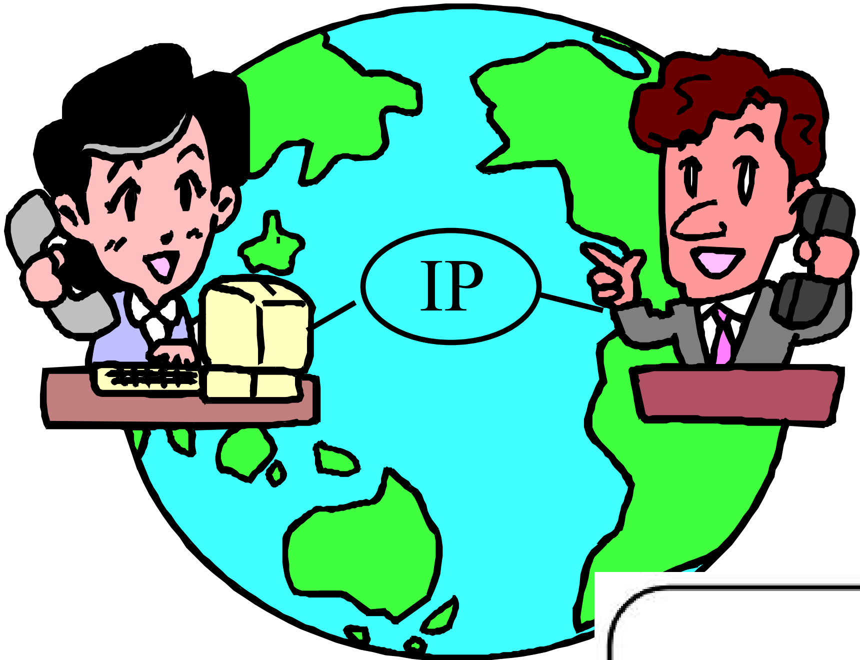
Voice over IP



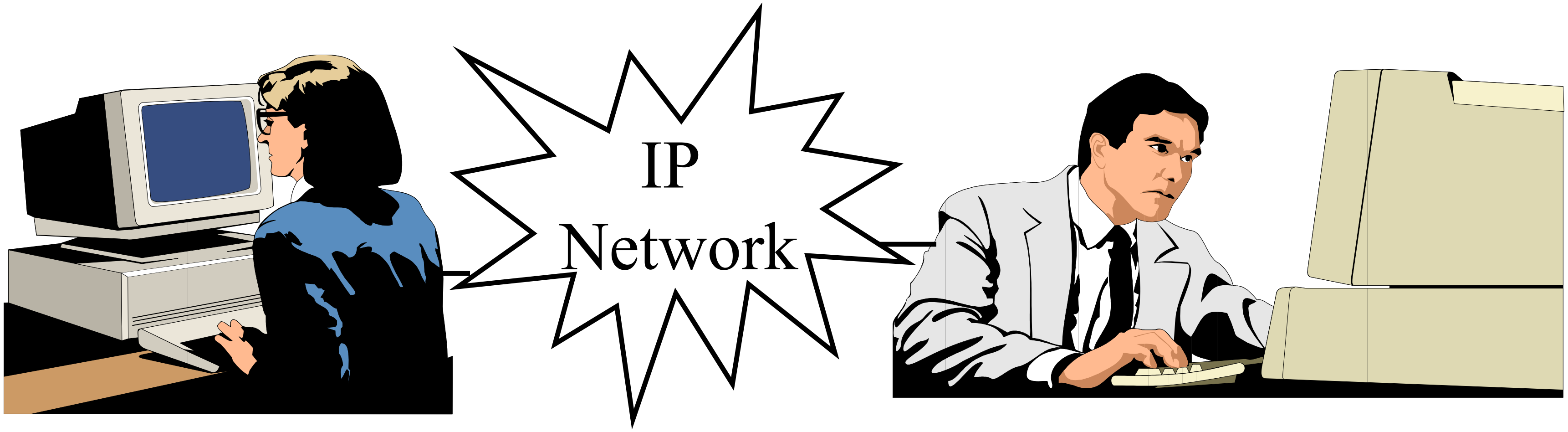
Raj Jain

ht

Raj Jain is now at
Washington University in Saint Louis
Jain@cse.wustl.edu
http://www.cse.wustl.edu/~jain/

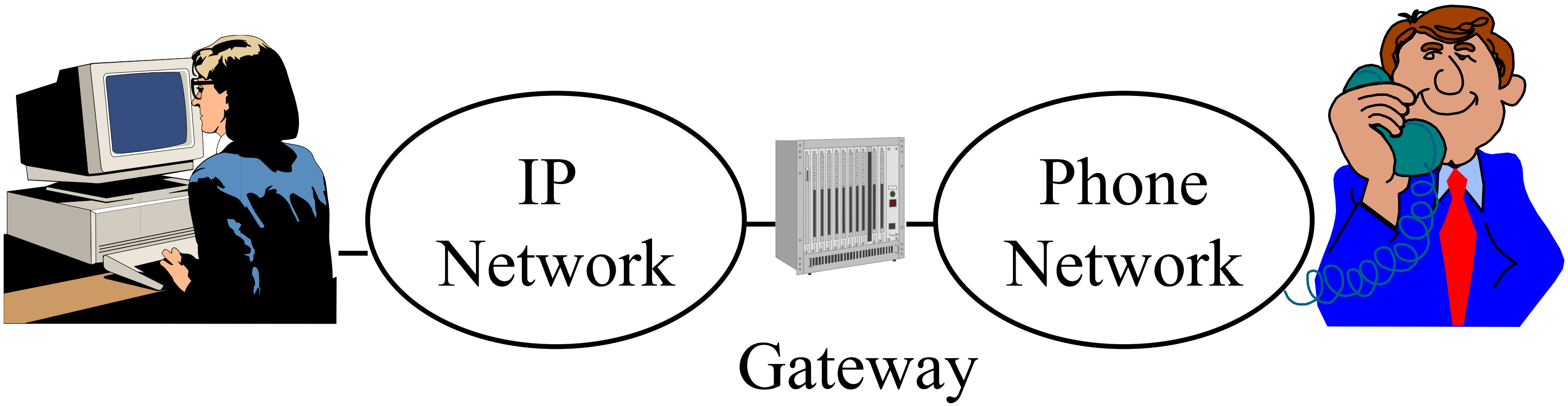
- Sample Products and Services
- 13 Technical Issues
- 4 Other Issues
- H.323 Standard
- Session Initiation Protocol (SIP)

Scenario 1: PC to PC



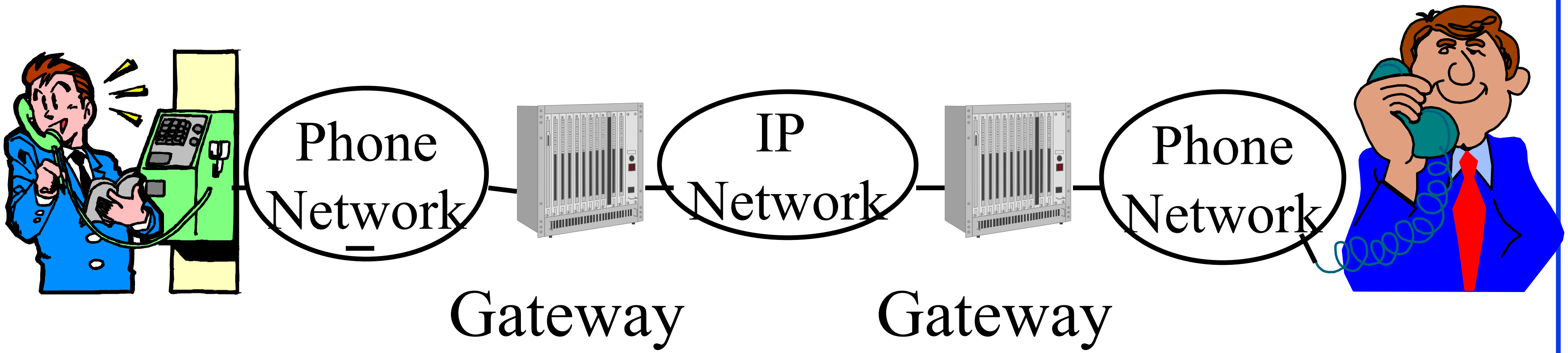
- Need a PC with sound card
- IP Telephony software: Cuseeme, Internet Phone, ...
- Video optional

Scenario 2: PC to Phone



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

Scenario 3: Phone to Phone



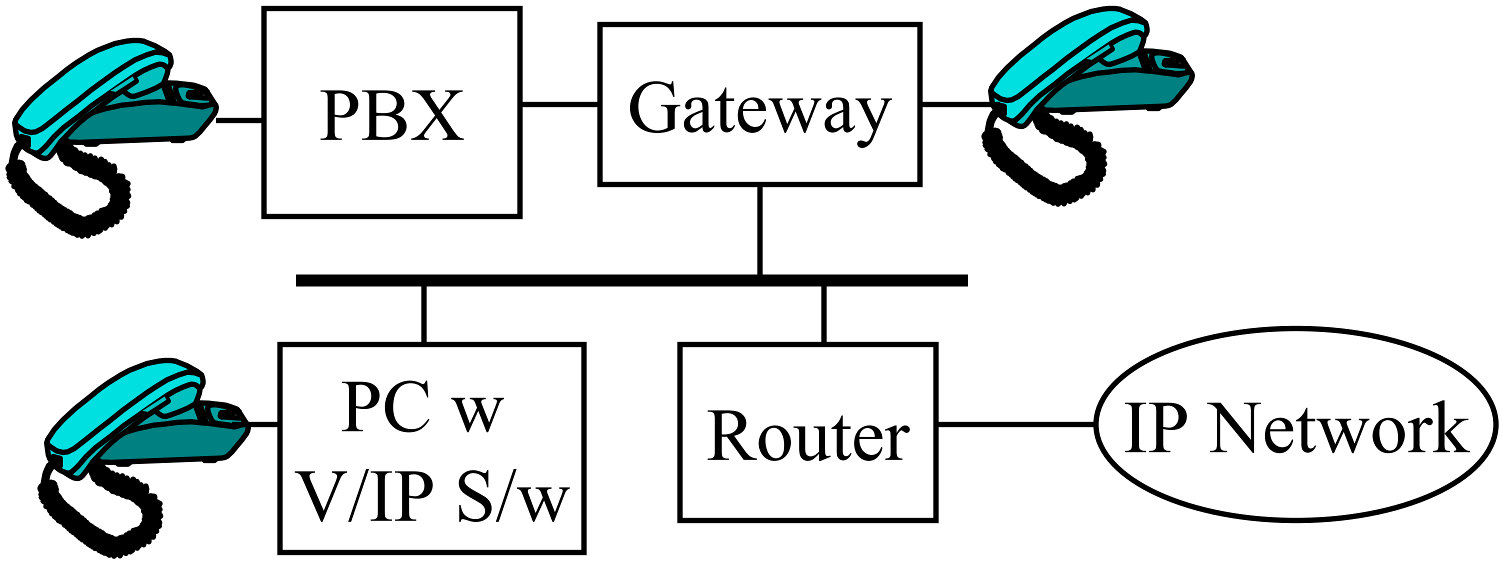
- Need more gateways that connect IP network to phone networks
- The IP network could be dedicated intra-net or the Internet.
- The phone networks could be intra-company PBXs or the carrier switches

Raj Jain

Sample Products

- VocalTec Internet Phone: PC to PC.
- Microsoft NetMeeting: PC to PC. Free.
- Internet PhoneJACK: ISA card to connect a standard phone to PC. Works with NetMeeting, InternetPhone etc. Provides compression.
- Internet LineJACK: Single-line gateway.
- Micom V/IP Family:
  - Analog and digital voice interface cards
  - PC and/or gateway

Products (Cont)



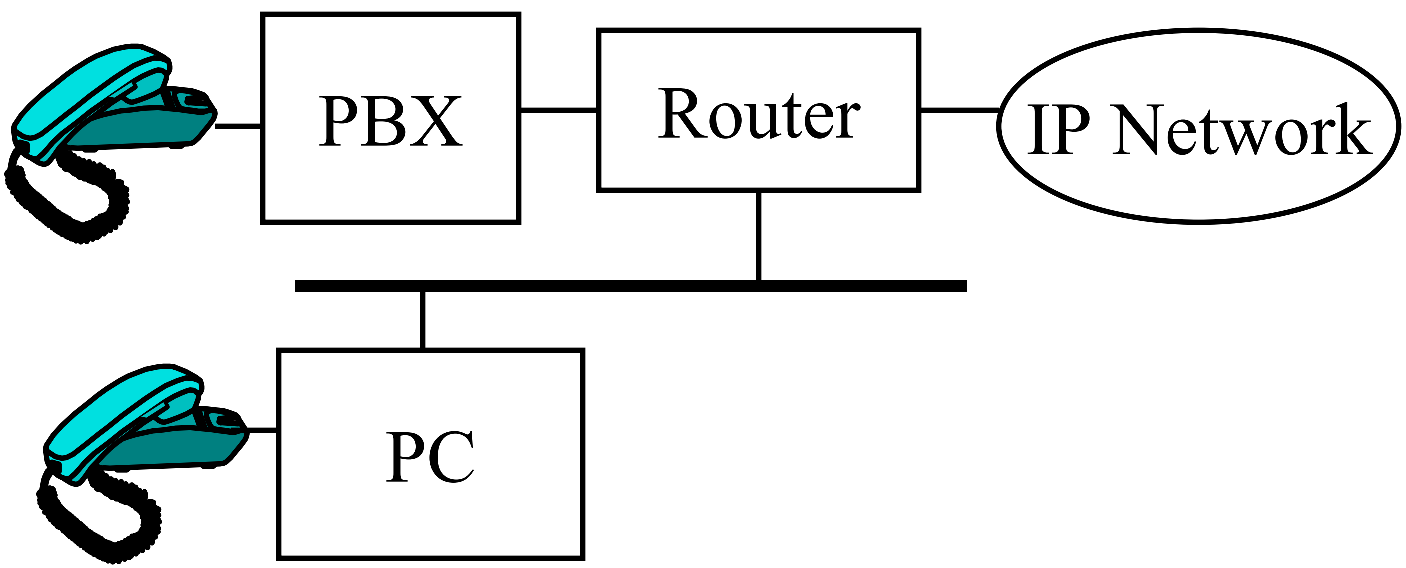
- Features:
  - Compression
  - Phone number to IP address translation.
  - Supports RSVP.
  - Limits number of calls.

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

- IDT 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.
- ITXC 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 cities.

Services (Cont)

- Other 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 telephony.
- Bell Canada has formed 'Emergis' division.
- Bellcore has formed 'Soliant Internet Systems' unit
- Bell Labs has formed 'Elemedia' division

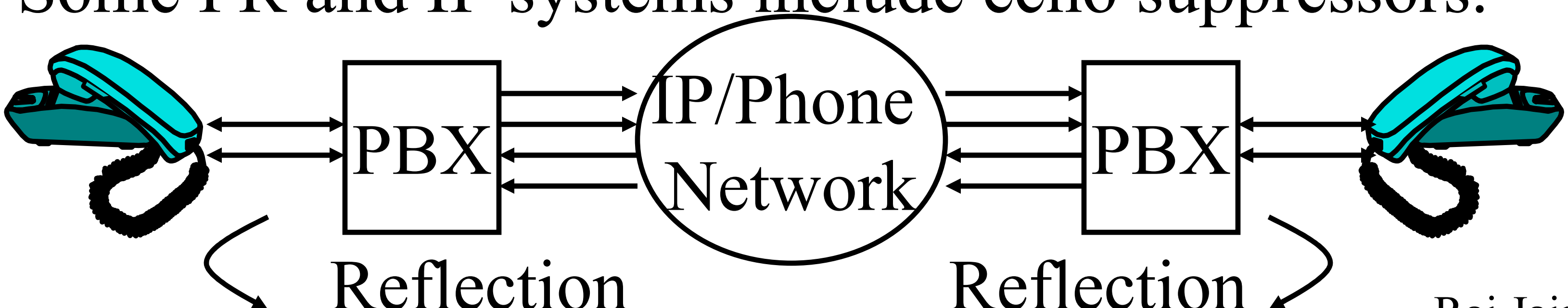
Technical Issues

1. 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)

2. Delay Jitter: Need priority for voice packets. Shorter packets? IP precedence (TOS) field.
3. Frame length: 9 kB at 64 kbps = 1.125 s
   Smaller MTU ⇒ Fragment large packets
4. Lost Packets: Replace lost packets by silence, extrapolate previous waveform
5. Echo cancellation: 2-wire to 4-wire.
   Some FR and IP systems include echo suppressors.



Technical Issues (Cont)

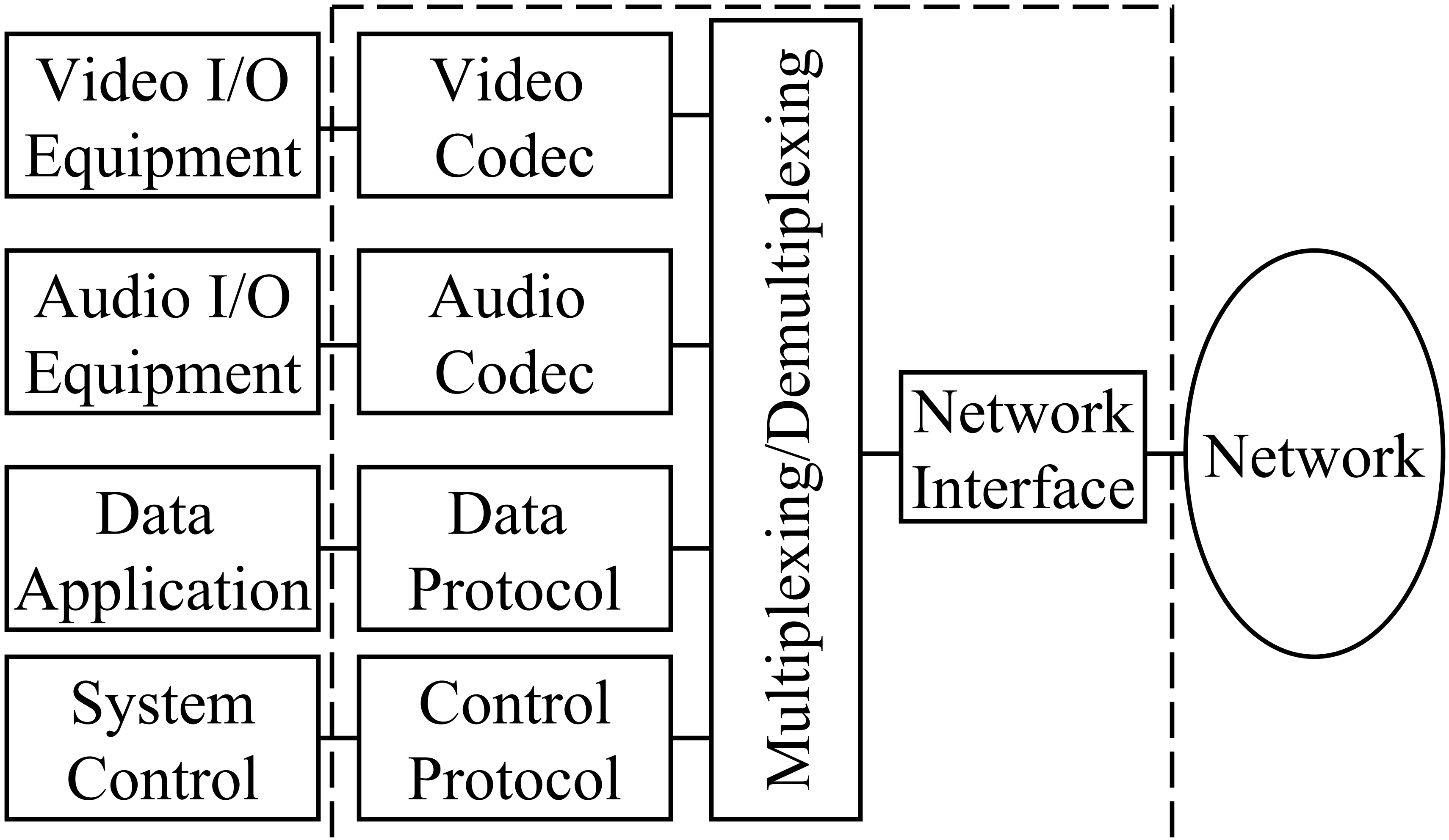
6. Silence suppression
7. Address translation: Phone # to IP. Directory servers.
8. Telephony signaling: Different PBXs may use different signaling methods.
9. Bandwidth Reservations: Need RSVP.
10. Multiplexing: Subchannel multiplexing
    $\Rightarrow$ Multiple voice calls in one packet.
11. Security: Firewalls may not allow incoming IP traffic
12. Insecurity of internet
13. Voice compression: Load reduction

Raj Jain

Other Issues

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

Telephony/Conferencing Systems



Raj Jain

Conferencing Standards

Network	ISDN	ATM	PSTN	LAN	POTs
Conf. Std.	H.320	H.321	H.322	H.323 V1/V2	H.324
Year Audio Codec	1990 G.711, G.722, G.728	1995 G.711, G.722, G.728	1995 G.711, G.722, G.728	1996/1998 G.711, G.722, G.723.1, G.728, G.729	1996 G.723.1, G.729
Audio Rates kbps Video Codec	64, 48-64 H.261	64, 48-64, 16 H.261, H.263	64, 48-64, 16 H.261, H.263	64, 48-64, 16, 8, 5.3/6.3 H.261 H.263	8, 5.3/6.3 H.261 H.263
Data Sharing	T.120	T.120	T.120	T.120	T.120
Control	H.230, H.242	H.242	H.242, H.230	H.245	H.245
Multiplexing	H.221	H.221	H.221	H.225.0	H.223
Signaling	Q.931	Q.931	Q.931	Q.931	-

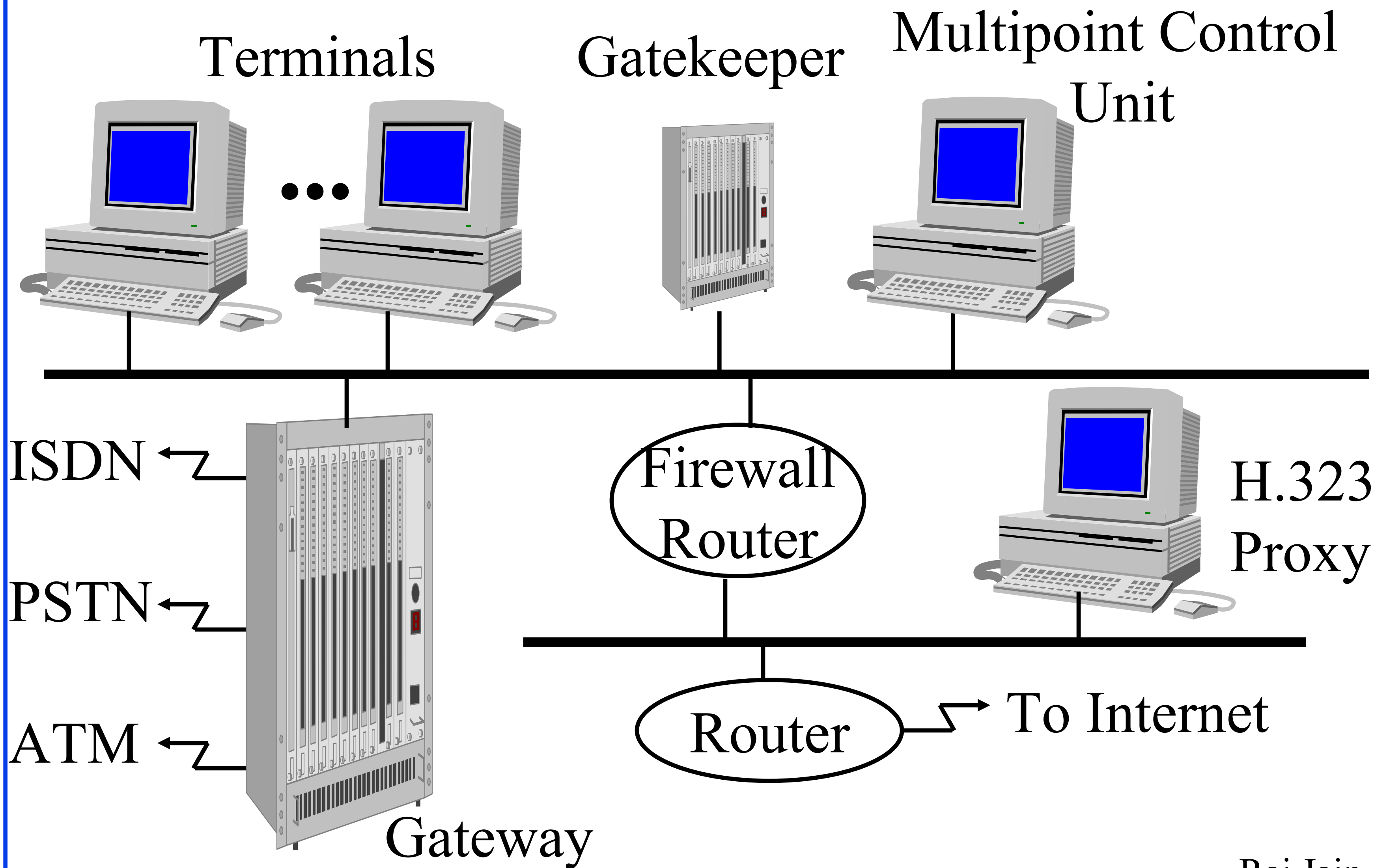
Raj Jain

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>Data</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.124</td></tr>
<tr><td colspan="2">RTP</td><td colspan="3">X.224 Class 0</td><td>T.125</td></tr>
<tr><td colspan="3">UDP</td><td colspan="3">TCP</td><td rowspan="2">T.123</td></tr>
<tr><td colspan="6">Network (IP)</td></tr>
<tr><td colspan="7">Datalink (IEEE 802.3)</td></tr>
</table>

H.323 Components



Raj Jain

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

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)

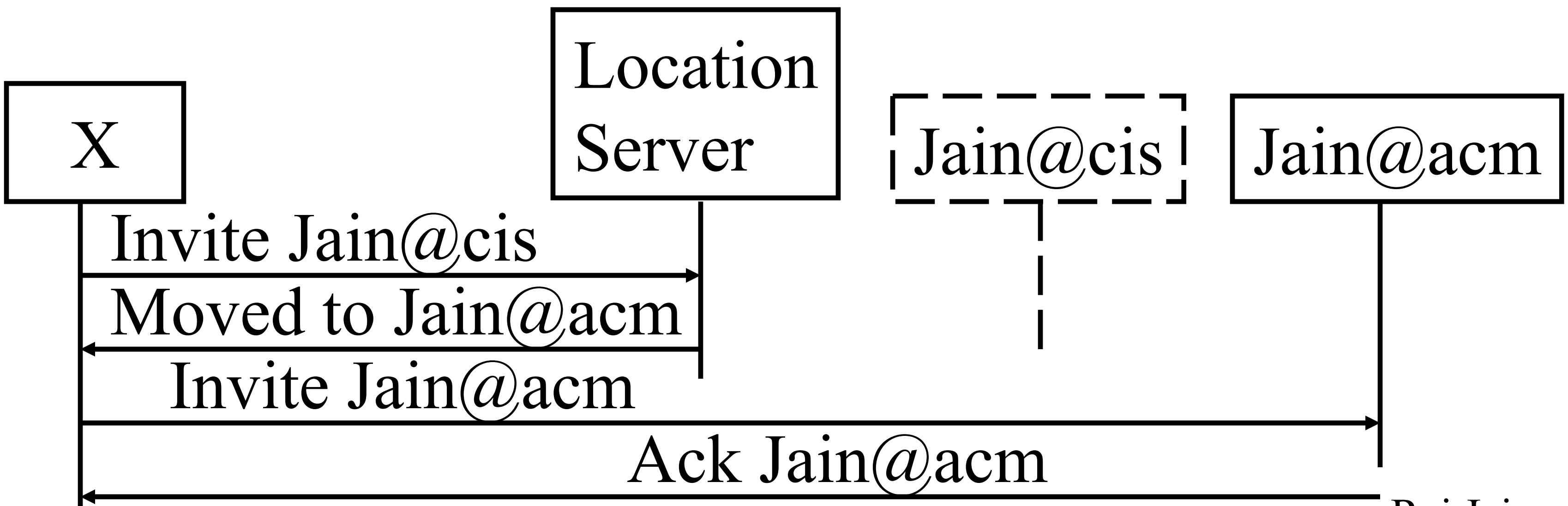
- SIP works in conjunction with other IP protocols for multimedia:
  - RSVP for reserving network resources
  - RTP/RTCP/RTSP for transporting real-time data
  - Session Announcement Protocol (SAP) for advertising multimedia session
  - Session description protocol (SDP) for describing multimedia session
- Can also be used to determine whether party can be reached via H.323, find H.245 gateway/user address

SIP (Cont)

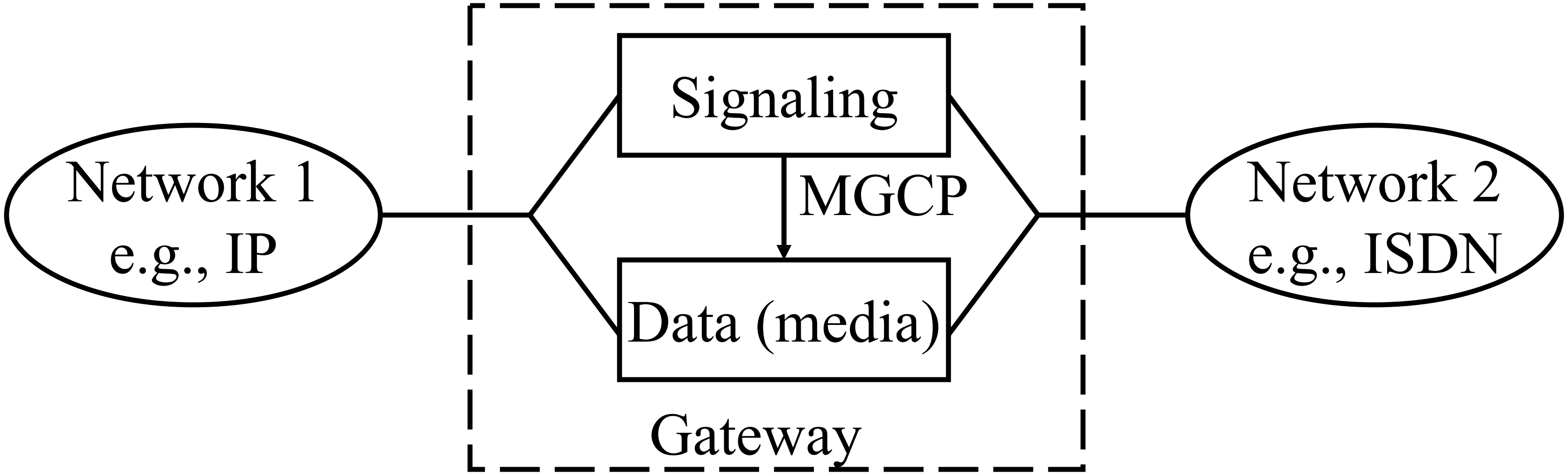
- SIP is text based (similar to HTTP)
  $\Rightarrow$ SIP messages can be easily generated by humans, CGI, Perl, or Java programs.
- SIP Uniform Resource Locators (URLs):
  Similar to email URLs
  sip:jain@cse.ohio-state.edu
  sip:+1-614-292-3989:123@osu.edu?subject=lecture
- SIP messages are sent to SIP server at the specified IP address
- SIP 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 callee
- SIP Messages: Ack, Bye, Invite, Register, Redirection, ...



Media Gateway Control Protocol

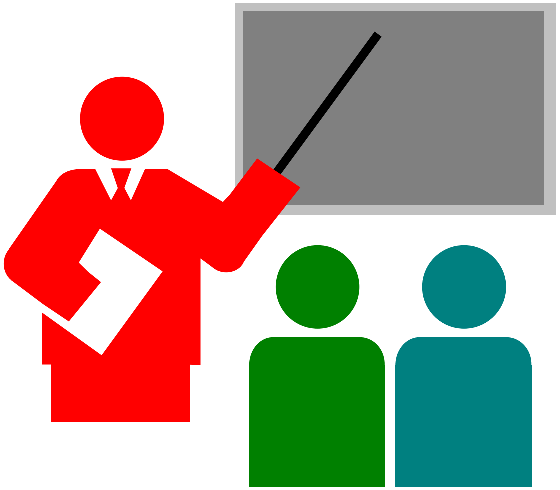


- Gateway = Signaling Fns + Media Transfer Fns
- Call Agents: Signaling functions ⇒ Intelligent
  ⇒ More complex ⇒ Fewer
  ⇒ Control multiple media gateways ⇒ Need MGCP
- MGCP =Simple Gateway Control Protocol (SGCP)
  + Internet Protocol Device Control (IPDC)

MGCP Commands

- Endpoint Configuration (EPCF): Specify coding
- Notification Request (RQNT): Watch for event
- Notify (NTFY): Used by gateway to inform Call agent
- 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 indicate initialization/shutdown of endpoints/gateway

Summary



- Voice over IP products and services are being rolled out
- Ideal for computer-based communications
- IP needs QoS for acceptable quality
- A number of working group at IETF are working on it
- H.323 provides interoperability

References

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