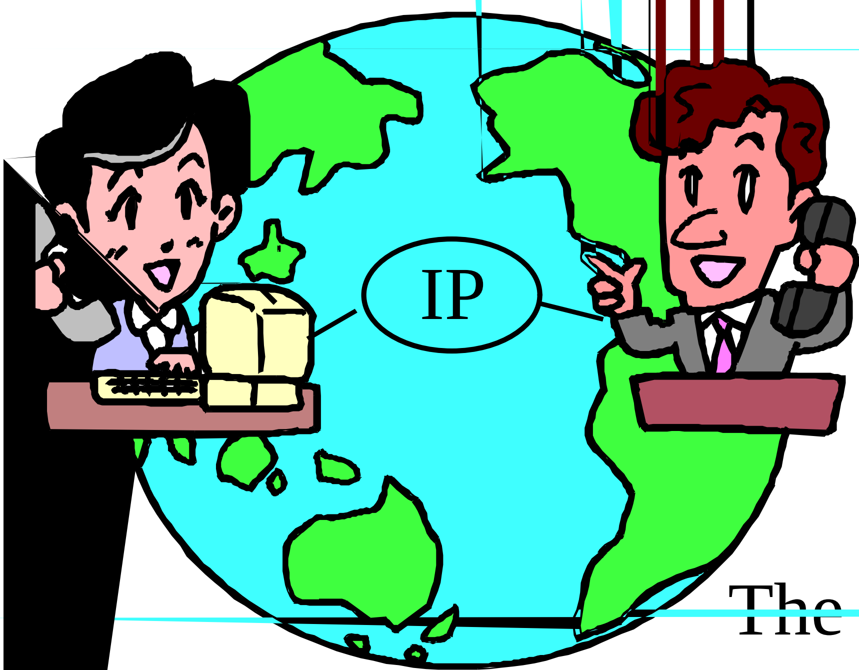
Voice over IP



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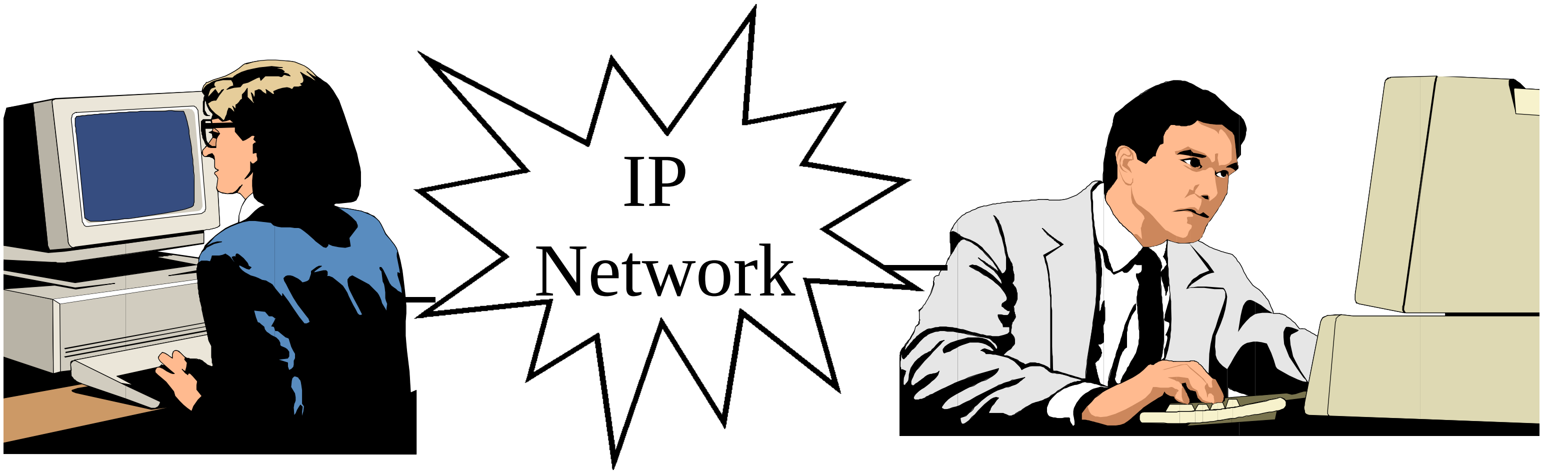
- Sample Products and Services
- 13 Technical Issues
- 4 Other Issues
- H.323 Standard
- Session Initiation Protocol (SIP)

Scenario 1: PC to PC



Need a PC with sound card

IP Telephony software: Cuseeme, Internet Phone, ..

Video optional

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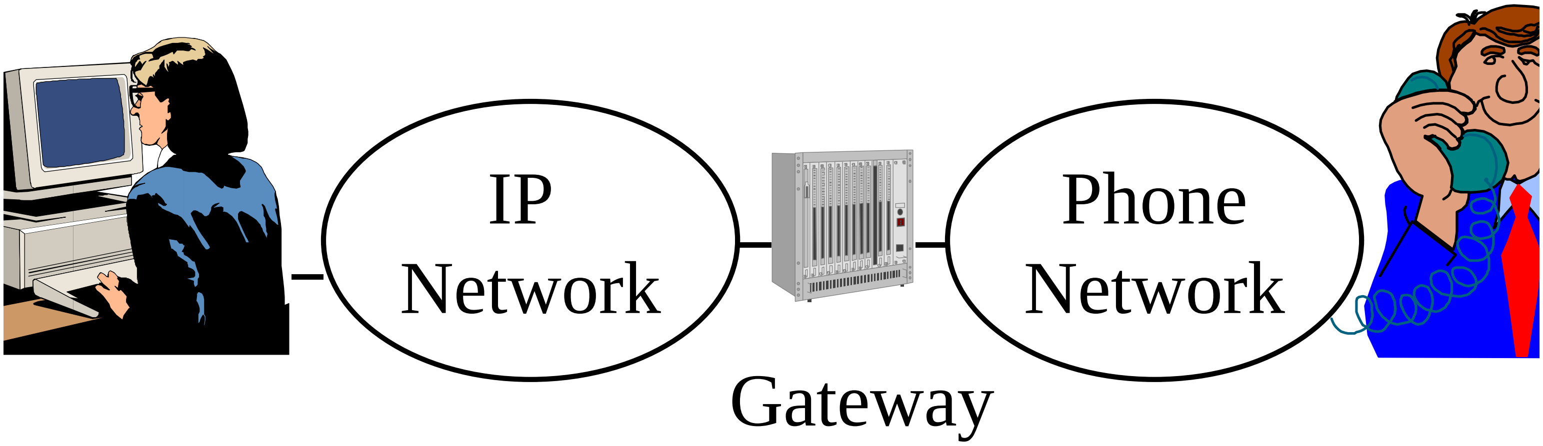
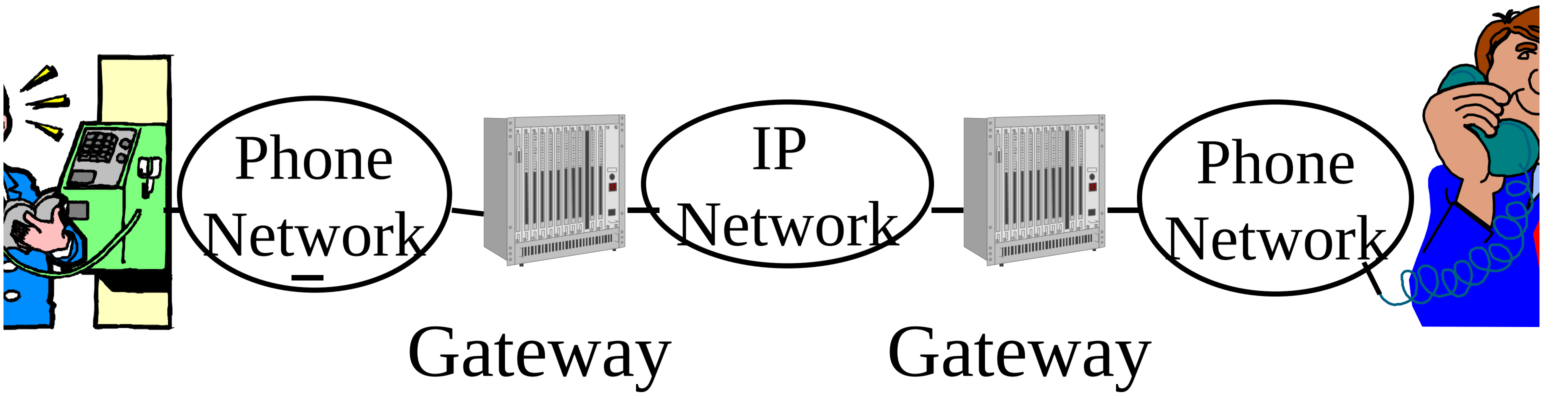
Scenario 2: PC to Phone



Need a gateway that connects IP network to phone network (Router to PBX)

Scenario 3: Phone to Phone



- Need more gateways that connect IP network to pho
networks
- The IP network could be dedicated intra-net or the
Internet.
- The phone networks could be intra-company PBXs o
he carrier switches

Sample Products

/ocalTec Internet Phone: PC to PC.

Microsoft NetMeeting: PC to PC. Free.

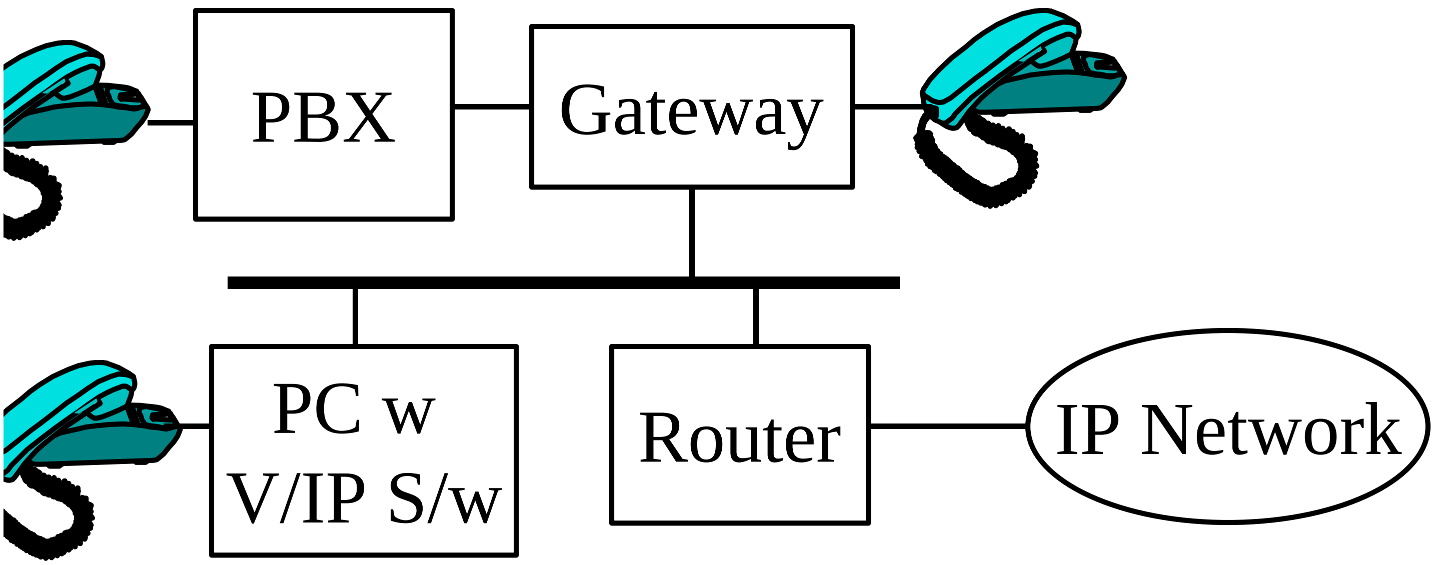
nternet PhoneJACK: ISA card to connect a standar phone to PC. Works with NetMeeting, InternetPhon etc. Provides compression.

nternet LineJACK: Single-line gateway.

Micom V/IP Family:

- Analog and digital voice interface cards
- PC and/or gateway

Products (Cont)



- Features:
  - Compression
  - Phone number to IP address translation.
  - Supports RSVP.
  - Limits number of calls.

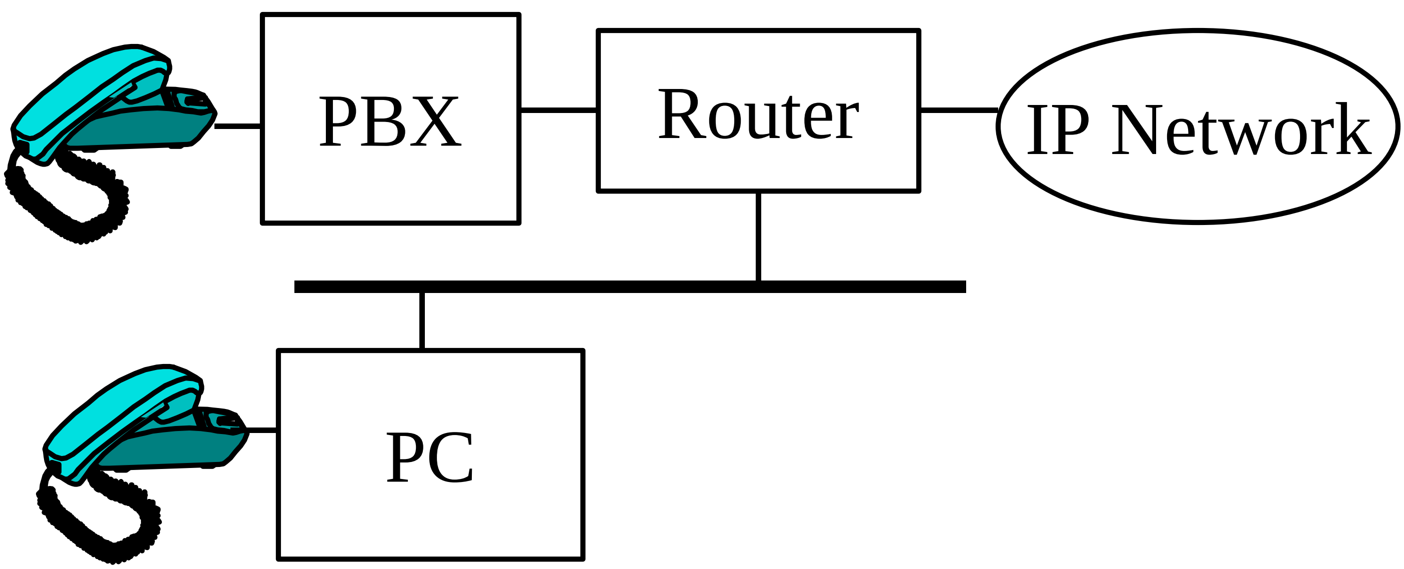
Products (Cont)

- VocalTec Internet Telephony Gateway:
  - Similar to Micom V/IP
  - Interactive voice response system for problem reporting
  - Allows WWW plug in
  - Can monitor other gateways and use alternate routes including PSTN
  - Sold to Telecom Finland. New Zealand Telecom.
- Lucent's Internet Telephony Server: Gateway|
- Lucent PathStar Access Server

Products (Cont)

CISCO 2600 Routers: Voice interface cards (VICs) Reduces one hop.

Baynetworks, 3COM, and other router vendors have announced product plans



Sample Services

- DT Corporation offers Net2Phone, Carrier2Phone, Phone2Phone services.
- Global Exchange Carrier offers international calls using VocalTec InternetPhone s/w and gateways
- Qwest offers 7.5¢/min VOIP Q.talk service in 16 cities.
- TXC provides infrastructure and management to Internet Telephone Service Providers (ITSPs)'
- America On-line offers 9¢/min service.
- AT&T announced 7.5¢/min VOIP trials in 9 US citi

Services (Cont)

)ther trials: USA Global link, Delta 3, WorldCom, MCI, U.S. West, Bell Atlantic, Sprint, AT&T/Japan KDD/Japan, Dacom/Korea, Deutsche Telekom in Germany, France Telecom, Telecom Finland, and New Zealand Telecom.

Level 3 is building a nation wide IP network for elephony.

Bell Canada has formed 'Emergis' division.

Bellcore has formed 'Soliant Internet Systems' unit

Bell Labs has formed 'Elemedia' division

Technical Issues

Large Delay

- Normal Phone: 10 ms/kmile ⇒ 30 ms coast-to-coast
- G.729: 10 ms to serialize the frame + 5 ms look ahead + 10 ms computation = 25 ms one way algorithmic delay
- G.723.1 = 100 ms one-way algorithmic delay
- Jitter buffer = 40-60 ms
- Poor implementations ⇒ 400 ms in the PC
- In a survey, 77% users found delay unacceptable.

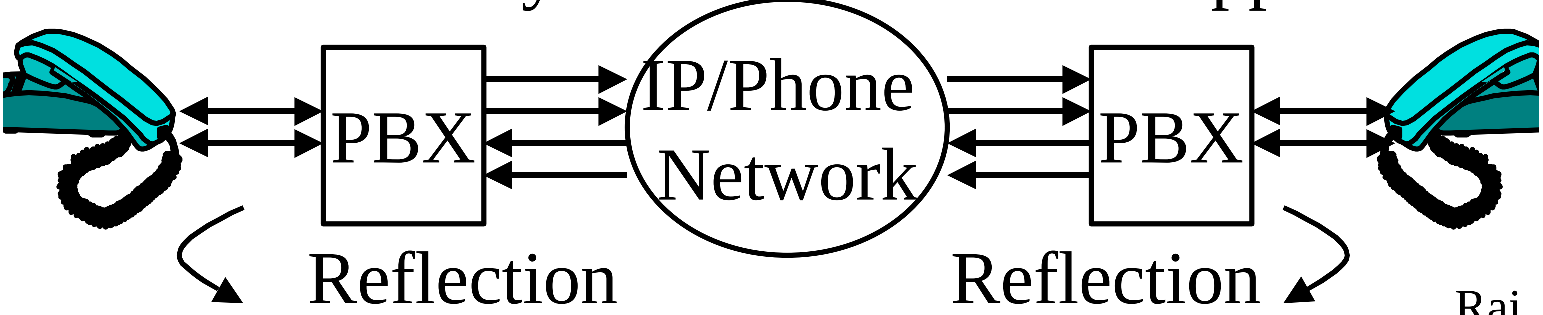
Technical Issues (Cont)

Delay Jitter: Need priority for voice packets. Shorter packets? IP precedence (TOS) field.

Frame length: 9 kB at 64 kbps = 1.125 s Smaller MTU ⇒ Fragment large packets

Lost Packets: Replace lost packets by silence, extrapolate previous waveform

Echo cancellation: 2-wire to 4-wire. Some FR and IP systems include echo suppressors.



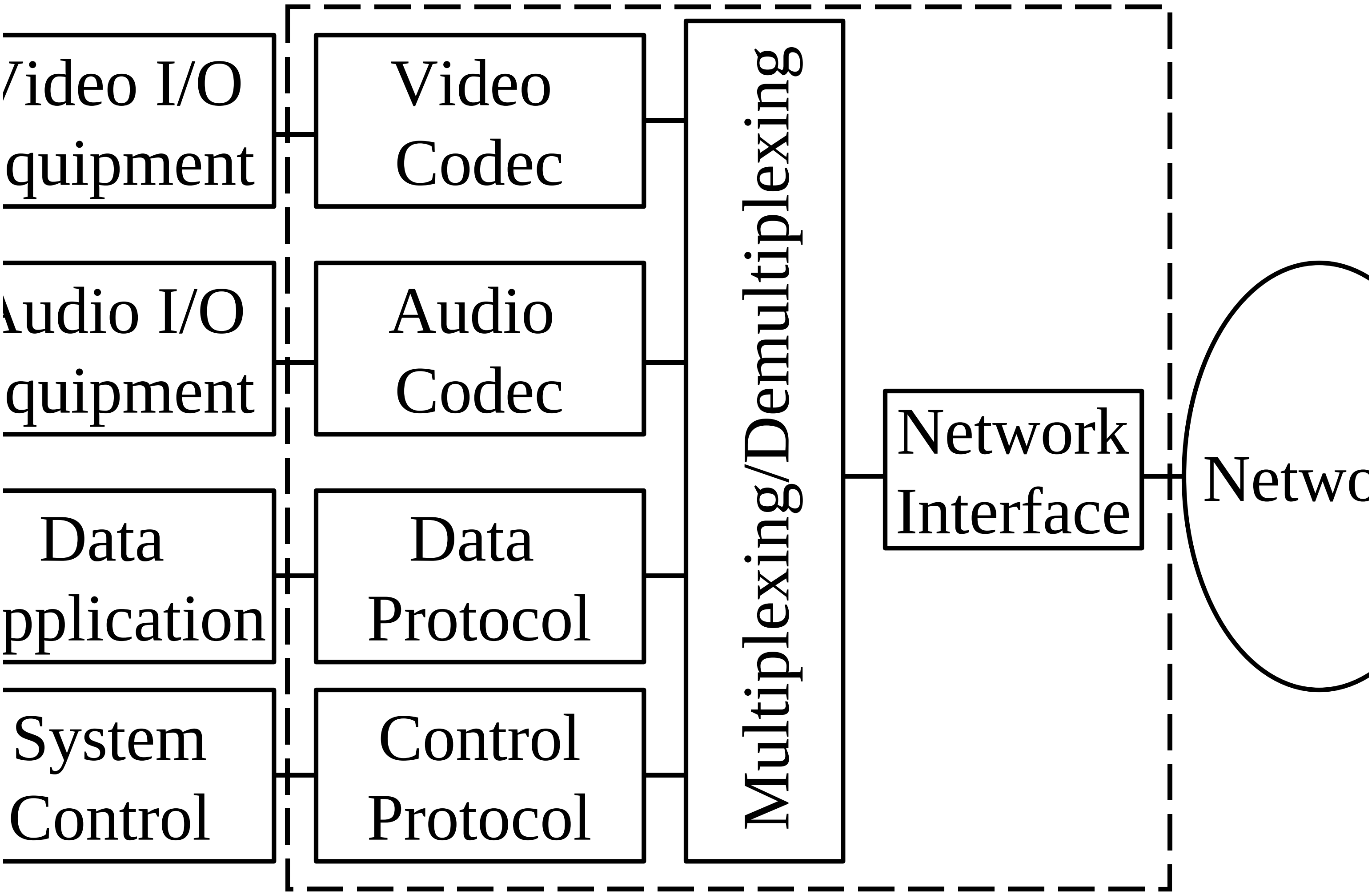
Technical Issues (Cont)

- Silence suppression
- Address translation: Phone # to IP. Directory servers
- Telephony signaling: Different PBXs may use differen signaling methods.
- Bandwidth Reservations: Need RSVP.
- Multiplexing: Subchannel multiplexing
  ⇒ Multiple voice calls in one packet.
- Security: Firewalls may not allow incoming IP traf
- Insecurity of internet
- Voice compression: Load reduction

Other Issues

- Per-minute distance-sensitive charge vs flat time-insensitive distance-insensitive charge
- Video requires a bulk of bits but costs little. Voice is expensive. On IP, bits are bits.
- National regulations and government monopolies ⇒ Many countries forbid voice over IP
- In Hungary, Portugal, etc., it is illegal to access a we site with VOIP s/w. In USA, Association of Telecommunications Carriers (ACTA) petitioned FC to levy universal access charges in ISPs
- Modem traffic can't get more than 2400 bps.

'elephony/Conferencing System



Conferencing Standards

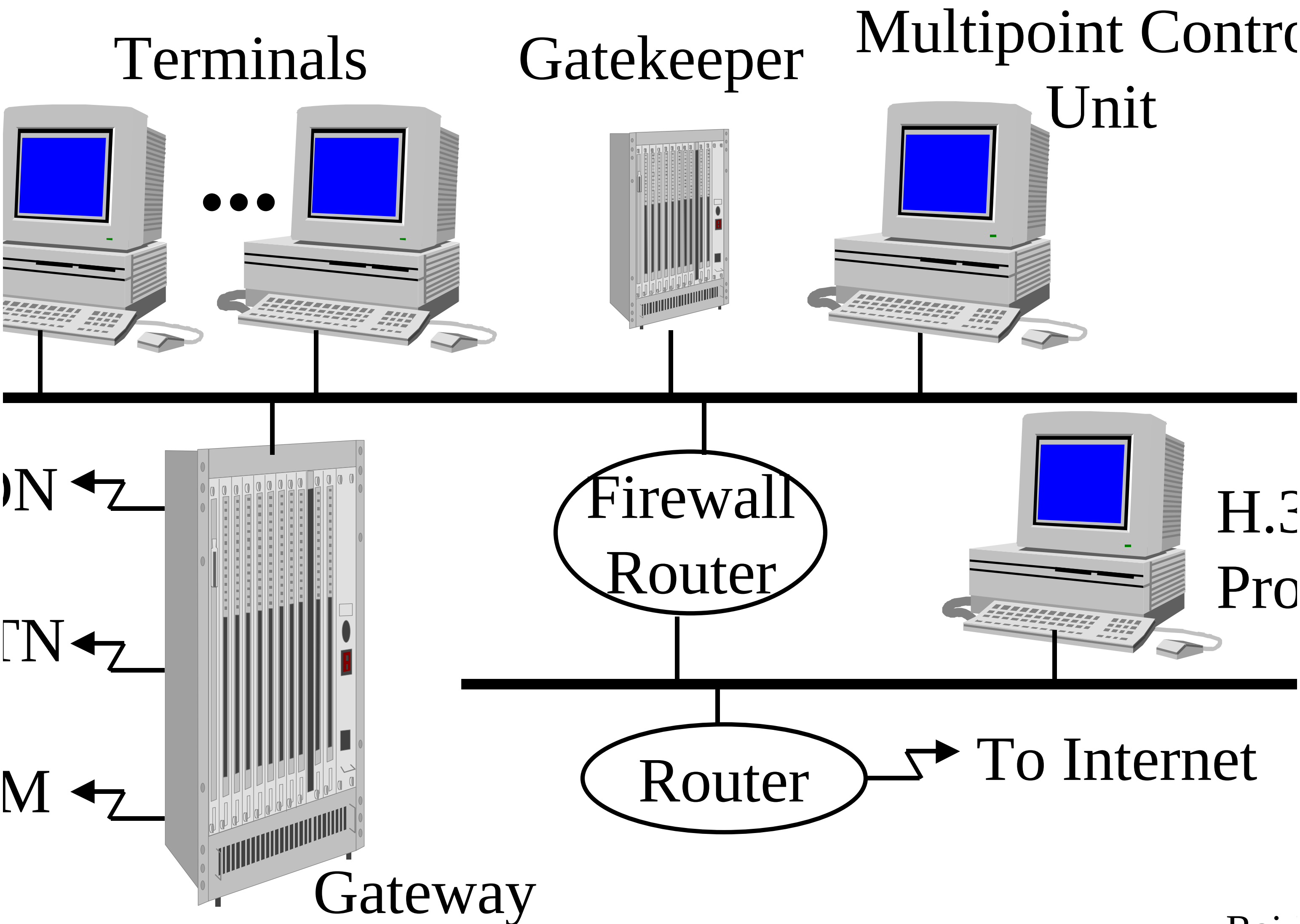
etwork	ISDN	ATM	PSTN	LAN	POTs
onf. Std.	H.320	H.321	H.322	H.323 V1/V2	H.324
ear	1990	1995	1995	1996/1998	1996
udio odec	G.711, G.722, G.728	G.711, G.722, G.728	G.711, G.722, G.728	G.711, G.722, G.723.1, G.728, G.729	G.723.1, G.729
udio Rates ps	64, 48-64	64, 48-64, 16	64, 48-64, 16	64, 48-64, 16, 8, 5.3/6.3	8, 5.3/6.
ideo odec	H.261	H.261, H.263	H.261, H.263	H.261 H.263	H.261 H.263
ata Sharing	T.120	T.120	T.120	T.120	T.120
ontrol	H.230, H.242	H.242	H.242, H.230	H.245	H.245
ultiplexing	H.221	H.221	H.221	H.225.0	H.223
gnaling	Q.931	Q.931	Q.931	Q.931	-

H.323 Protocols

- Multimedia over LANs
- Provides component descriptions, signaling procedures, call control, system control, audio/video codecs, data protocols

<table>
<tr><td>Video</td><td>Audio</td><td colspan="4">Control and Management</td><td>Da</td></tr>
<tr><td>H.261
H.263</td><td>G.711, G.722,
G.723.1, G.728,
G.729</td><td rowspan="2">RTCP</td><td>H.225.0
RAS</td><td>H.225.0
Signaling</td><td>H.245
Control</td><td>T.1</td></tr>
<tr><td colspan="2">RTP</td><td colspan="3">X.224 Class 0</td><td>T.1</td></tr>
<tr><td colspan="3">UDP</td><td colspan="3">TCP</td><td rowspan="2">T.1</td></tr>
<tr><td colspan="6">Network (IP)</td></tr>
<tr><td colspan="7">Datalink (IEEE 802.3)</td></tr>
</table>

H.323 Components



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H.323 Gatekeepers

- Provide call control services to registered end points
- One gatekeeper can serve multiple LANs
- Address translation (LAN-IP)
- Admission Control: Authorization
- Bandwidth management
- (Limit number of calls on the LAN)
- Zone Management: Serve all registered users within its zone of control
- Forward unanswered calls
- May optionally handle Q.931 call control

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Session Initiation Protocol (SIP)

- Application level signaling protocol
- Allows creating, modifying, terminating sessions with one or more participants
- Carries session descriptions (media types) for user capabilities negotiation
- Supports user location, call setup, call transfers
- Supports mobility by proxying and redirection
- Allows multipoint control unit (MCU) or fully meshed interconnections
- Gateways can use SIP to setup calls between them

SIP (Cont)

SIP works in conjunction with other IP protocols for multimedia:

- RSVP for reserving network resources
- RTP/RTCP/RTSP for transporting real-time data
- Session Announcement Protocol (SAP) for advertising multimedia session
- Session description protocol (SDP) for describing multimedia session

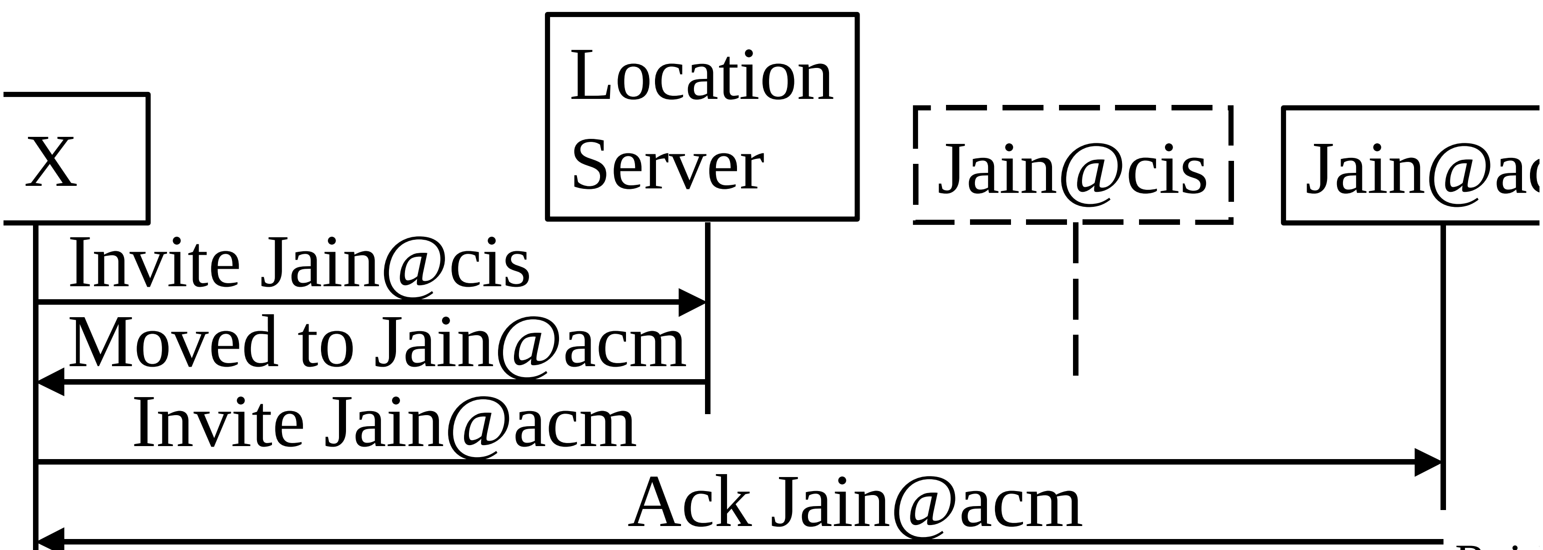
Can also be used to determine whether party can be reached via H.323, find H.245 gateway/user address

SIP (Cont)

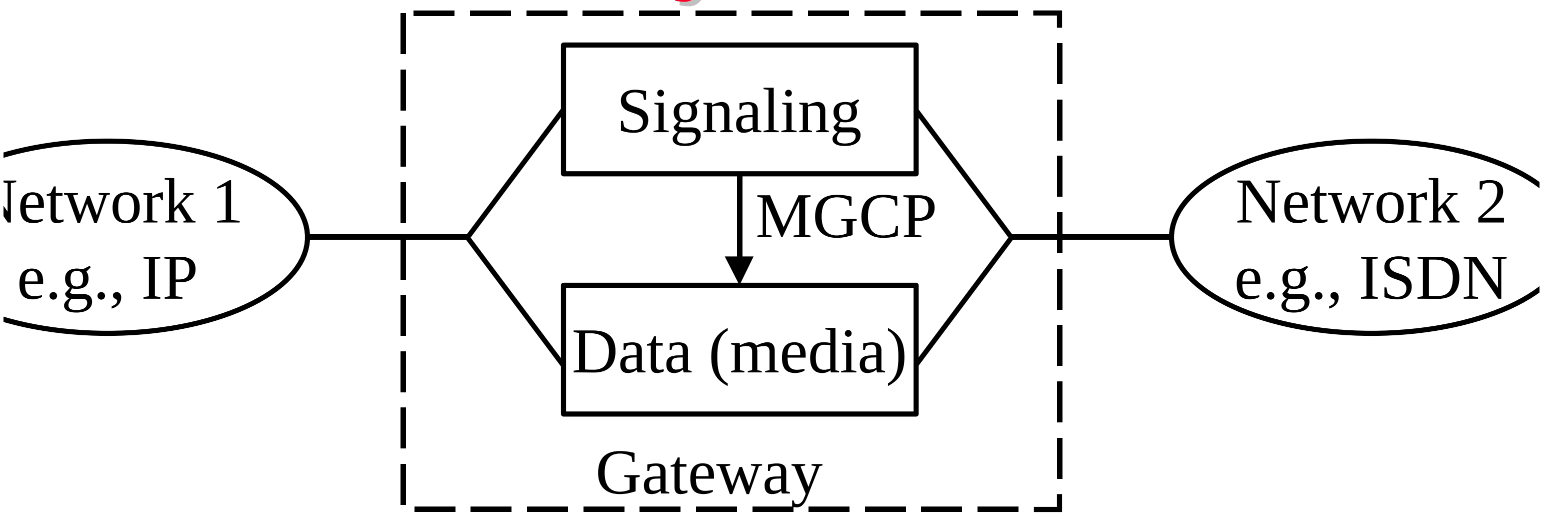
- ;IP is text based (similar to HTTP)
  ⇒ SIP messages can be easily generated by humans
  CGI, Perl, or Java programs.
- ;IP Uniform Resource Locators (URLs):
  ;imilar to email URLs
  ;ip:jain@cis.ohio-state.edu
  ;ip:+1-614-292-3989:123@osu.edu?subject=lecture
- ;IP messages are sent to SIP server at the specified
  ıddress
- ;IP can use UDP or TCP

Locating using SIP

- Allows locating a callee at different locations
- Callee registers different locations with SIP Server
- Servers can also use finger, rwhois, ldap to find a ca
- SIP Messages: Ack, Bye, Invite, Register, Redirecti



Media Gateway Control Protoc



- Gateway = Signaling Fns + Media Transfer Fns
- Call Agents: Signaling functions ⇒ Intelligent
  ⇒ More complex ⇒ Fewer
  ⇒ Control multiple media gateways ⇒ Need MGCP
- MGCP =Simple Gateway Control Protocol (SGCP)
  + Internet Protocol Device Control (IPDC)

MGCP Commands

Endpoint Configuration (EPCF): Specify coding

Notification Request (RQNT): Watch for event

Notify (NTFY): Used by gateway to inform Call age

Create Connection (CRCX)

Modify Connection (MDCX)

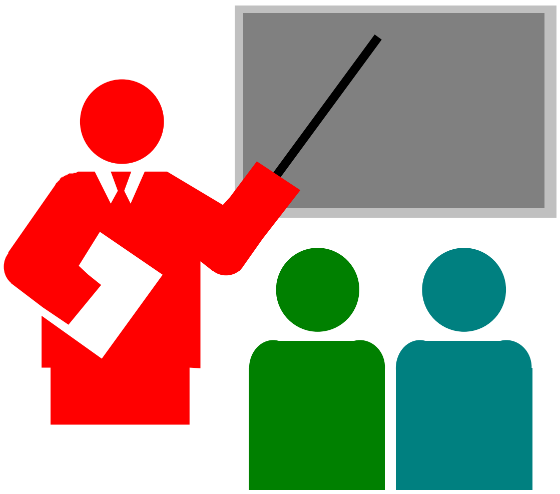
Delete Connection (DLCX)

Audit Endpoint (AUEP): Give me status

Audit Connection (AUCX)

Restart in Progress (RSIP): Used by gateway to
ndicate initialization/shutdown of endpoints/gatewa

Summary



- /oice over IP products and services are being rolled
- deal for computer-based communications
- P needs QoS for acceptable quality
- \ number of working group at IETF are working on
- I.323 provides interoperability

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References

See
http://www.cis.ohio-state.edu/~jain/refs/ref_voip.htm
for a detailed list of references.