Voice over IP

Presentation to Muug
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March 9, 2004

VoIP Info - www.voip-info.org

Asterisk Open Source PBX - www.asterisk.org

VoIP interface cards - www.digium.com

Free World Dialup - www.freeworldialup.com

What I plan to talk about

- VoIP technology
- SIP
- Asterisk
- SIP phones
- Demonstration

Plain Old Telephone Service vs VoIP

- Time Division Multiplexing vs Packet
- Circuit vs Packet
- Separate vs Converged
- Dumb end points vs Smart end points
- 99.999 (5 min/yr) vs ??
- Central control vs Distributed
- Busy vs Degraded quality

VoIP Issues

- Quality of Service
- Latency or delay (RTT < 300ms)
- Jitter
- Voice quality
  - Distortion
  - Background Noise
  - Echo cancelation
  - Tandem conversions
- Regulatory

Signalling

- On hook / Off hook
- Ringing and busy tones
- Dual Tone Multi-Frequency (digits)
- Inband vs Out-of-band
- VoIP signalling protocols
  - H.323, Media Gateway Control Protocol
  - Session Initiation Protocol
  - Inter Asterisk eXchange

Voice Transport

- Codecs - COmpressor/DECompressor
  - GSM – 13Kbps, 20ms frame
  - G.711 (alaw/ulaw) 64Kbps
  - G.723.1 – 5.3/6.3Kbps, 30ms frame
  - G.726 – 16/24/32/40 Kbps
  - G.729 – 8Kbps, 10ms frame
  - iLBC – 15Kbps, 20 ms frame
- Real Time Protocol

SIP

- Stateless or stateful
- Similar formats and syntax of HTTP (clear text)
- SIP message is opaque
- Identifies a user with a URI sip:reid@muug.mb.ca
- Session Description Protocol
- Servers: registrars, proxy, redirect

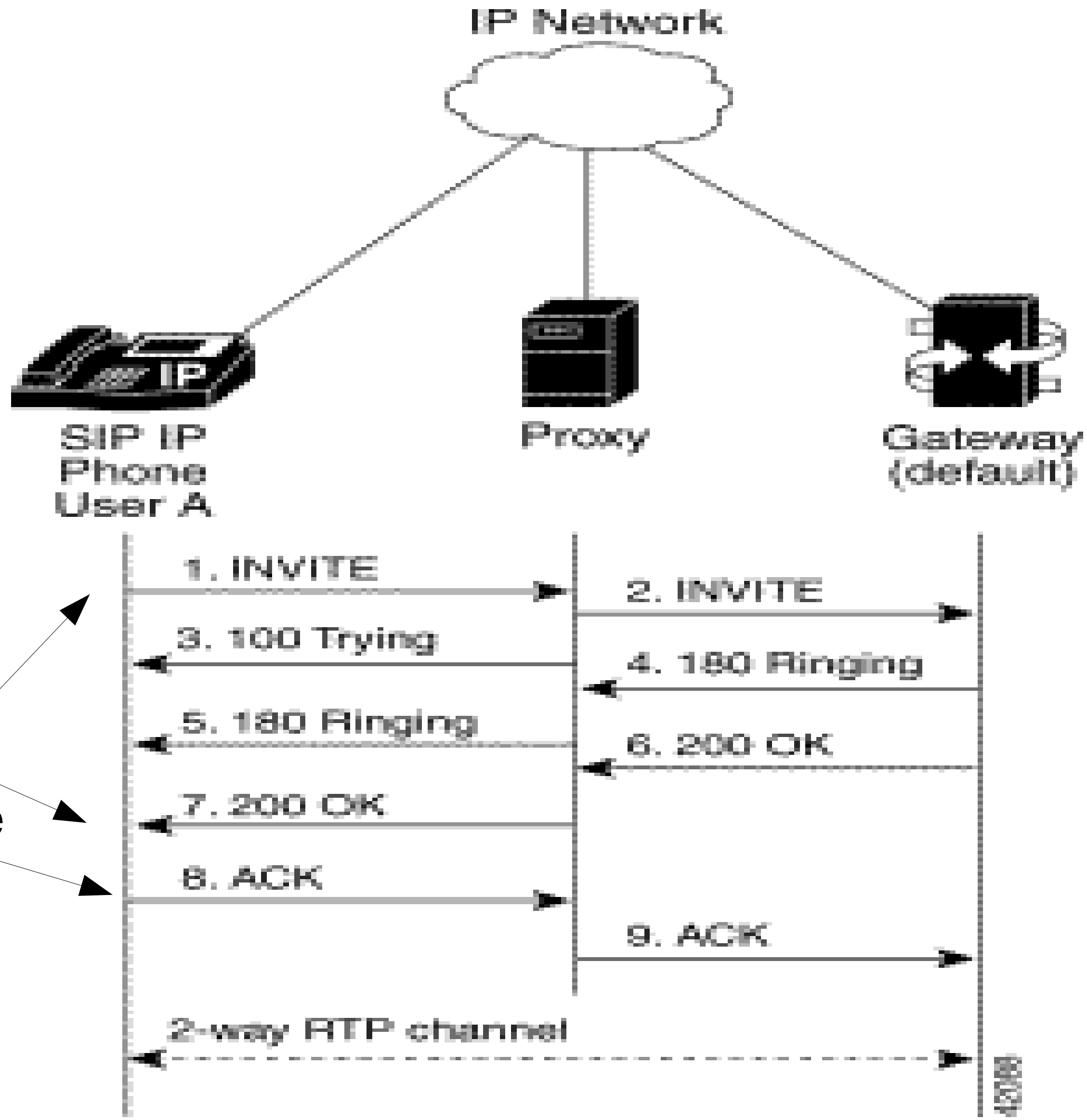
Requests

- INVITE
- ACK
- OPTIONS
- BYE
- CANCEL
- REGISTER

Responses

- 100-199 Informational
- 200-299 Success
- 300-399 Redirection
- 400-499 Client error
- 500-599 Server error
- 600-699 Global failure

SIP Phone to SIP Gateway

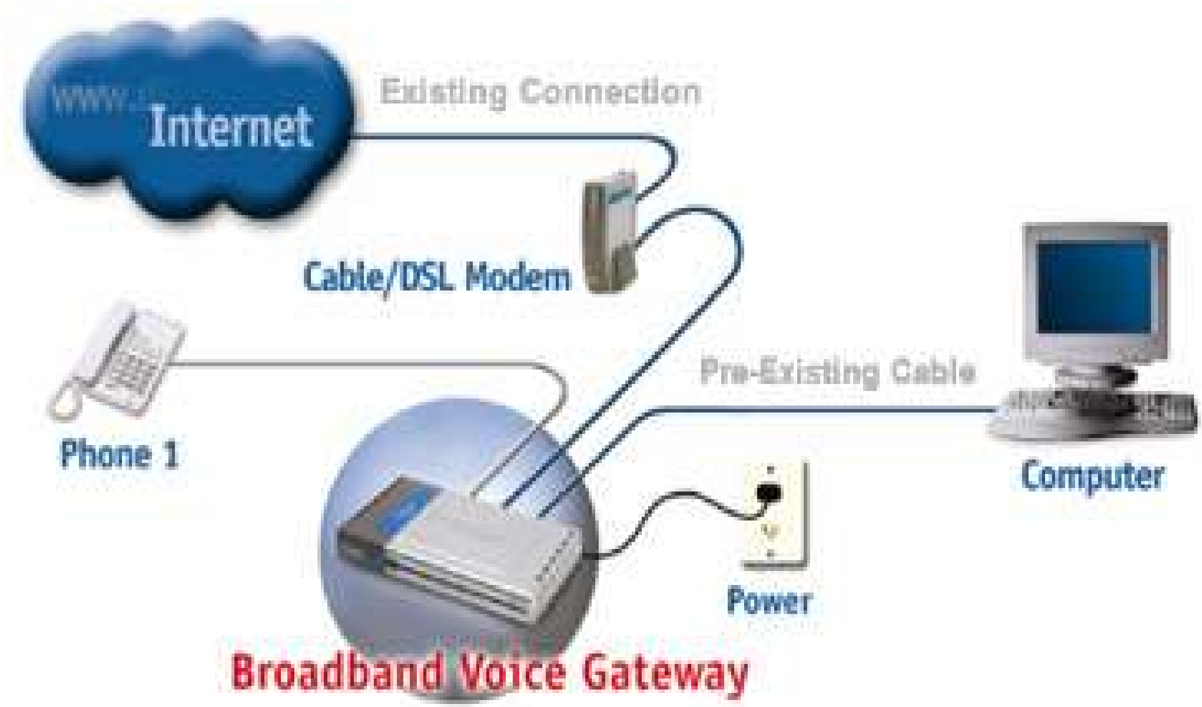


3-way handshake

VoIP Deployment

- Power Over Ethernet – 802.3af
  - 48 volts AC, 350 milliamps
  - Uses 2 wire pairs
  - Detects non-compliant devices
- Private LAN
- Branch office
- Toll Bypass
- Commercial VoIP to PSTN
  - Vonage(US) supports SIP

Primus's TalkBroadband



POTS Analog Interfaces

- Foreign eXchange Subscriber
  - Plug in the wall
  - Dial tone
  - Battery current
  - Ring voltage
- Foreign eXchange Office
  - Plug in phone
  - on-hook/off-hook indication

Asterisk

- Open Source hybrid TDM and packet voice PBX
- IVR platform with ACD functionality
- Written in C
- Runs only on Linux
- Sponsored by Digium
- Active user community
  - Email lists, www.voip-info.org (Wiki)

Asterisk Features

- Seamlessly supports VoIP, digital and analog channels
- Standard call features: 3-way calling, CallerID, Call Waiting, Call Forwarding
- Advanced features: voicemail, conferencing, Interactive Voice Response(IVR), automatic call distribution (ACD)

Asterisk Dialplan

- Directs routing of calls through Asterisk
- Composed of extension contexts
- Contexts are groups of extensions
- Contexts can include one another
- Each step in the Dialplan is an Application
- Each step is assigned a priority sequence

Major Applications

- Voicemail
- MeetMe – conference bridge
- MusicOnHold
- Directory – voicemail extensions
- Monitor – records a channel
- Festival – Text to Speech
- AGI – like the a Web CGI – shell, Perl, Python, C, Java
- Authenticate
- Zapateller – block telemarketers