Introduction to SIP

Russ Clark
September 8, 2008
IP Telephony
IP Telephony

Voice over IP - VoIP

- We want to take the analog, connection-oriented voice service and provide it over a digital, best-effort, datagram service.

- Are we crazy or what?
IP Telephony

Voice over IP - VoIP

- We want to take the analog, connection-oriented voice service and provide it over a digital, best-effort, datagram service.
- Are we crazy or what?
- Must deal with Analog to Digital Conversion
  - Codecs: e.g. G.711 - 64Kbps, G.729A - 8Kbps, G.723.1 - 6.4Kbps
IP Telephony

Voice over IP - VoIP

• We want to take the analog, connection-oriented voice service and provide it over a digital, best-effort, datagram service.

• Are we crazy or what?

• Must deal with Analog to Digital Conversion
  • Codecs: e.g. G.711 - 64Kbps, G.729A - 8Kbps, G.723.1 - 6.4Kbps

• Must deal with Delay, Jitter, Packet Loss
  • <150 ms is imperceptible, 150-400 ms is OK, >400 ms is way out
IP Telephony

Voice over IP - VoIP

• We want to take the analog, connection-oriented voice service and provide it over a digital, best-effort, datagram service.

• Are we crazy or what?

• Must deal with Analog to Digital Conversion

  • Codecs: e.g. G.711 - 64Kbps, G.729A - 8Kbps, G.723.1 - 6.4Kbps

• Must deal with Delay, Jitter, Packet Loss

  • <150 ms is imperceptible, 150-400 ms is OK, >400 ms is way out

• Must create the illusion of a call session

  • need a session protocol
VoIP Protocols
VoIP Protocols

- SIP - Session Initiation Protocol - RFC 3261
- Call Management, Call Setup and Control
VoIP Protocols

- SIP - Session Initiation Protocol - RFC 3261
  - Call Management, Call Setup and Control
- SDP - Session Description Protocol - RFC 2327
  - Describe the parameters for the voice session, carried in SIP INVITE
VoIP Protocols

- **SIP** - Session Initiation Protocol - RFC 3261
  - Call Management, Call Setup and Control
- **SDP** - Session Description Protocol - RFC 2327
  - Describe the parameters for the voice session, carried in SIP INVITE
- **RTP** - Real Time Transport Protocol - RFC 3550
  - For the data exchange, a stream of encoded voice packets
VoIP Protocols

- **SIP - Session Initiation Protocol - RFC 3261**
  - Call Management, Call Setup and Control
- **SDP - Session Description Protocol - RFC 2327**
  - Describe the parameters for the voice session, carried in SIP INVITE
- **RTP - Real Time Transport Protocol - RFC 3550**
  - For the data exchange, a stream of encoded voice packets
- **RSVP - ReSerVation Protocol - RFC 2205**
  - Establish Priority and Reservations INSIDE the network
VoIP Protocols

- User Agents (UA) connected over an IP network
A SIP server isn’t technically required.
But we always use it.
Why? To find you.
SDP is carried as payload in the SIP INVITE and response.
VoIP Protocols

- RTP may not follow the same path as SIP
VoIP Protocols

- RSVP is internal to the network
- Current IMS implementations are limited here