Hit the Ground Running

VoIP

Robert Sparks
VoIP: Voice Over IP

• Real-time interactive Voice (and Video)
  – Not the same as streaming media, but there are some mechanisms in common

• Evolution path for telephony
  – Consolidation of networks and applications
  – Richer Services
  – Universal Accessability
  – Lower Cost
Two Classes of VoIP Systems

• Open Standards based systems
  – SIP
    • Vonage, AT&T, Yahoo, AOL
    • Hundreds of service providers/vendors
  – MGCP/Megaco
  – H.323

• Proprietary, closed systems
  – Skype
High-Level Concepts

• Identity
  – Who are you?

• Presence
  – Are you available to talk?

• Rendezvous
  – How do other people find you?

• Media Negotiation
  – How will you exchange voice or other media?
High-Level Concepts

• Signaling
  - Setting up and controlling a media session
  - Encompasses Rendezvous and Negotiation
  - Can take place over a variety of transports

• Media
  - Usually uses a different transport than signaling
  - Encoded using negotiated codecs.
  - Usually carried using RTP/RTCP (Real Time Transport Protocol) over UDP
Signaling and Media

Rendezvous Service

Signaling
(e.g. SIP over UDP, TCP, or TLS)

Media
(typically RTP over UDP using ephemeral source and destination ports)

Phil → voice → Robert
Codecs

• Agreed encoding of media (voice, video)
• Differing properties
  - Bandwidth consumption (bitrate)
  - Audio quality
  - Resiliency to packetloss and jitter
• Common codecs include
  - G.711
  - G.729
  - ILBC (Internet Low Bitrate Codec)
  - SPEEX wideband
Use of DNS

• SIP uses NAPTR/SRV RRs to
  – Select transport protocols and ports
  – Distribute load between elements

$ORIGIN example.com.
  IN NAPTR 50 50 "s" "SIPS+D2T" "" _sips._tcp.example.com.
  IN NAPTR 90 50 "s" "SIP+D2T" "" _sip._tcp.example.com.
  IN NAPTR 100 50 "s" "SIP+D2U" "" _sip._udp.example.com.

$ORIGIN _sip._tcp.example.com.
  IN SRV 0 1 5060 server1.example.com.
  IN SRV 0 2 5060 server2.example.com.

• ENUM uses NAPTR RRs to map E.164 numbers to Internet Services

$ORIGIN 3.8.0.0.6.9.2.3.6.1.4.4.e164.arpa.
  NAPTR 10 100 "u" "E2U+sip" "!^.*$!sip:info@example.com!".
  NAPTR 10 102 "u" "E2U+msg" "!^.*$!mailto:info@example.com!".


The Landscape

User Agents (end-user endpoints)

• ATA (Analog Terminal Adaptor)
  – Connects a legacy analog telephone to a VoIP system

• Hard Phone
  – Looks like a phone, acts like a phone (and more) but has an ethernet port instead of an analog RJ11 jack

• Soft Client
  – Programs that run on general purpose PCs
The Landscape
Rendezvous using SIP

Address of Record (AoR)

INVITE sip:RjS@estacado.net
From: sip:pckizer@tamu.edu

INVITE sip:9725551212@telco.com
From: sip:pckizer@tamu.edu

REGISTER sip:estacado.net
To: sip:RjS@estacado.net
Contact: sip:9725551212@telco.com
The Landscape
Trapezoid model

tamu.edu

SIP Proxy

Phil

estacado.net

SIP Proxy

Robert

SIP

RTP

voice
Ecosystem Members

Session Border Controllers

tamu.edu

SBC
UA | UA

estacado.net

SBC
UA | UA

Phil

Robert

SIP

RTP
Ecosystem Members

PSTN Gateways

SIP Proxy

SIP/PSTN Gateway

Phil

Robert

SIP
RTP
Ecosystem Members
Voice Mail / IVR systems

- Phil
- SIP Proxy
- Robert’s Voicemail Server
- SIP
- RTP
Ecosystem Members
Conference Servers

SIP Proxy

Conference Server

SIP Proxy

SIP
RTP
Ecosystem Members

Presence Servers

- SIP Proxy
- Presence Server
  - Subscribe
  - Notify
  - Manage Buddylist

- Publish
- Grant Permissions

SIP

XCAP
Forking

tamu.edu

SIP Proxy

Phil

estacado.net

SIP Proxy

Voicemail Server

Robert

some provider

Gateway

Robert’s home phone

some ISP

SIP
Hot Topics

• Authentication
  – Digest
  – Certificate-based (SIP Identity, TLS)

• Securing Media
  – SRTP
  – RTP over DTLS

• Nat/Firewall Traversal
  – STUN (Simple Traversal of UDP through Nats)
  – TURN (Traversal Using Relay Nat)
  – ICE (Interactive Connectivity Establishment)
Hot Topics

• ENUM
  – Using DNS to bind E.164 numbers to Internet Services

• Fixed/Mobile Convergence
  – Handoff between WiFi and Cellular

• E911 (Enhanced 911)
SIP Implementations and Services

• Hundreds available. Some information at
  – www.sipforum.org
  – www.sipcenter.com

• Many Open-source implementations, including
  – www.sipfoundry.org
  – www.iptel.org
  – www.asterisk.org
Evaluating Implementations

• Interoperability is the most important aspect to evaluate

• Useful question: Has the implementation been to SIPit?
  – International Interop Test Event
  – Held twice a year
  – ~100 implementations from ~80 vendors
  – www.sipit.net
Other Resources

- IETF: www.ietf.org
  - Working Groups
    - SIP, SIPPING, SIMPLE, AVT, MMUSIC, BEHAVE, SPEER, XCON, ENUM

- Books
  - “Internet Communications Using SIP”, H. Sinnreich, A. Johnston
  - “SIP beyond VoIP”, H. Sinnreich, A. Johnston, R. Sparks
  - “SIP Demystified”, G. Camarillo
  - “RTP: Audio and Video for the Internet”, C. Perkins