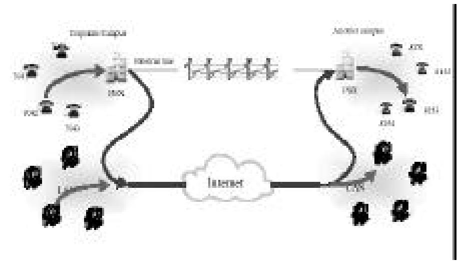
Internet Telephony



Shivkumar Kalyanaraman

Based upon slides of Henning Schulzrinne (Columbia)

Overview

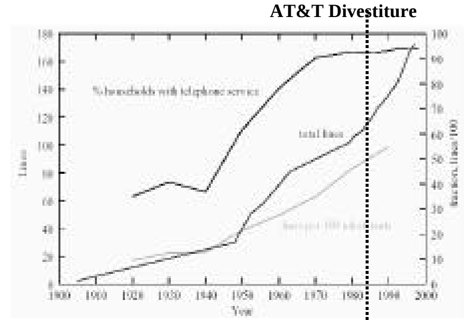
- Telephony: history and evolution
- IP Telephony: Why ?
  - Adding interactive multimedia to the web
  - Being able to do basic telephony on IP with a variety of devices
- Basic IP telephony model
- Protocols: SIP, H.323, RTP, Coding schemes, MGCP, RTSP
- Future: Invisible IP telephony and control of appliances

Public Telephony (PSTN) History

- 1876 invention of telephone
- 1915 first transcontinental telephone (NY–SF)
- 1920's first automatic switches
- 1956 TAT-1 transatlantic cable (35 lines)
- 1962 digital transmission (T1)
- 1965 1ESS analog switch
- 1974 Internet packet voice
- 1977 4ESS digital switch
- 1980s Signaling System #7 (out-of-band)
- 1990s Advanced Intelligent Network (AIN)

Telephone Service in the US

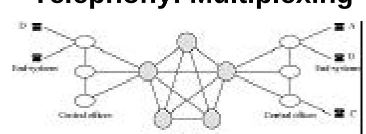
AT&T Divestiture



Telephone System Overview

- Analog narrowband circuits: home-> central office
- 64 kb/s continuous transmission, with compression across oceans
  - μ-law: 12-bit linear range -> 8-bit bytes
- Everything clocked a multiple of 125 s
  - Clock synchronization ⇔ framing errors
- AT&T: 136 "toll"switches in U.S.
  - Interconnected by T1, T3 lines & SONET rings
- Call establishment "out-of-band" using packet-switched signaling system (SS7)

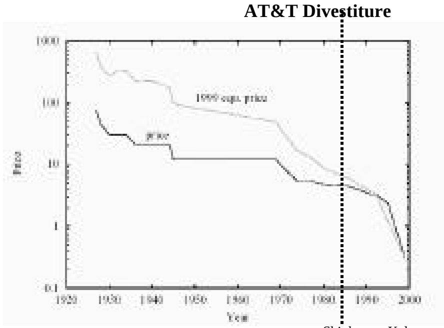
Telephony: Multiplexing



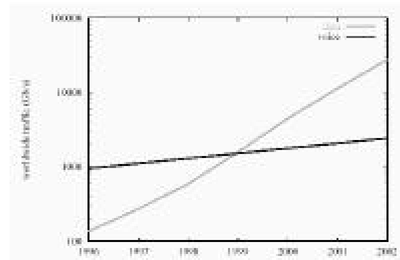
- ***Telephone Trunks*** **between central offices carry hundreds of conversations: Can't run thick bundles!**
- **Send many calls on the same wire: *multiplexing***
- **Analog multiplexing**
  - **bandlimit call to 3.4 KHz and frequency shift onto higher bandwidth trunk**
- **Digital multiplexing: convert voice to *samples***
  - **8000 samples/sec => call = 64 Kbps**

Trends: Price of Phone Calls: NY - London

AT&T Divestiture

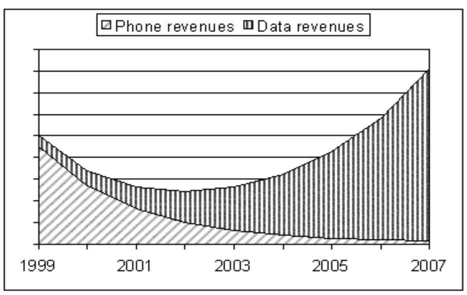


Trends: Data vs Voice Traffic



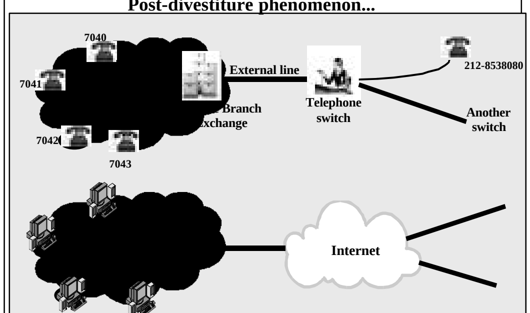
Since we are building future networks for data, can we slowly junk the voice infrastructure and move over to IP?

Trends: Phone vs Data Revenues

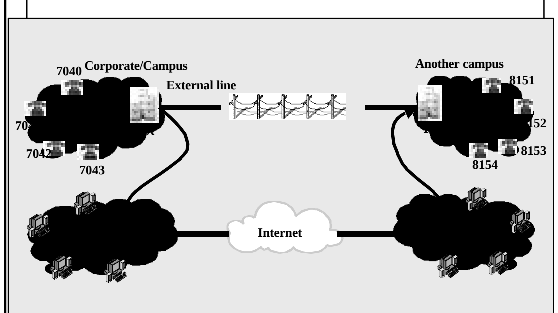


Private Branch Exchange (PBX)

Post-divestiture phenomenon...

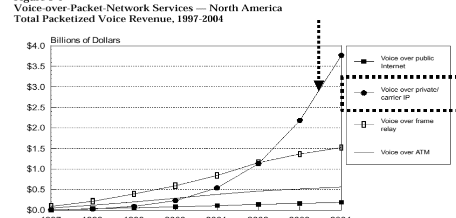


IP Telephony: PBX Replacement



Voice over Packet Market Forecast – North America

Figure 1-4
Voice-over-Packet-Network Services — North America
Total Packetized Voice Revenue, 1997-2004



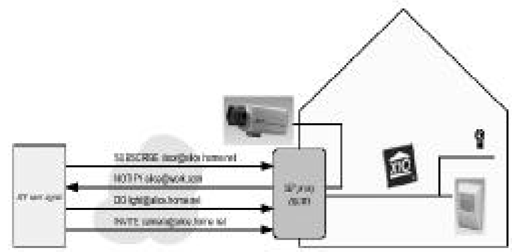
Source: Dataquest (March 2000 Estimates)

March 24, 2000

Invisible Internet Telephony

- VoIP technology will appear in . . .
  - Internet appliances
  - home security cameras, web cams
  - 3G mobile terminals
  - fire alarms
  - chat/IM tools
  - interactive multiplayer games

IPtel for appliances: "Presence"



Taxonomy of Speech Coders

Speech Coders
- **Waveform Coders**
  - **Time Domain: PCM, ADPCM**
  - **Frequency Domain: e.g. Sub-band coder, Adaptive transform coder**
- **Source Coders**
  - **Linear Predictive Coder**
  - **Vocoder**

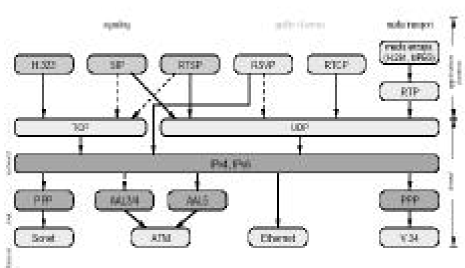
- **<u>Waveform coders:</u> attempts to preserve the signal waveform not speech specific.**
  - **PCM 64 kbps, ADPCM 32 kpbs, CVSDM 32 kbps**
- **<u>Vocoders:</u>**
  - **Analyse speech, extract and transmit model parameters**
  - **Use model parameters to synthesize speech**
  - **LPC-10: 2.4 kbps**
- **<u>Hybrids:</u> Combine best of both… Eg: CELP**

VoIP Camps

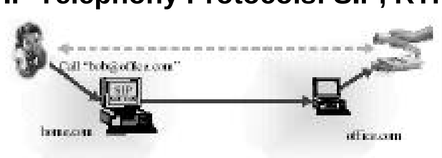
Conferencing Industry	Netheads "IP over Everything"	Circuit switch engineers "We over IP"	"Convergence" ITU standards
H.323	**SIP**	**"Softswitch"**	**BICC**
ISDN LAN conferencing	I-multimedia WWW	Call Agent SIP & H.323	BISDN, AIN H.xxx, SIP
IP	IP	IP	"any packet"

Our focus

Internet Multimedia Protocol Stack

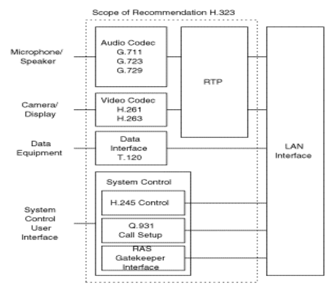


IP Telephony Protocols: SIP, RTP



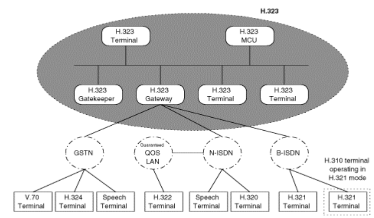
- Session Initiation Protocol - SIP
  - **Contact "office.com" asking for "bob"**
  - **Locate Bob's current phone and ring**
  - **Bob picks up the ringing phone**
- Real time Transport Protocol - RTP
  - **Send and receive audio packets**

Internet Telephony Protocols: H.323

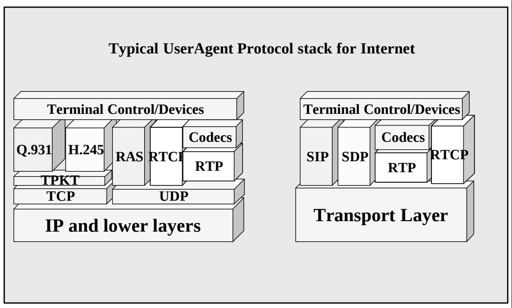


H.323 (contd)

- Terminals, Gateways, Gatekeepers, and Multipoint Control Units (MCUs)



H.323 vs SIP



SIP vs H.323

- **Text based request response**
- **SDP (media types and media transport address)**
- **Server roles: registrar, proxy, redirect**

- **Binary ASN.1 PER encoding**
- **Sub-protocols: H.245, H.225 (Q.931, RAS, RTP/RTCP), H.450.x...**
- **H.323 Gatekeeper**

- Both use RTP/RTCP over UDP/IP
- H.323 perceived as "heavyweight"

Light-weight signaling: Session Initiation Protocol (SIP)

- IETF MMUSIC working group
- Light-weight generic signaling protocol
- Part of IETF conference control architecture:
  - SAP for "Internet TV Guide" announcements
  - RTSP for media-on-demand
  - SDP for describing media
  - others: malloc, multicast, conference bus, . . .
- Post-dial delay: 1.5 round-trip time (with UDP)
- Network-protocol independent: UDP or TCP (or AAL5 or X.25)

SDP: Session Description Protocol

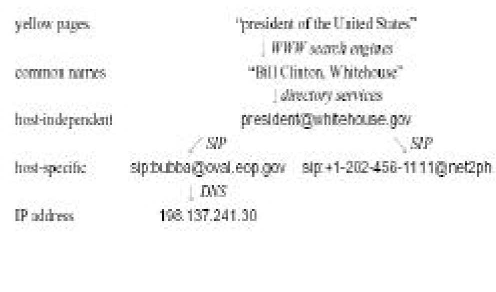
- **Not really a *protocol* – describes data carried by other protocols**
  - **Used by SAP, SIP, RTSP, H.332, PINT. Eg:**

```
v=0
o=g.bell 877283459 877283519 IN IP4 132.151.1.19
s=Come here, Watson!
u=http://www.ietf.org
e=g.bell@bell-telephone.com
c=IN IP4 132.151.1.19
b=CT:64
t=3086272736 0
k=clear:manhole cover
m=audio 3456 RTP/AVP 96
a=rtpmap:96 VDVI/8000/1
m=video 3458 RTP/AVP 31
m=application 32416 udp wb
```

SIP functionality

- 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

SIP Addresses Food Chain



SIP components

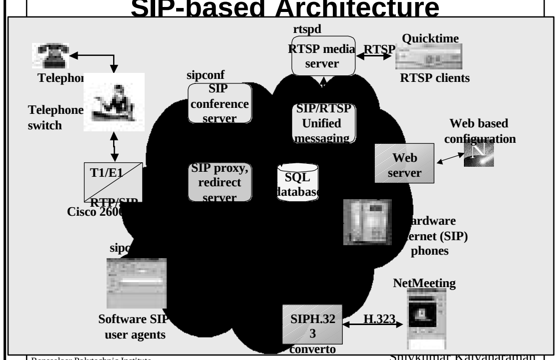
- **UAC:** user-agent client (caller application)
- **UAS:** user-agent server à accept, redirect, refuse call
- **redirect server:** redirect requests
- **proxy server:** server + client
- **registrar:** track user locations
- user agent = UAC + UAS
- often combine registrar + (proxy or redirect server)

IP SIP Phones and Adaptors

Are true Internet hosts
- **Choice of application**
- **Choice of server**
- **IP appliances**

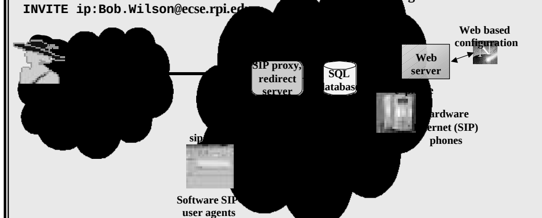
Implementations
- **3Com (3)**
- **Columbia University**
- **MIC WorldCor**
- **Mediatrix (1)**
- **Nortel (4)**
- **Siemens (5)**

SIP-based Architecture

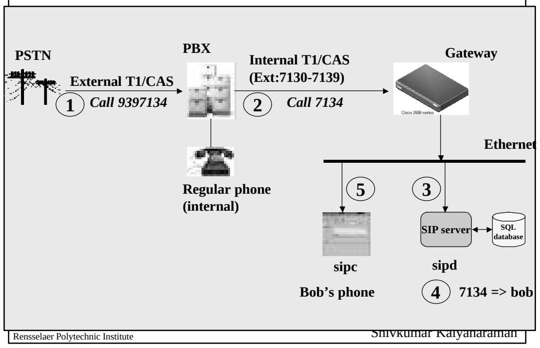


Example Call

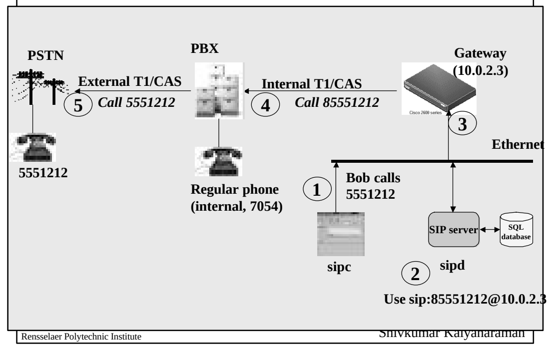
- Bob signs up for the service from the web as "bob@ecse.rpi.edu"
- He registers from multiple phones
- Alice tries to reach Bob
  `INVITE ip:Bob.Wilson@ecse.rpi.ed`
- sipd canonicalizes the destination to `sip:bob@ecse.rpi.edu`
- sipd rings both e*phone and sipc
- Bob accepts the call from sipc an ing



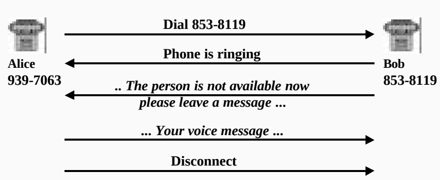
PSTN to IP Call



IP to PSTN Call

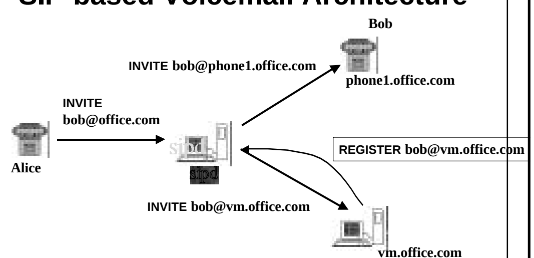


Traditional voice mail system



Bob can listen to his voice mails by dialing some number.

SIP-based Voicemail Architecture

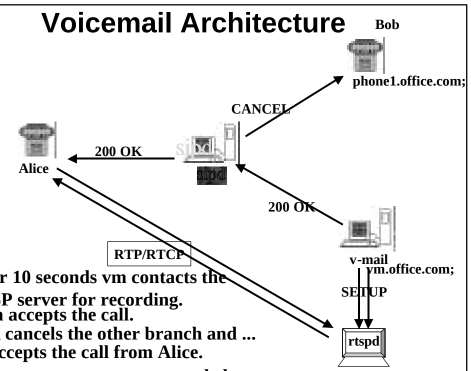


The voice mail server registers with the SIP proxy, sipd

Alice calls *bob@office.com* through SIP proxy.

SIP proxy forks the request to Bob's phone as well as to a voicemail server.

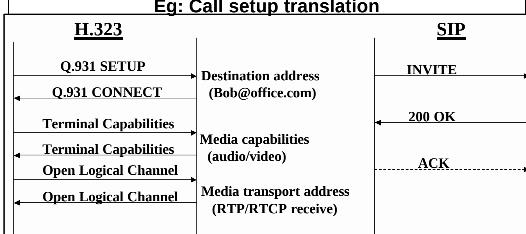
Voicemail Architecture



After 10 seconds vm contacts the RTSP server for recording.
vm accepts the call.
Sipd cancels the other branch and ...
...accepts the call from Alice.
Now user message gets recorded

SIP-H.323: Interworking Problems

Eg: Call setup translation



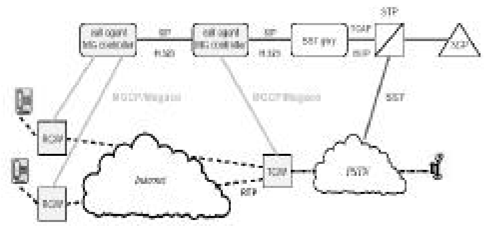
- H.323: Multi-stage dialing

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**
  - **Call control "intelligence" is outside the gateways and handled by external call control elements**
- Goal: scalable gateways between IP telephony and PSTN
- Successor to MGCP: H.248/Megaco

MGCP Architecture

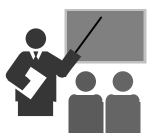
Goal: large-scale phone-to-phone VoIP deployments



RGW: Residential Gateway

TGW: Trunk Gateway

Summary



- Telephony and IP Telephony
- Protocols: SIP, SDP, H.323, MCGP
- Example operation and services:
  - Calls, voice mail etc
- Future: Integration with Web and long-term replacement for current telephone systems