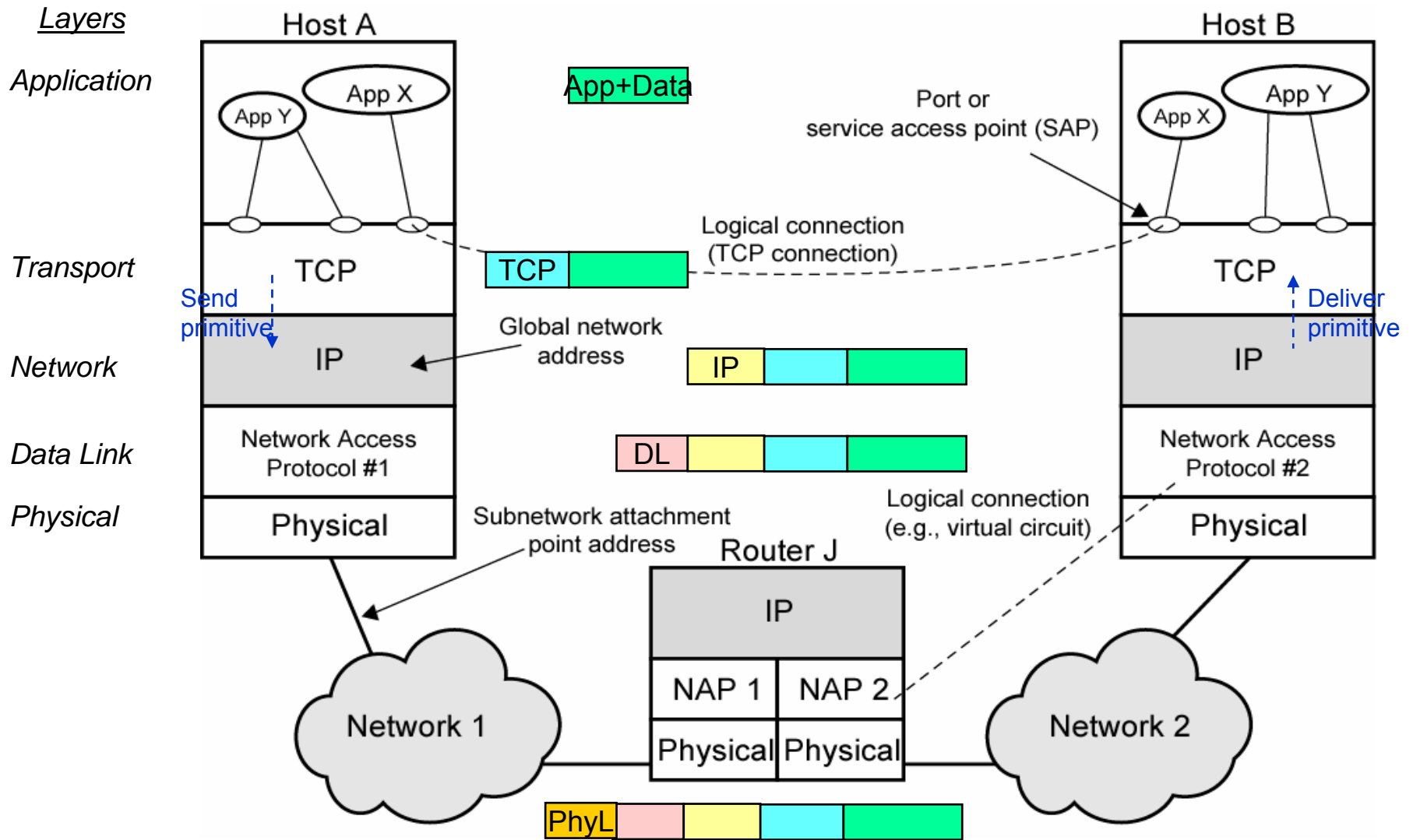
FCC = Federal Communications Commission

References

- ITU-T H.323, "Packet Based Multimedia Communications Systems," ITU-T, 11/2000
- RFC 2205, "Resource ReSerVation Protocol (RSVP) – Functional Specification," IETF, September 1997.
- RFC 2750, "RSVP Extensions for Policy Control," IETF, January 2000 (Updates RFC2205).
- RFC 2865, "Remote Authentication Dial In User Service (RADIUS)," IETF, June 2000 .
- RFC 3015, "MEGACO Protocol Version 1.0," IETF, November 2000.
- RFC 3054, "MEGACO IP Phone Media Gateway Application Profile," IETF, January 2001.
- RFC 3550, "A Transport Protocol for Real-Time Applications," IETF, July 2003 (Obsoletes 1889).
- RFC 3551, "RTP Profile for Audio and Video Conferences with Minimal Control," IETF, July 2003 (Updated 1890).
- RFC 3261, "SIP: Session Initiation Protocol," IETF, June 2002.
- RFC 3265, "Session Initiation Protocol (SIP)-Specific Event Notification," IETF, June 2002 (Updates RFC 3261).
- RFC 3266, "Support for IPv6 in Session Description Protocol (SDP)," IETF, June 2002 (Updates RFC2327).
- RFC 3435, "Media Gateway Control Protocol (MGCP)," IETF, January 2003 (Obsoletes RFC2705).
- AF-SAA-0124.000, "Gateway for H.323 Media Transport Over ATM," ATM Forum, July, 1999.
- "Computer Networks," Andrew Tanenbaum. 4th ed., Prentice Hall 2003.
- "Voice over IP," Uyles Black, 2nd ed., Prentice Hall, 2002.
- "Putting VOIP to Work: Softswitch Network Design and Testing," Bill Douskalis, Prentice Hall PTR, 2001.
- "IP Telephony Demystified" Ken Camp, McGraw-Hill Companies, 2003.
- "Voice-Enabling the Data Network: H.323, MGCP, SIP, QoS, SLAs, and Security," James Durkin, Prentice Hall, 2003.
- "Role of 3GPP/IMS in AT&T's RSOIP Network," AT&T whitepaper, V. 2.0, August 1, 2005.

BACK UP SLIDE

TCP/IP Concepts



TCP/IP Concepts (2)

IP (Internet Protocol)

- A connectionless Layer 3 protocol
- Allows the packet to route the network hop by hop
 - From source to destination
 - One at a time
 - Without any connection setup
- Packets carry full source and destination addresses
- Packets may choose different routes
 - Useful when a network node fails/congests
- Easier for internetworking among heterogeneous networks
- Provides a best-effort service
 - Not reliable (a problem)
 - Packet may get lost due to congestion

TCP/IP Concepts (3)

UDP (User Datagram Protocol)

- A connectionless Layer 4 protocol
- Takes packets from end-to-end
 - From one transport port to another
 - Quickly, low delay
 - Without any connection setup
 - Without acknowledgment (unreliable)
 - Good for real-time applications
 - Voice, video

TCP (Transmission Control Protocol)

- A connection-oriented Layer 4 protocol
- Takes packets from end-to-end
 - From one transport port to another
 - Reliably with delay
 - Need a connection setup
 - With acknowledgment
 - Good for non-real-time applications
 - File transfer, email, web browsing, signaling

ITU-T Common Encoding Standards*

Assume Intrinsic Quality < 70 not satisfactory

Standard	Bit Rate (Kb/s)	Voice Frame Size (msec)	Bits / Sample	Codec Delay (msec)	Intrinsic Qual/100	Year Finalized
G.711 PCM	64	0.125	8	0.125	94.3	1972
G.726 ADPCM	40	0.125	5	0.125	92.3	1990
G.726 ADPCM	32	0.125	4	0.125	87.3	1990
G.728 LD-CELP	16	0.625	10	0.625	87.3	1994
G.729 A CS-ACELP	8	10	80	15	84.3	1996
G.723.1 ACELP	5.3	30	160	37.5	75.3	1995
G.723.1 MPC-MLQ	6.4	30	192	37.5	74.3	1995
G.726 ADPCM	24	0.125	3	0.125	69.3	1990
G.726 ADPCM	16	0.125	2	0.125	44.3	1990

- ADPCM = Adaptive Differential PCM
- ACELP = Algebraic CELP
- CELP = Code Exited Linear Prediction
- CS-ACELP = Conjugate Structure ACELP
- LD-CELP = Low Delay CELP
- MPC = Multi-Pulse Coding
- MLQ = Maximum Likelihood Quantization
- PCM = Pulse Code Modulation

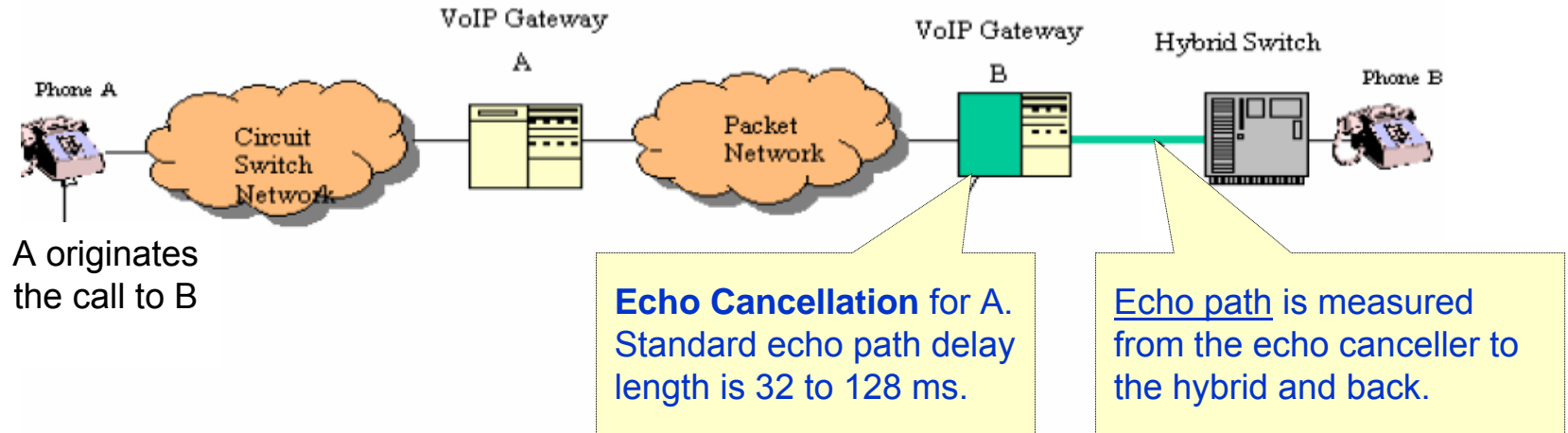
Use different compression algorithms to make different tradeoffs between voice quality and bandwidth. E.g., G.723.1 encoding takes a block of 240 samples (30 msec of speech) to reduce to either 24 bytes or 20 bytes for an output rate of 6.4 kbps or 5.3 kbps (compression factor of 10 or 12).

* From several sources including: Richard Cox, AT&T Labs, IEEE Communications, Sept. 1997, Uyless Black/VOIP

Calculations for the default voice payload sizes in Cisco CallManager or Cisco IOS® Software H.323 gateways

Codec Information				Bandwidth Calculations					
Codec & Bit Rate (Kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Mean Opinion Score (MOS)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Bandwidth MP or FRF.12 (Kbps)	Bandwidth w/c RTP MP or FRF.12 (Kbps)	Bandwidth Ethernet (Kbps)
G.711 (64 Kbps)	80 Bytes	10 ms	4.1	160 Bytes	20 ms	50	82.8 Kbps	67.6 Kbps	87.2 Kbps
G.729 (8 Kbps)	10 Bytes	10 ms	3.92	20 Bytes	20 ms	50	26.8 Kbps	11.6 Kbps	31.2 Kbps
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	3.9	24 Bytes	30 ms	34	18.9 Kbps	8.8 Kbps	21.9 Kbps
G.723.1 (5.3 Kbps)	20 Bytes	30 ms	3.8	20 Bytes	30 ms	34	17.9 Kbps	7.7 Kbps	20.8 Kbps
G.726 (32 Kbps)	20 Bytes	5 ms	3.85	80 Bytes	20 ms	50	50.8 Kbps	35.6 Kbps	55.2 Kbps
G.726 (24 Kbps)	15 Bytes	5 ms		60 Bytes	20 ms	50	42.8 Kbps	27.6 Kbps	47.2 Kbps
G.728 (16 Kbps)	10 Bytes	5 ms	3.61	60 Bytes	30 ms	34	28.5 Kbps	18.4 Kbps	31.5 Kbps

Echo Cancellation

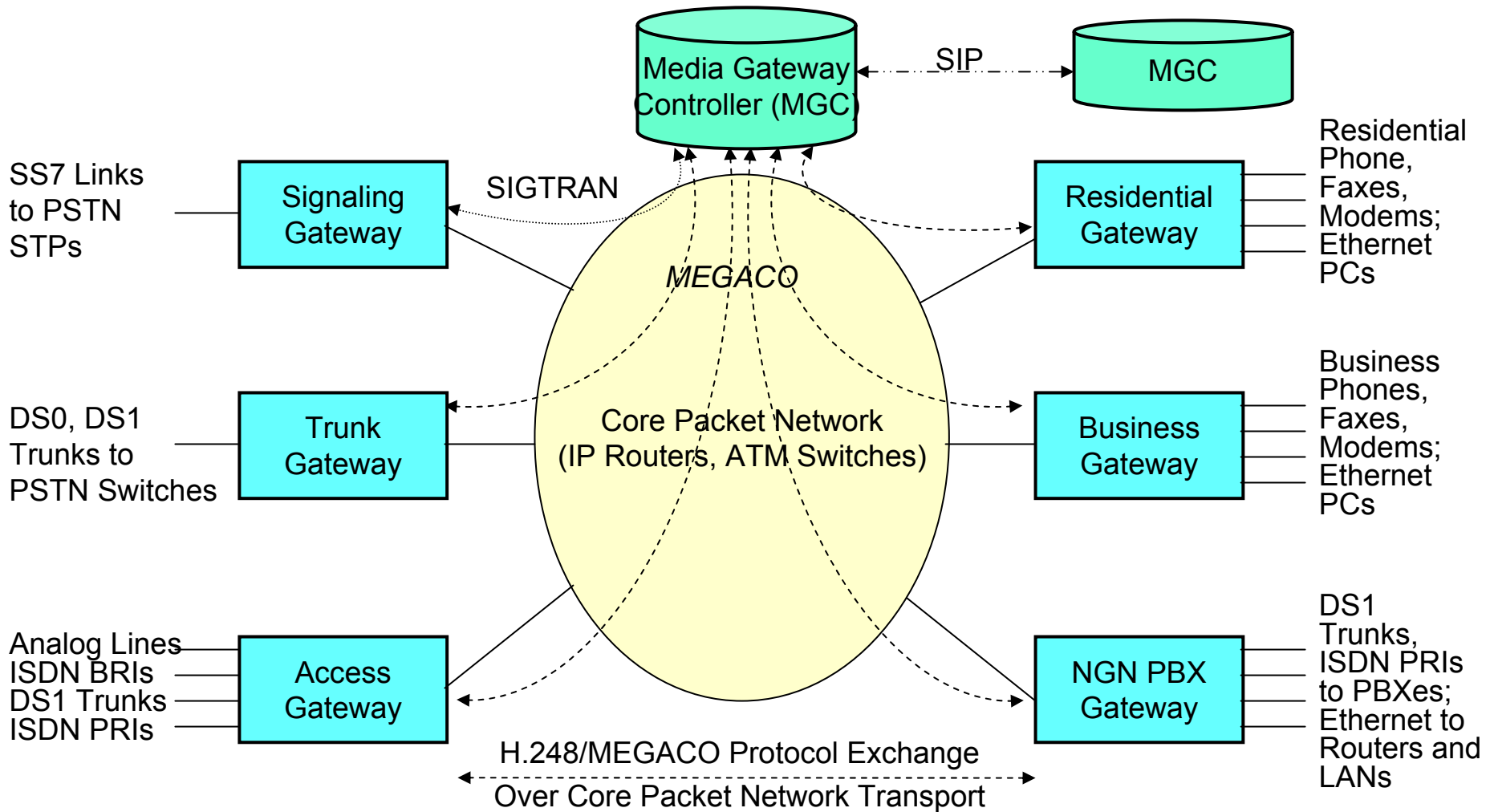


A originates the call to B

- Echo canceller subtracts a delayed and attenuated copy of a transmitted signal from the received signal to cancel the echo component
- To identify a reflected signal, echo canceller first stores the incoming signal
 - All voice traffic transiting IP network is stored in a FIFO buffer in VOIP GW B
 - Size of the buffer is determined by the expected echo path delay
 - The longer the expected echo path delay the larger the required buffer
- Echo cancellation performed in GW B is referred to as Near End Echo Cancellation
 - Since it is nearest to the echo source
- It is possible to perform echo cancellation at the far end, GW A
 - But the echo path will be much longer
 - Requiring more storage & processing power

GW = Gateway

MEGACO in VoP NGN Architecture



- SIGTRAN = Signaling Transport - an IETF standard that deals with Signaling System #7 (SS7; specifically Transaction Capabilities Application Part or TCAP-over-IP)
- STP = Signaling Transfer Point in SS7

VOIP Signaling Protocols And Elements

VOIP Signaling Protocols	Definitions	Components	Related protocols
ITU-T H.323	Packet Based MM Communications System	Terminal, Gatekeeper, Gateway, Multi-point Control Unit	H.225, H.235, H.245. T.120, H.26x, G.7xx, RTP/RTCP
IETF SIP	Client-server-based, request-response signaling protocol for telephony applications	User Agent Client & Server, Proxy, Redirect, and Register Servers	SDP, RTP/RTCP, DNS, LDAP, RSVP, RADIUS
IETF MEGACO = ITU-T H.248	Telephony signaling protocol based on existing PSTN	Media Gateway and Media Gateway Control	RTP/RTCP, DNS, LDAP