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
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)

- PBX
- Gateway
- PC w V/IP S/w
- Router
- IP Network

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

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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.
3. Frame length: 9 kB at 64 kbps = 1.125 s Smaller MTU $\Rightarrow$ 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
    ⇒ Multiple voice calls in one packet.
11. Security: Firewalls may not allow incoming IP traffic
12. Insecurity of internet
13. Voice compression: Load reduction
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
   ⇒ 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

- Video I/O Equipment
- Audio I/O Equipment
- Data Application
- System Control
- Video Codec
- Audio Codec
- Data Protocol
- Control Protocol
- Multiplexing/Demultiplexing
- Network Interface
- Network

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## Conferencing Standards

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H.323 Protocols

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

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Network (IP)

Datalink (IEEE 802.3)

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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
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)
  ⇒ 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, ...

![Diagram showing the process of locating a callee using SIP with messages and locations labeled.]
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)

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