Voice over IP Quality of Service and Reliability

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Slides 1

Course Outline

- Introduction (Ch. 1)
  - Overview of Voice over IP systems, QoS issues
- Speech Coding and Voice Quality/Cognition (Ch. 2)
  - A/D converters, Vocoders, Hybrid coding, Perception
- VoIP Network Quality Factors (Ch. 3)
  - Packetization, loss, delay, jitter, echo, reliability
- Internet QoS and VoIP (Ch. 4)
- Network Dimensioning and Reliability (Ch. 5)
- Exam

Agenda

- Overview of VoIP technology and issues
  - Review of Telephony
  - What is VoIP and Why?
  - Basic IP telephony models
  - VoIP Protocols: H.323, RTP, SIP, MGCP, RTSP
  - QoS issues

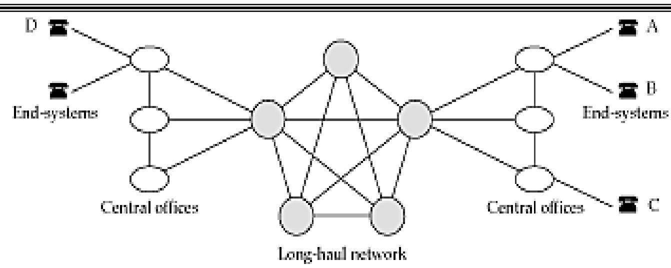
Voice over IP Systems

- Voice over IP (VoIP):
  - Technology providing voice telephony services using the Internet Protocol.
  - Involves converting analog audio to digital, packetization and transport over an IP network.
  - Internet Telephony (IP Telephony) more general concept then VoIP
    - includes VoIP plus signaling/call control and telephony features (call waiting, 800 calls, three way calling, billing, etc.), interworking with existing telephone network
  - Will use VoIP = Internet Telephony = IP Telephony interchangeably
  - Review of Telephony

Public Switched Telephone Network (PSTN)

- Long History 1876 invention of telephone
  - Switch from Analog to digital phones began in 60's
  - Advanced services and digital signaling added 70's
  - Cell phones 90's (now more than wired phones)
- PSTN characteristics
  - Developed for single basic service: *two-way* voice
  - Circuit switched (guaranteed bandwidth) end-to-end connection
  - 64 kb/s continuous voice transmission 8000 samples/sec 8 bits per sample,
  - Everything clocked a multiple of 125 μs
  - low end-to-end delay
  - Call establishment "out-of-band" using packet-switched signaling system (SS7)

PSTN Architecture



- Standardized interface between CO and end-system => digital handsets, cordless/cellular phones
- CO supplies the power for the phone (power life line when electricity out)
- Hierarchically allocated telephone number space (country, area, local)

Transmission Multiplexing

- Calls multiplexed to higher data rates in core
- Multiplexed trunks can be multiplexed further

Digital Signal Number	Number of previous level circuits	Number of voice circuits	Bandwidth
DS0		1	64 Kbps
DS1	24	24	1.544Mbps
DS2	4	96	6.312 Mbps
DS3	7	672	44.736 Mbps

Telephone Switch

- Transfers data from an input to an output
- Switching systems come in two parts: *switch* and *switch controller*
- Switch: time and space division

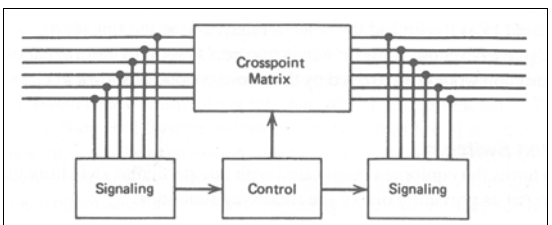
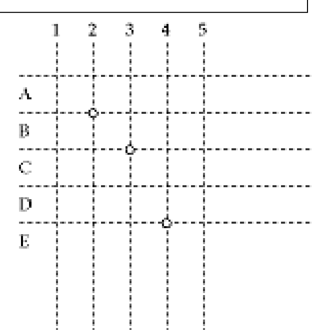
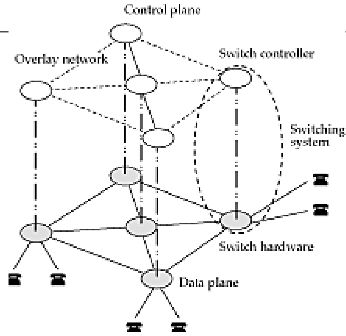


Figure 1.9 Switching system components.



Signaling

- Signaling/switch controllers are special purpose computers
- Linked by their own internal packet swithced network
  - *Common Channel Interoffice Signaling (CCIS) network*
- Messages on CCIS conform to *Signaling System 7 (SS7)*
- Switch controller is in the *control* plane
  - Does not touch voice samples
  - Circuit-setup (I.e. the signaling call) is what is routed.
  - Voice then follows route, and claims reserved resources.



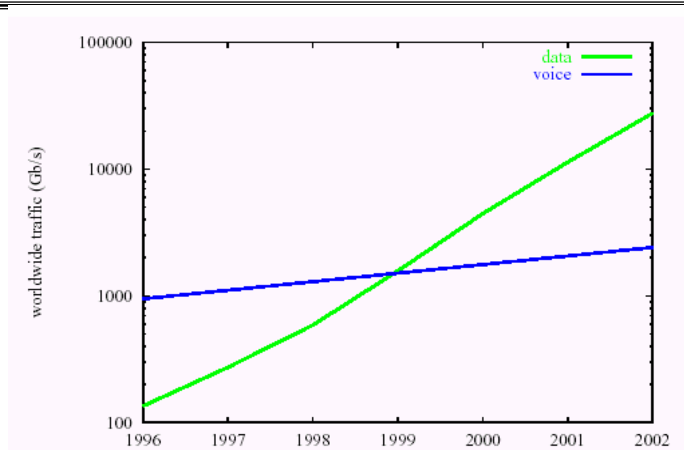
Signaling (contd)

- Manages the network
  - call routing (collect *dialstring* and forward call)
  - alarms (ring bell at receiver)
  - billing
  - directory lookup (for 800/888 calls)
- Switch controller: keep track of *state* of every endpoint
- *State transition diagram*

Features of PSTN

- ❑ Resource reservation aspects:
  - ❑ Resource reservation is coupled with path reservation
    - ❑ Connections need resources (same 64kbps)
    - ❑ Signaling to reserve resources and the path
  - ❑ Stable load
    - ❑ Network built for voice only.
    - ❑ Can predict pairwise load throughout the day
    - ❑ Can choose optimal routes in advance – high call completion rate
  - ❑ Simplified topology:
    - ❑ Very highly connected network
- ❑ Technology and economic aspects:
  - ❑ Extremely reliable switches (99.999% availability ~ 5mins downtime per year)
    - ❑ End-systems (phones) dumb because computation was non-existent in early 1900s
    - ❑ Advanced features provided by signaling network (911, 800 number, etc)
    - ❑ Regulated monopoly or oglopoly in many areas
- ❑ Organizational aspects:
  - ❑ Single organization controls large sections of the network (core, LEC)
  - ❑ Afford the scale economics to build expensive
  - ❑ International Telecommunications union
- ❑ Since the Plain old Telephone (POTS) in place and works why VoIP?

Trends: Data vs Voice Traffic



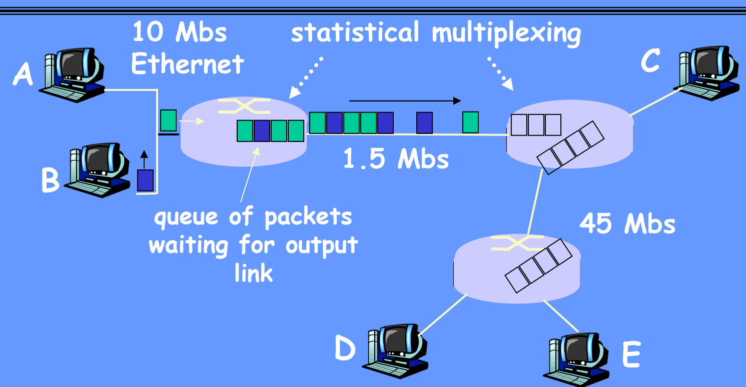
Service provider thinking - since we are building future networks for data, can we slowly junk the voice infrastructure and move over to VoIP? New services? Security with encryption, multi-media web, etc.

Internet

- Internet characteristics:
  - *Internet Protocol developed in 60's*
  - *Network of networks - no centralized control/management*
  - *Standards developed by open consensus IETF*
  - *Dynamic routing – adaptive to congestion and failure*
  - *Largely unregulated*
  - *Packet switching – bandwidth as needed*
  - *Best-effort service*

    (no performance guarantees)
  - *Packet-by-packet variations – packets can take different routes, arrive out of order, be dropped , etc.*

Packet Switching



- **Cost: <u>self-descriptive header</u> per-packet, buffering and <u>delays</u> due to statistical multiplexing at switches.**
- **Need to <u>either reserve resources</u> or dynamically detect and <u>adapt to overload</u> for <u>stability</u>**

Voice over IP Systems

- Voice over Internet Protocol (VoIP):
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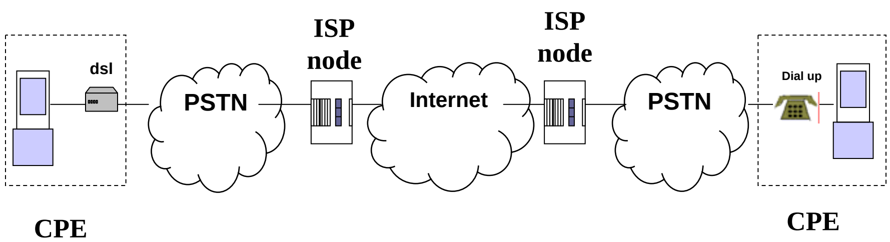
  - Internet Telephony (IP Telephony) more general concept then VoIP includes VoIP plus signaling/call control and telephony features (call waiting, 800 calls, three way calling, billing, etc.), interworking with existing telephone network
  - Will use VoIP and Internet Telephony interchangeably
  - Several categories/configurations for VoIP/Internet Telephony each with different motivation

VoIP Configurations/Categories

1. End-to-End over the Internet
   - Motivation – free/cheap phone calls
2. Enterprise VoIP
   - Small IP phone deployments, IP PBX, Call manager, etc.
   - Managed vs. Unmanaged
3. IP- switched local access phone carrier :
   - Motivation cheaper phone service – competitive local service long distance toll bypass services
   - VoIP wholesale, using equipment by vendors such as Cisco, Lucent, Avaya, etc.
4. VoIP through middleware
5. IP-switched service provider core
6. Telco Grade : Local and long distance service
   - Last mile delivery, total phone services, high dependability and availability.

1. End –to-End VoIP

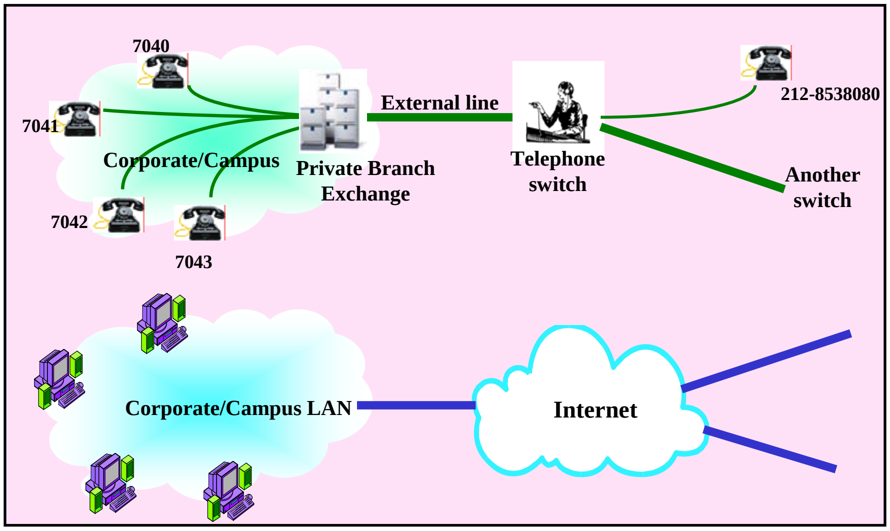
- **End-to-End VoIP telephony over the Internet (using PCs)**
  - Skype, Microsoft Netmeeting, Asterisk*, Marratech, etc.
  - No gateway with public switched telephone network (PSTN)
  - PSTN by-passed except for access (dialup, DSL)
  - An Internet application between users over unmanaged network
  - Not possible to detect or regulate
  - Business model?
  - Quality, features less than PSTN



2. Enterprise VoIP

- Motivation Integrating private voice & data network
  - No traditional economy of scale
    - Integrated network must be over-designed →
    - To provide acceptable voice & data service
  - $ managing 2 networks >> $ managing 1??
  - Which technology for common network?
    - In Enterprise data traffic > voice traffic then integrate on one
    - Managed or Unmanaged Enterprise core??
- Issues Quality and Reliability
  - Interest in this in the US → example General Motors (GM)

Enterprise Example



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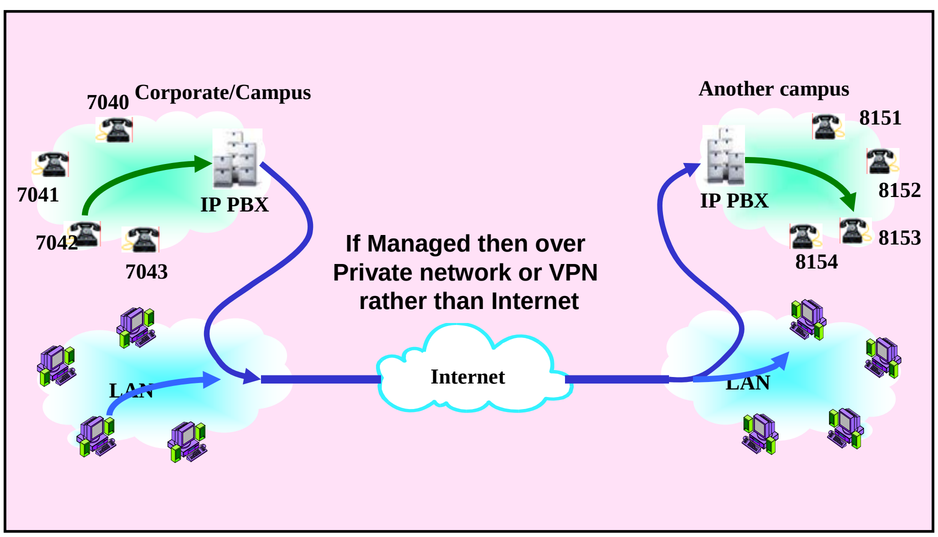
Enterprise IP Telephony



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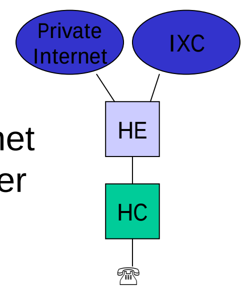
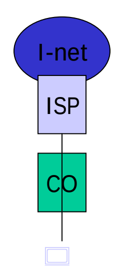
3. VoIP-Switched CLEC

- Competitive local phone provider
  - For example Cable TV
- Home controller (set-top box)
  - Could be a card in a PC, Also provides CATV
- Call Processing at Cable head-end
  - Provides features, 800,…
- Can control end-to-end delay
  - Local: Private internet, not The Internet
  - Long Distance: Inter-Exchange Carrier
- Example Comcast in U.S.



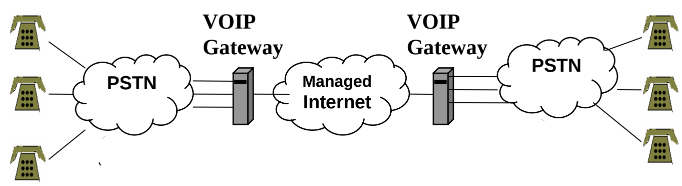
4. VoIP thru Middleware

- PC users only, with IP interface
  - Excludes non-PC market
  - Subscriber calls ISP thru local phone company (LEC)
- Artificial economy - don't pay:
  - LD access charge, LD excise tax (voice & data)
- ISP does Call Processing at its gateway
  - Features, 800, etc?
  - Signaling?
  - Charge users?, Include in monthly flat rate?
  - AOL is offerering



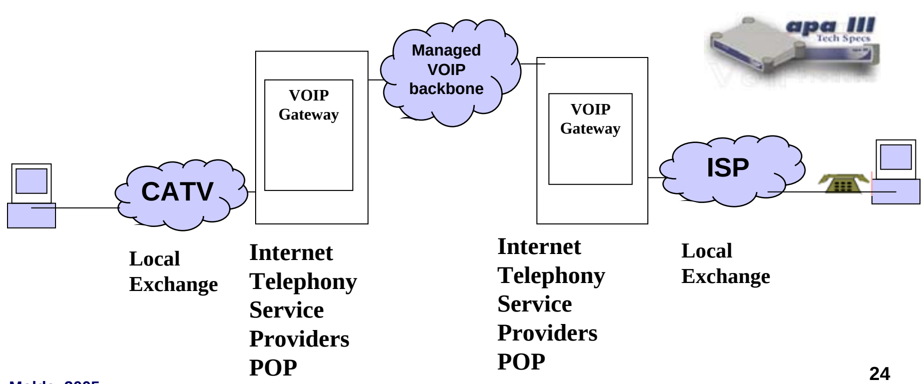
5. IP-Switched IXC Core

- Conventional telephone
  - Conventional local service, call processing, SS7 etc.
  - Subscriber calls an IXC long distance carrier
- True economy - subscriber pays:
  - LD access charge, LD excise tax
- IXC switches calls internally by IP
  - Private internet, DS0-IP interface at IXC boundary
  - LEC at both ends
- No advantage for user at PC
- Quest is doing this in the U.S., Telus in Canada



VoIP-Switched End-to-End Telco Grade Carrier

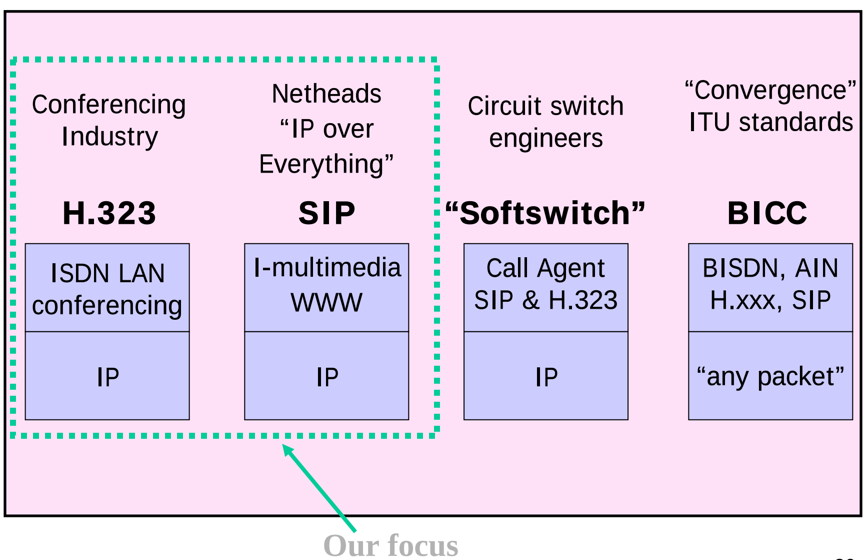
- Connect VoIP phones and PCs
  - Can't push any cost into ILEC
  - CATV leverage (bundling)
  - Bypass LD access charge
    - Avoid Universal Service in U.S, but pay all excise taxes
- Economics? Features? Quality?



IP Telephony - Regulatory Issues

- Bypassing the "Access Charge"
  - Local exchange carriers have some legitimate cost
  - Long Distance subsidy of Universal Service
  - Inequity → "bypass" industry
    - For enterprise, not residential
- Why no charge for Internet access?
  - IXC could "looks like" an ISP
  - Significant cost/price advantage
- VoIP has variety of implementations to go along with different configurations/categories
  - Standards bodies (IETF, ITU, ETSI, etc.)

VoIP Camps



VoIP Protocols

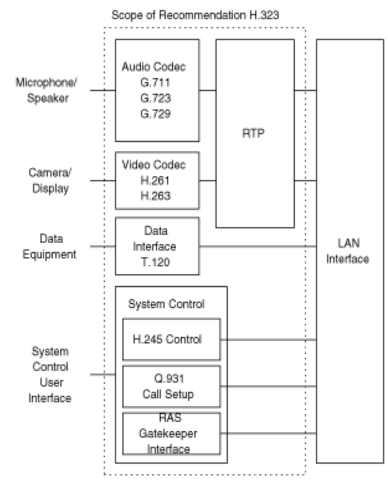
- No one standard for Signaling:
  - SIP ,H.323, MGCP
- RTP for transport
  - RTP (Real-Time Transport Protocol)
  - RTP is a UDP stream with no intelligence for QoS or resource reservation
  - Contains a packet number for detection of packet loss and re-sequencing of out of order packets.
  - Unidirectional : two streams in any call

How they fit in: The ISO Model

ISO Model Layer	Protocol or Standard
Presentation	VoIP CODECS
Session	H.323 & SIP&MGCP
Transport	RTP / UDP / TCP
Network	IP
Data Link	ATM, FR, PPP, Ethernet, SONET, etc.

VoIP Signaling

- Most VoIP implementations use H.323 protocol
  - Same protocol that is used for IP video.
- H.323 Multimedia Standard developed by ITU
  - H.225 RAS - Registration, Admission, Status
  - Q.931 - Call Signaling (Setup & Termination)
  - H.245 - Call Control (Preferences, Flow Control, etc.)
  - Lots of G.7XX CODECS for audio



H.323 Entities

- Terminals
  - An endpoint that supports 2-way streaming with another H.323 terminal or gateway
  - Originates and terminates calls
  - Includes videoconferencing stations, hard phones, & soft phones
- Gateways (Optional but really useful)
  - Device that connects H.323 voice network to non-H.323 voice network (SIP or PSTN)
  - Allows H.323 terminals to communication with non-H.323 terminals
- Gatekeepers (Also optional)
  - Provides address translation (H.323 & E.164 to IP)
  - Admission control for H.323 terminals and gateways
  - Manage bandwidth allocation
- MCU (multipoint control unit)
  - MC – multipoint controller
    - Routes call and control signaling to ensure endpoint compatibility
  - MP – multipoint processor
    - Switches, mixes and processes vice and video streams to conferencing equipment

Call Setup using H.225 RAS

- Registration, Admission and Status (RAS), is responsible for registration, admission, and disengaging procedures between H.323 Gatekeeper and Gateway.
- Discovery: GRQ, GCF, GCR
  - Unicast or multicast Discovery
- Registration by terminals, Gateways & MCUs using H.323 ID
  - RRQ Registration Request
  - RCF Registration Confirm
  - RRJ Registration Reject
  - URQ Un-registration Request
  - URF Un-registration Confirm
  - URJ Un-registration Confirm

H.323 – H.225 RAS Messages

- LRQ – location request
  - Gatekeeper A requests contact information from directory gatekeeper.
- LCF – location confirm
  - Gatekeeper B returns IP address of destination gateway to gatekeeper A.
- ARQ – admission request
  - Gateway A requests admission to make a call.
- ACF – admission confirm
  - Gatekeeper A responds with IP address of destination gateway.

Signaling using Q.931 messages

- Q.931 is a signaling protocol used to setup, manage, and terminate H.323 connections between endpoints.
  - ARQ, ACF, ARJ Admission messages
  - LRQ, LCF, LRJ Location Request messages
  - IRQ, IRR, IACK, INAK Status messages
  - BRQ, BCF, BRJ, RAI, RAC Bandwdith messages
- Release Complete
  - H.225 (Q.931) call has been released, signaling channel is now open
- Status
  - Sent when unknown call signaling message or a status inquiry message is received
- Status Inquiry
  - Requests a call's status

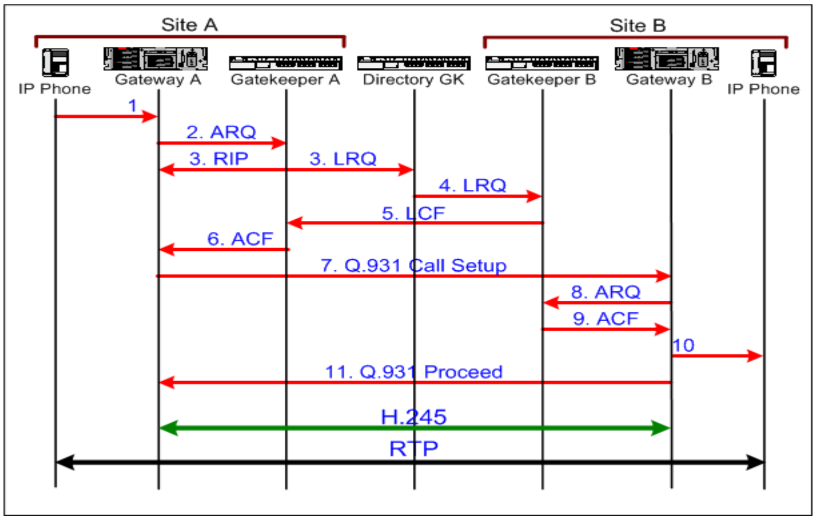
H.323 – H.245 Messages

- Establishes logical channels for transmission of H.323 data
- Master/Slave Determination
  - Determines which terminal will be master which will be slave in the call
- Terminal Capability Set
  - Contains information on a terminal's ability to send and receive multimedia streams
- Open Logical Channel
  - Opens logical channel for transport of multimedia data
- Close Logic Channel
  - Closes the logical channel between two endpoints

H.323 – H.245 Messages

- Request Mode
  - Receive terminal requests type of transportation from a transmit terminal
  - Types of Modes:
    - Video
    - Audio
    - Data
    - Encryption
- Send Terminal Capacity Set
  - Instructs far-end terminal to send transmit and receive capabilities
- End Session Command
  - Indicates the end of the H.245 session

H.323 Call Setup via Gatekeepers



VoIP Signaling Protocols

- Session InitiationProtocol (SIP)
- IETF-standardized *peer-to-peer* signaling protocol (RFC 2543):
  - Locate user given email-style address
  - Setup session (call)
  - (Re)-negotiate call parameters
  - Manual and automatic forwarding
  - Personal mobility: different terminal, same identifier
  - Call center: reach first (load distribution) or reach all (department conference)
  - Terminate and transfer calls
- Light-weight generic signaling protocol

SIP Addresses Food Chain

yellow pages	"president of the United States"	
	↓ *WWW search engines*	
common names	"Bill Clinton, Whitehouse"	
	↓ *directory services*	
host-independent	president@whitehouse.gov	
	↙ *SIP*	↘ *SIP*
host-specific	sip:bubba@oval.eop.gov	sip:+1-202-456-1111@net2ph
	↓ *DNS*	
IP address	198.137.241.30	

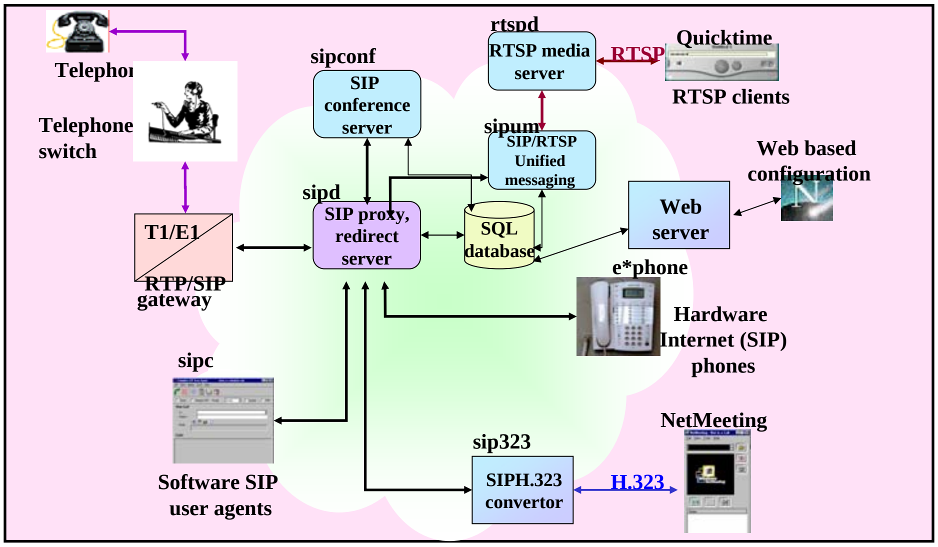
SIP Architecture

- SIP uses client/server architecture
  - Client - originates message
  - Server - responds to or forwards message
- Elements:
  - SIP User Agents (SIP Phones)
    - User Agent Client (UAC)
      - Initiates SIP requests
    - User Agent Server (UAS)
      - Returns SIP responses
  - SIP Servers (Proxy or Redirect - used to locate SIP users or to forward messages.)
    - Can be stateless or stateful
  - SIP Gateways:
    - To PSTN for telephony interworking
    - To H.323 for IP Telephony interworking

SIP Messages

- INVITE: Initiates call
  - Session description included in message body
- Re-INVITEs used to change session state
- ACK confirms session establishment, can only be used with INVITE
- BYE terminates a session/call (hanging up)
- CANCEL cancels a pending invite and ringing
- REGISTER: binds a permanent address to current location, may convey user data
- OPTIONS: capability inquiry, that is determine features supported by the other side of the call

SIP-based Architecture

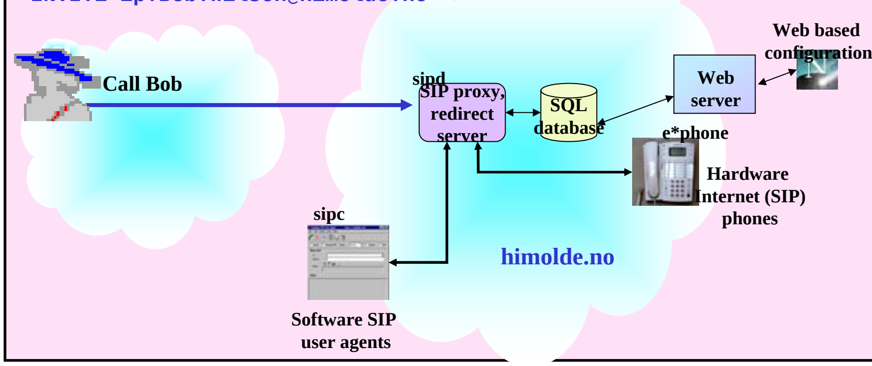


Example Call

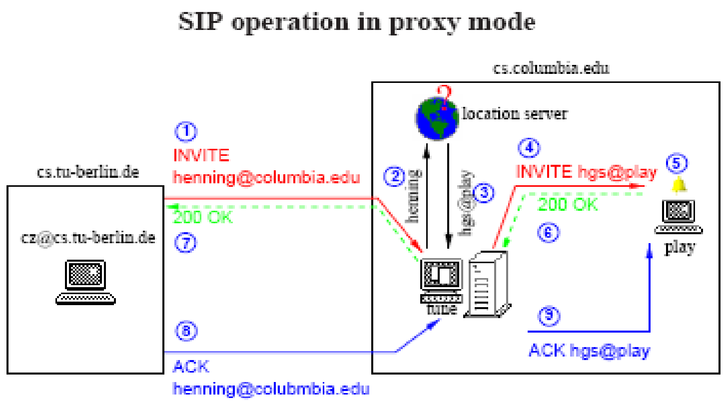
- Bob signs up for the service from the web as "bob@himolde.no"
- He registers from multiple phones
- Alice tries to reach Bob

`INVITE ip:Bob.Wilson@himolde.no`

- sipd canonicalizes the destination to `sip:bob@himolde.no`
- sipd rings both e*phone and sipc
- Bob accepts the call from sipc and starts talking



Another Example



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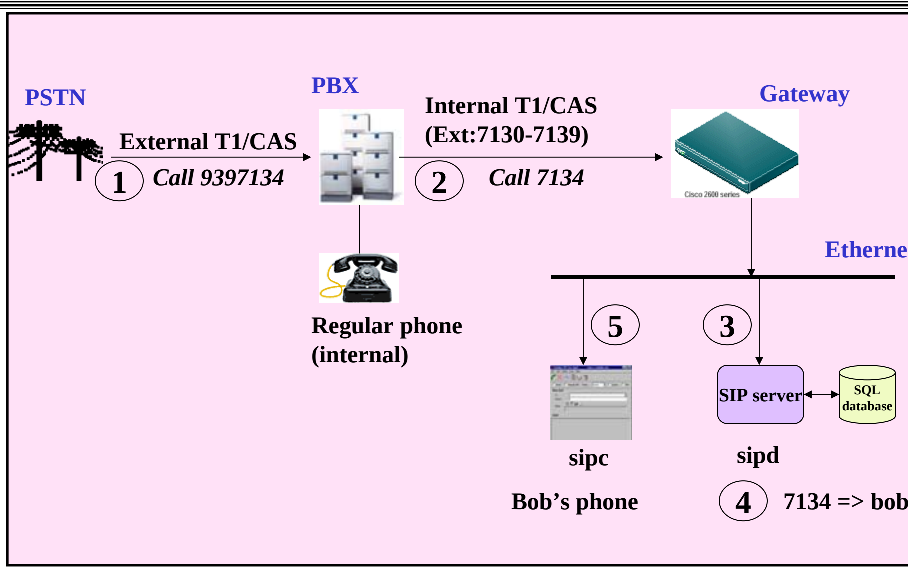
PSTN to IP Call



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Authentication & Encryption

- SIP supports a variety of approaches:
  - end to end encryption
  - hop by hop encryption
- Proxies can require authentication:
  - Responds to **INVITE**s with **407 Proxy-Authentication Required**
  - Client **INVITE**s again with **Proxy-Authorization** header.
  - Stacked!!
- SIP Users can require authentication:
  - Responds to **INVITE**s with **401 Unauthorized**
  - Client **INVITE**s with **Authorization** header

SIP vs H.323

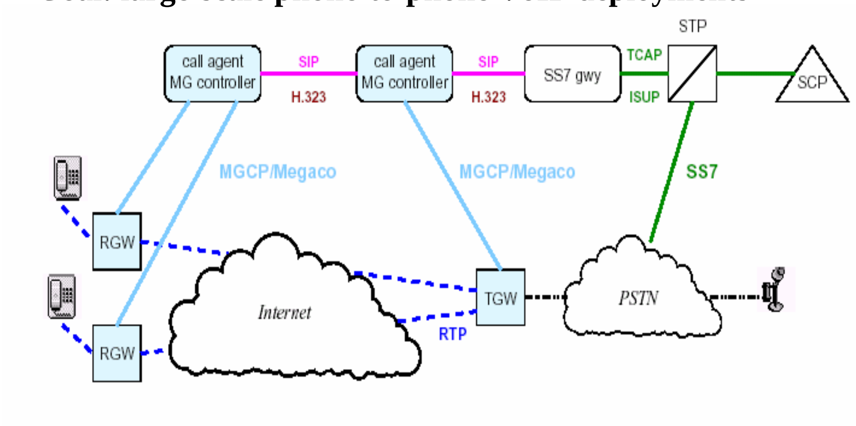
- What makes SIP > H.323?
  - Simpler & faster call setup
  - Multicast experience (MBONE)
  - Easier to add third-party services
  - SIP uses UDP for setup gives better timing control then TCP used by H.323
  - SIP uses an HTTP-like syntax much easier to write/debug.
- What makes H.323 > SIP?
  - Separates signaling & media channels
  - Conference control
  - Compatible with Q.931
  - More stable and encompasses broader system (ITU standard)
  - Embedded base of H.323 telephones
- European Telecommunications Standards Institute Tiphon group working on SIP ←→ H.323 gateway/portal
  http://www.etsi.org

MGCP and Megaco

- Media Gateway Controller Protocol (RFC 2705)
- Controlling Telephony Gateways from *external* call control elements called media gateway controllers (MGC) or call agents
  - Gateways: Eg: RGW : physical interfaces between VoIP network and residences
  - MGCP is a master/slave protocol where all the smarts resides in the gateway controller and not the gateway.
  - Goal: Scalable gateways between VoIP telephony and PSTN
    - Simplify PSTN- VoIP interworking
    - Carriers don't want H.323's direct-call model
    - IP signaling must interface to SS7:
- Successor to MGCP: H.248/Megaco

MGCP Architecture

Goal: large-scale phone-to-phone VoIP deployments



RGW: Residential Gateway
TGW: Trunk Gateway

SoftSwitch

- MGCP signaling controller
  - SW that controls calls to the PSTN
  - Billing
  - Assumes/controls Indirect calls
- Evolved - may or may not include:
  - Gatekeeper functionality
  - Directories & Feature SW
  - Conference control → VoIP, PSTN mix
  - More later (time permitting)
- Be careful of this word

MGCP Analysis

- What makes MGCP > H.323v1?
  - Better with complex calls & carrier functions
    - Network services, Billing
  - H.323v1 uses TCP (could be changed)
- H.323v2 > MGCP?
  - Better with embedded base
    - PCs, IP telephones
  - Better with intelligent end-points
- It's a mess

Comparison of Packet vs. Circuit Switching

	Circuit	Packet
Call Setup	Database / SS 7 Overlay	H.323 & SIP
Communications Channel	Dedicated	Shared
Addressing	International Numbering Plan	IPv4 & IPv6

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VoIP Issues and QoS

- VoIP needs to replicate POTs features?
  - 911, Power to phone, 800 numbers, etc
- Quality of Service issues
  - Voice codec choice
  - Echo
  - Delay
  - Delay variation (jitter)
  - Packet loss
  - Reliability/Availability
- Lots of Hype for VoIP the last 7 years or so – reality is small percentage of voice traffic is VoIP
  - Forrester Group predicts 10% of residential voice service VoIP by 2010
  - Hope for VoIP is adoption by business – leading to buildout of service and lower cost

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