Hit the Ground Running

VoIP

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ESTACADO
SYSTEMS

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

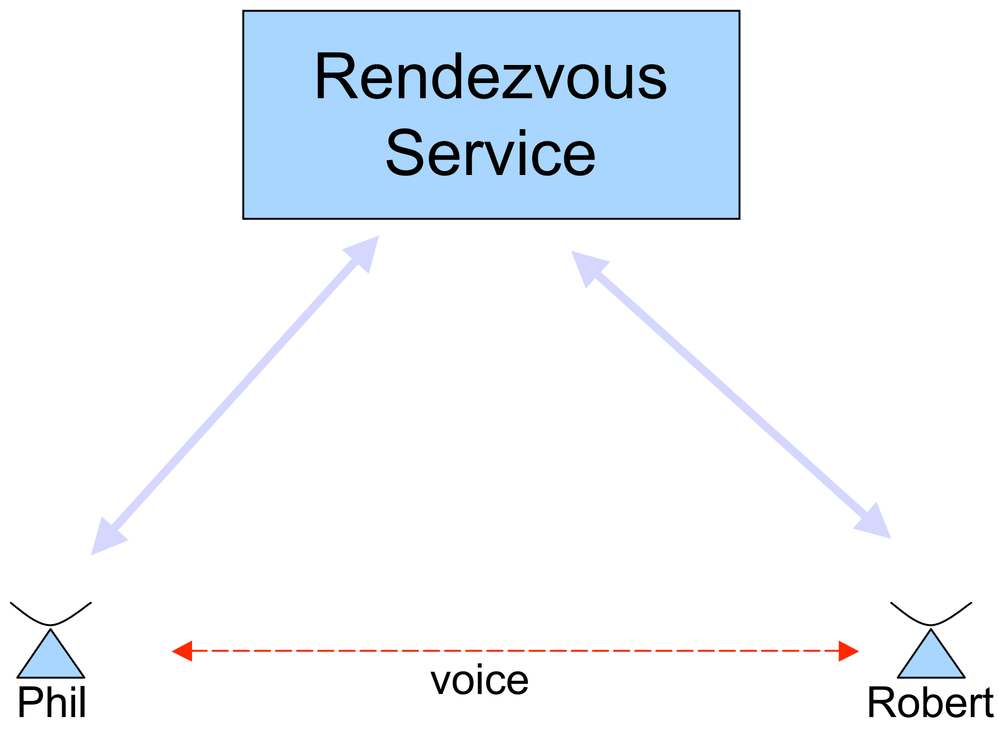
High-Level Concepts

- 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



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

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

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

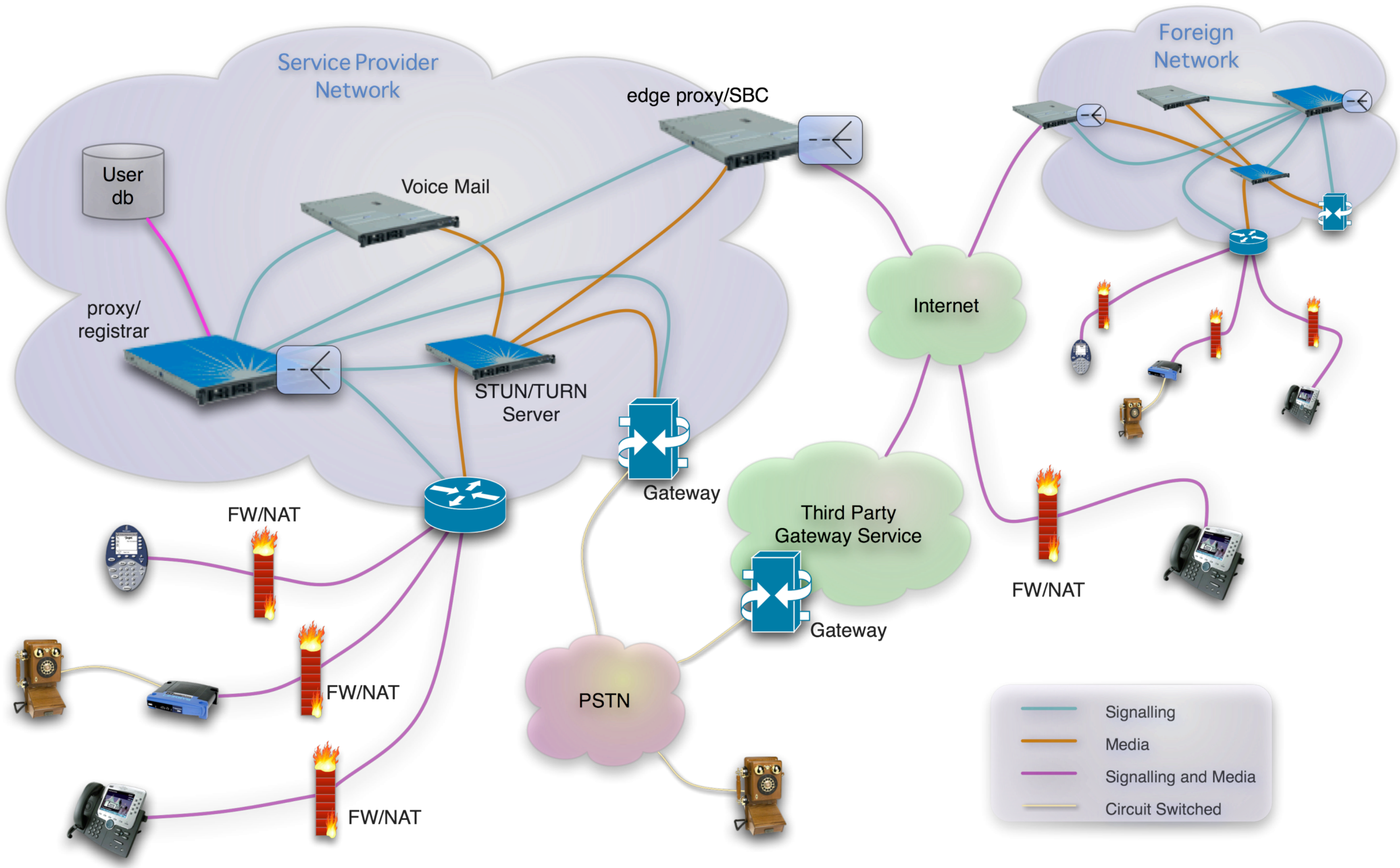
- SIP uses NAPTR/SRV RRs to
  - Select transport protocols and ports
  - Distribute load between elements

```
$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!" .
```

Typical Landscape

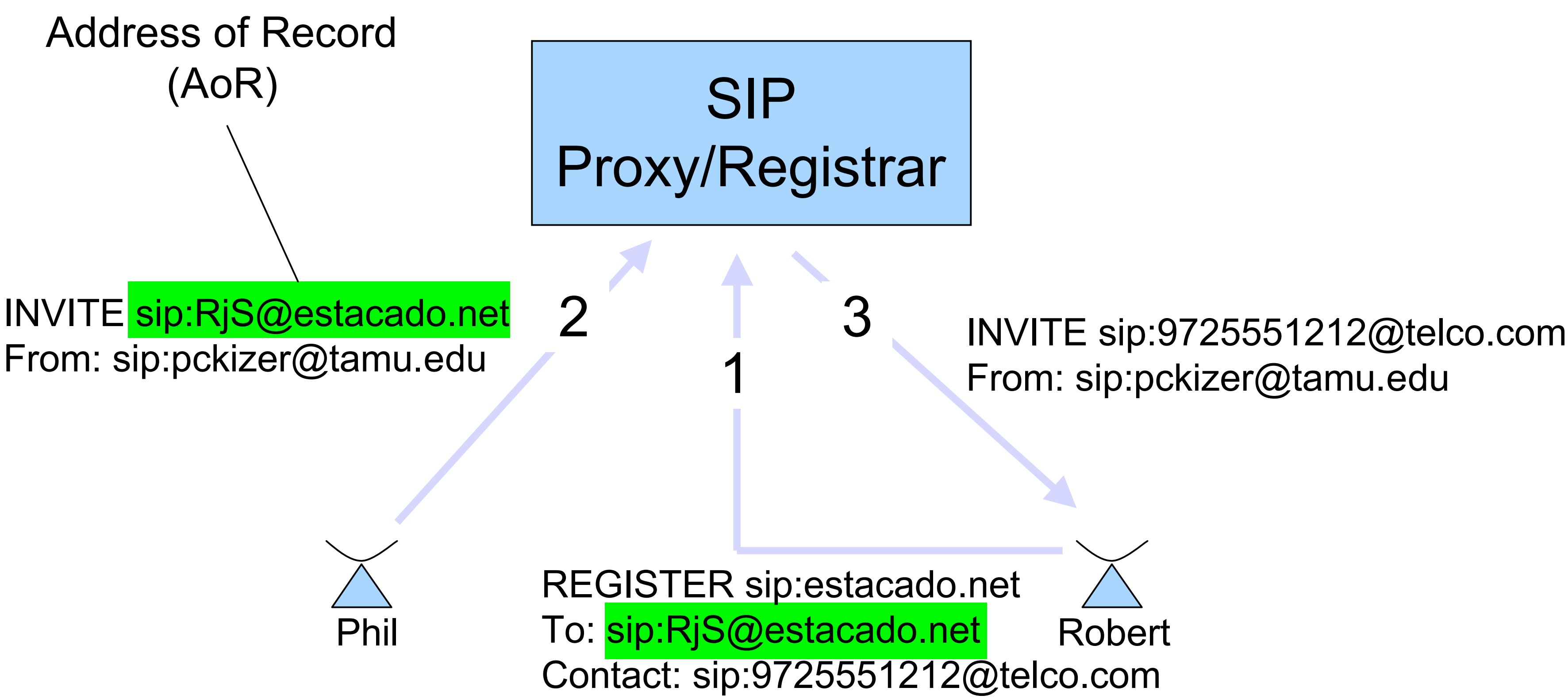


The Landscape

User Agents (end-user endpoints)

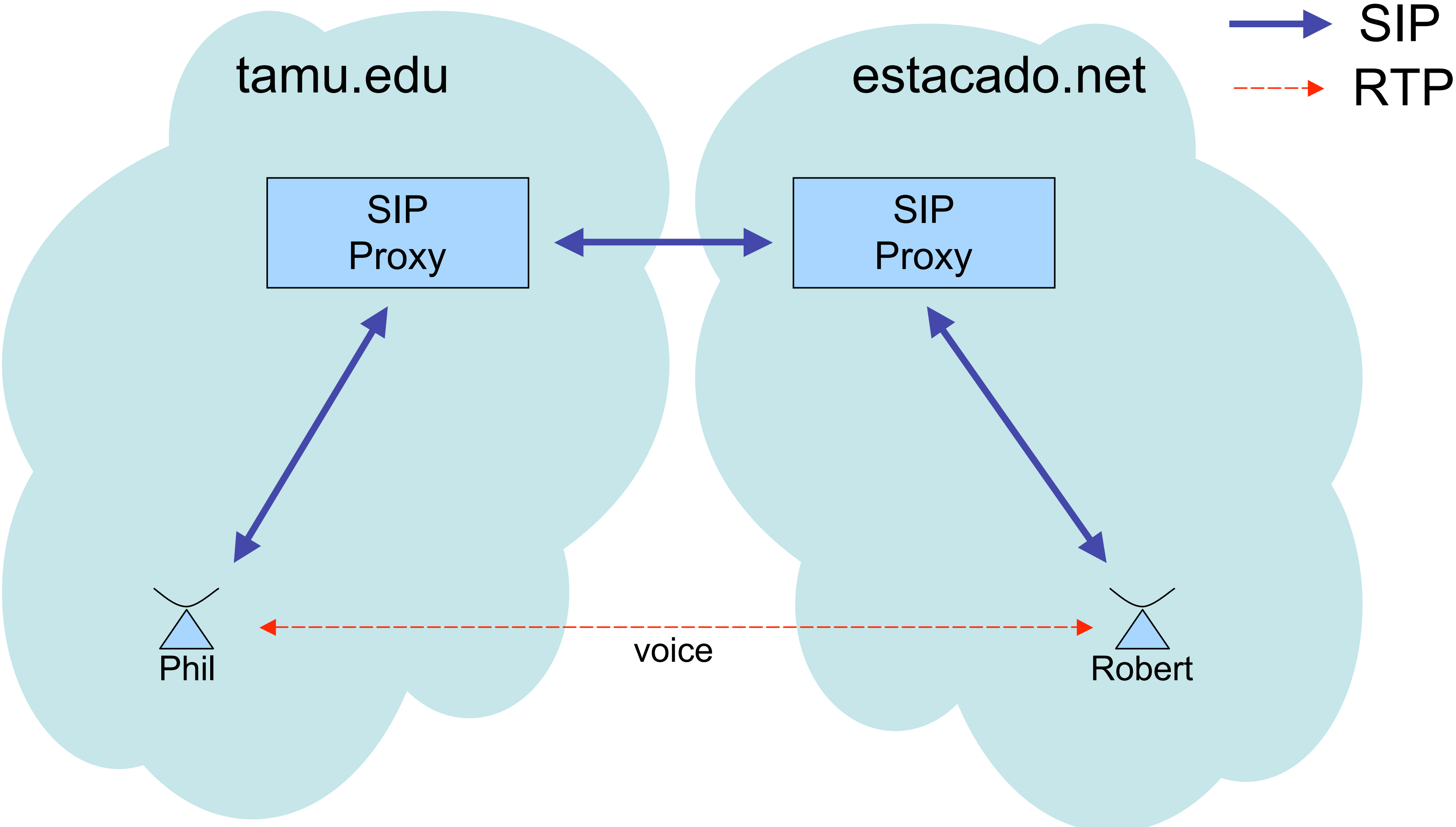
- ATA (Analog Terminal Adaptor)
  - Connects a legacy analog telephone to a VoIP system
- Hard Phone
  - Looks like a phone, acts like a phone (and more) but has an ethernet port instead of an analog RJ11 jack
- Soft Client
  - Programs that run on general purpose PCs

The Landscape
Rendezvous using SIP



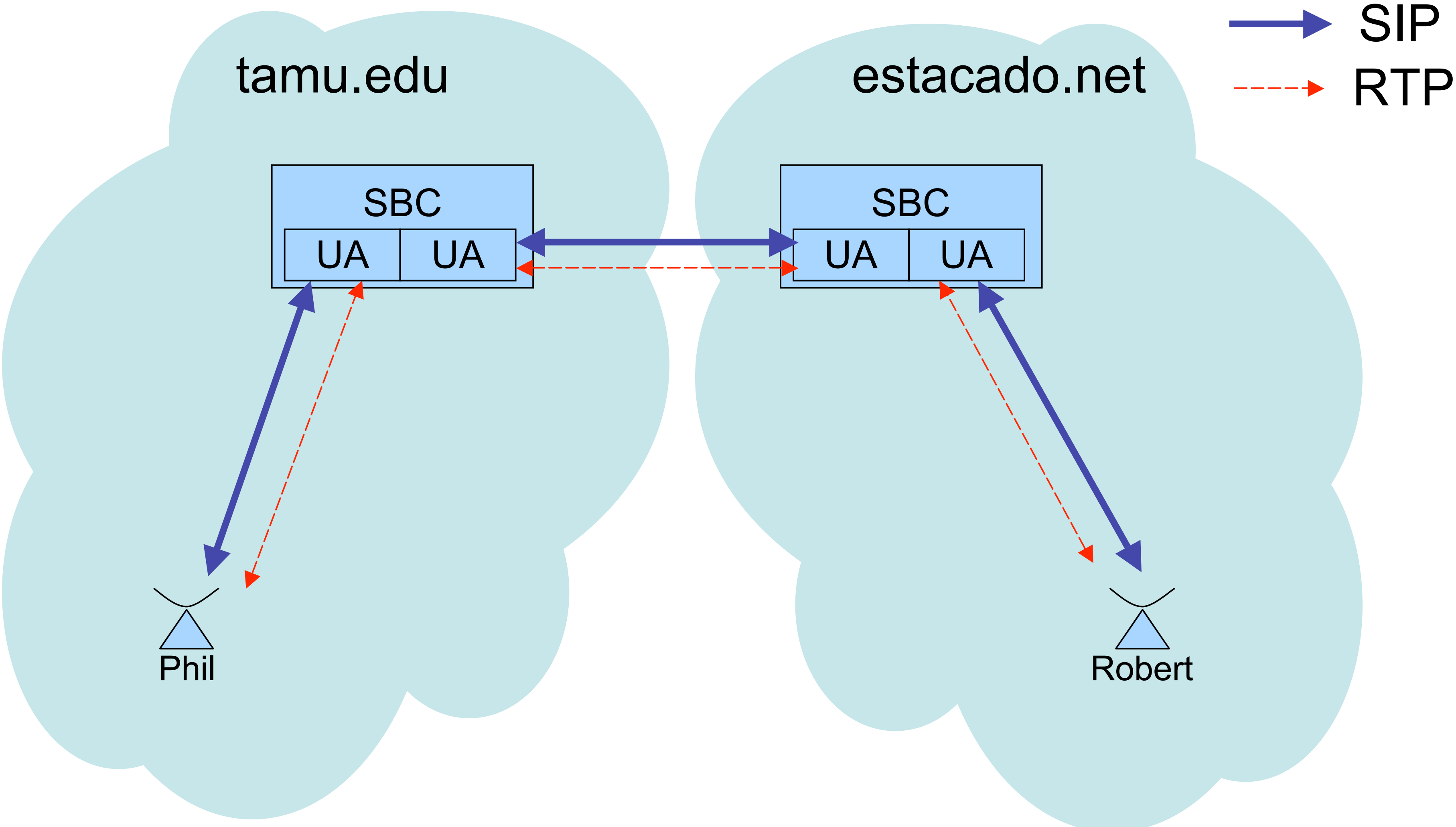
The Landscape

Trapezoid model



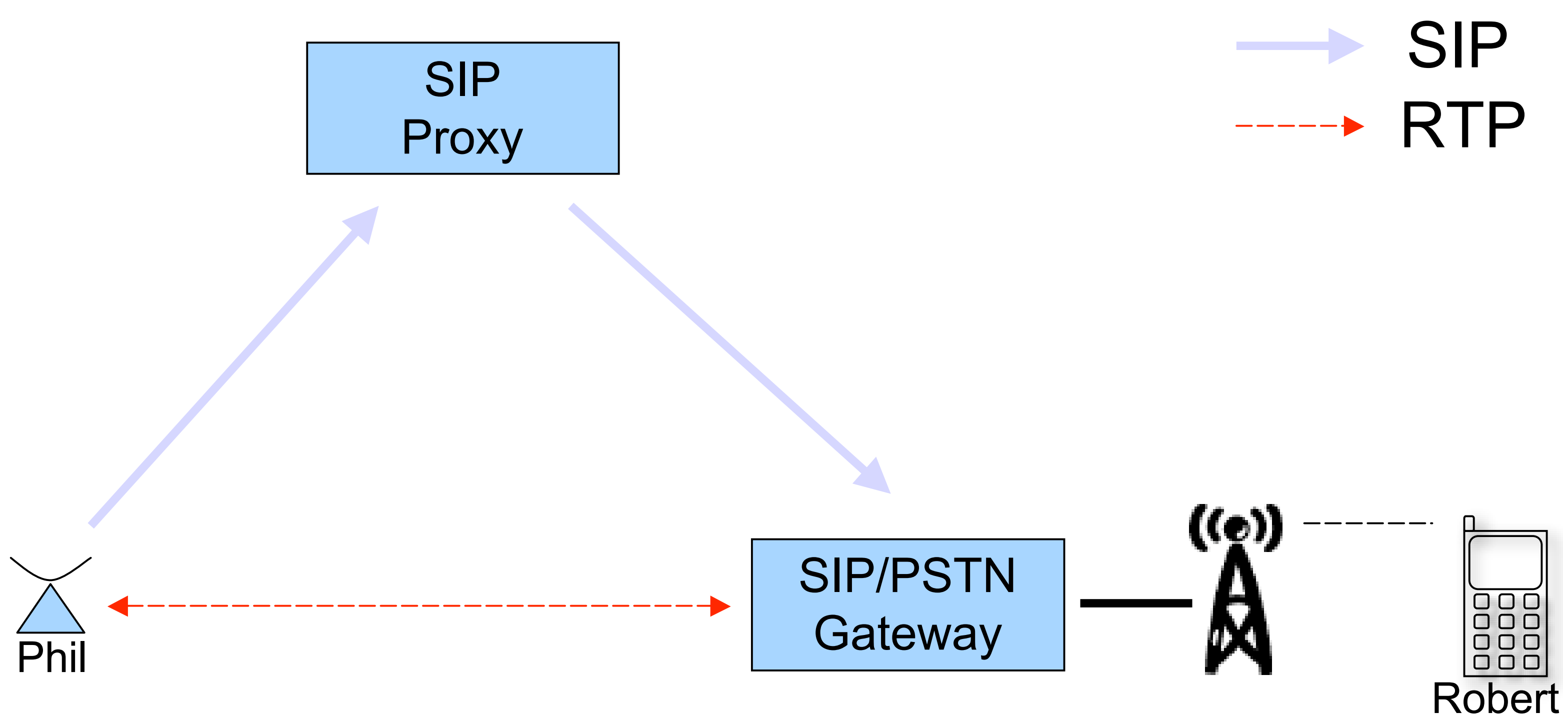
Ecosystem Members

Session Border Controllers



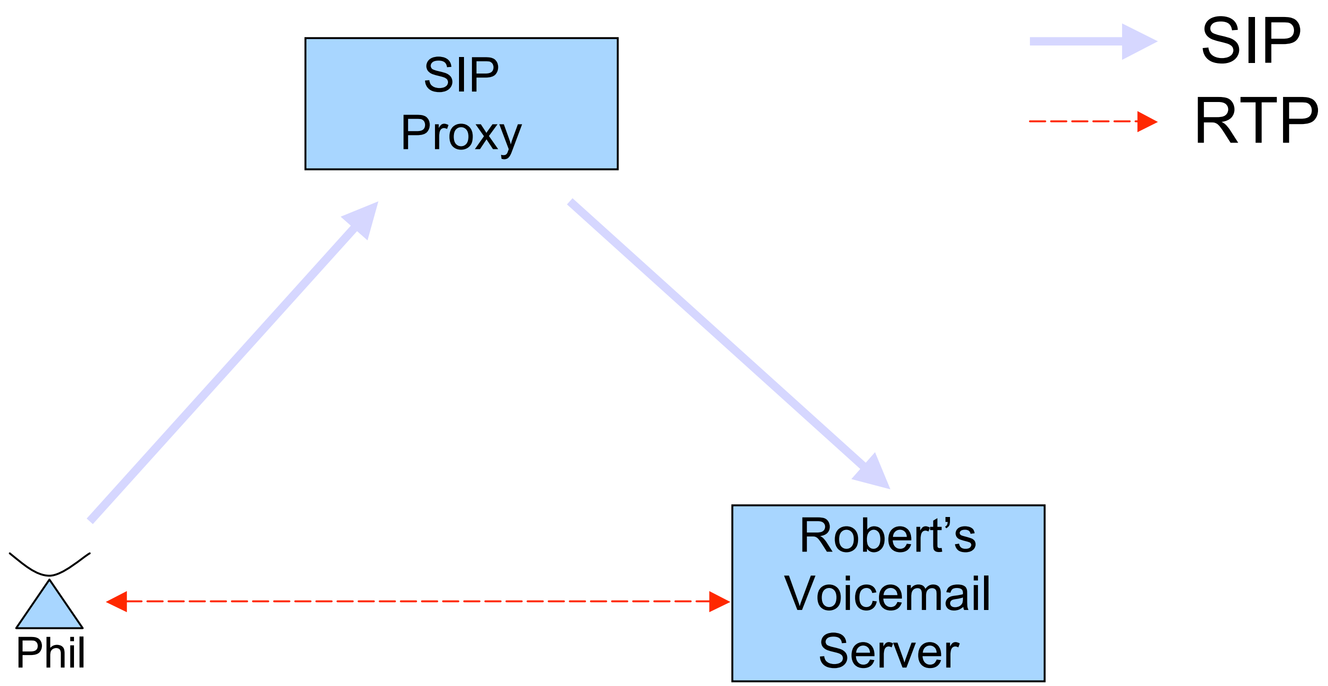
Ecosystem Members

PSTN Gateways



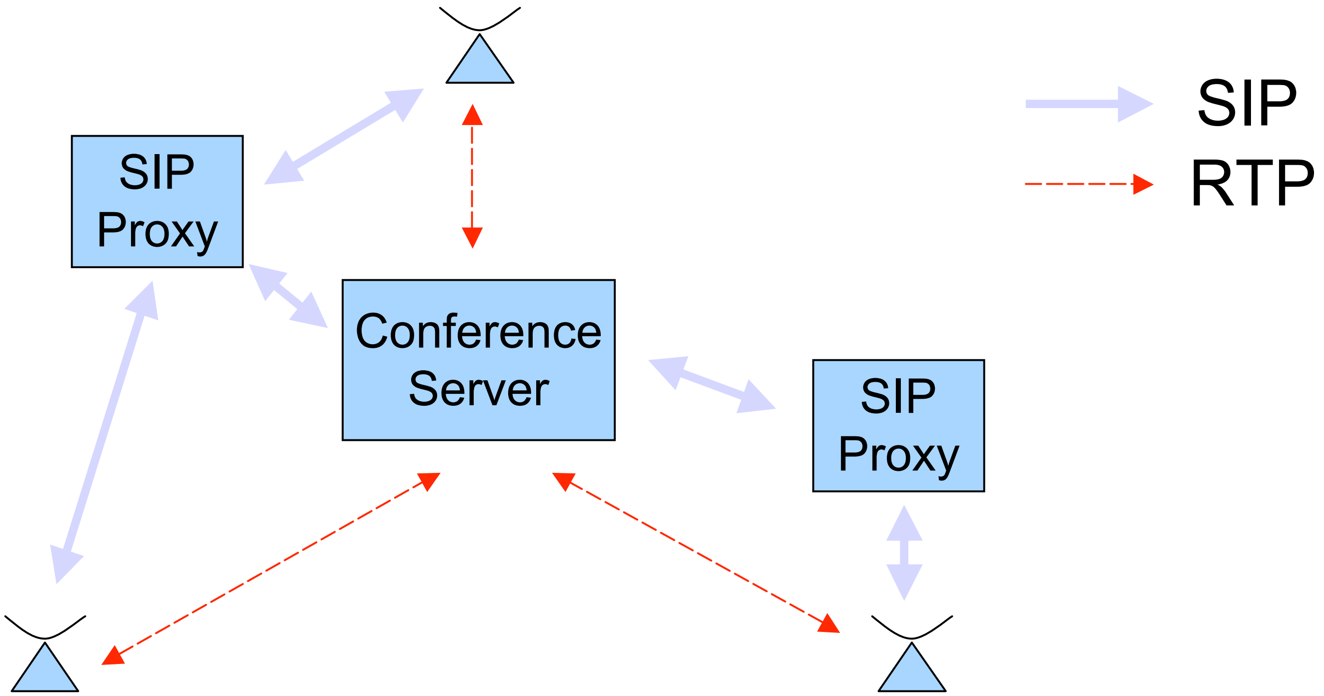
Ecosystem Members

Voice Mail / IVR systems



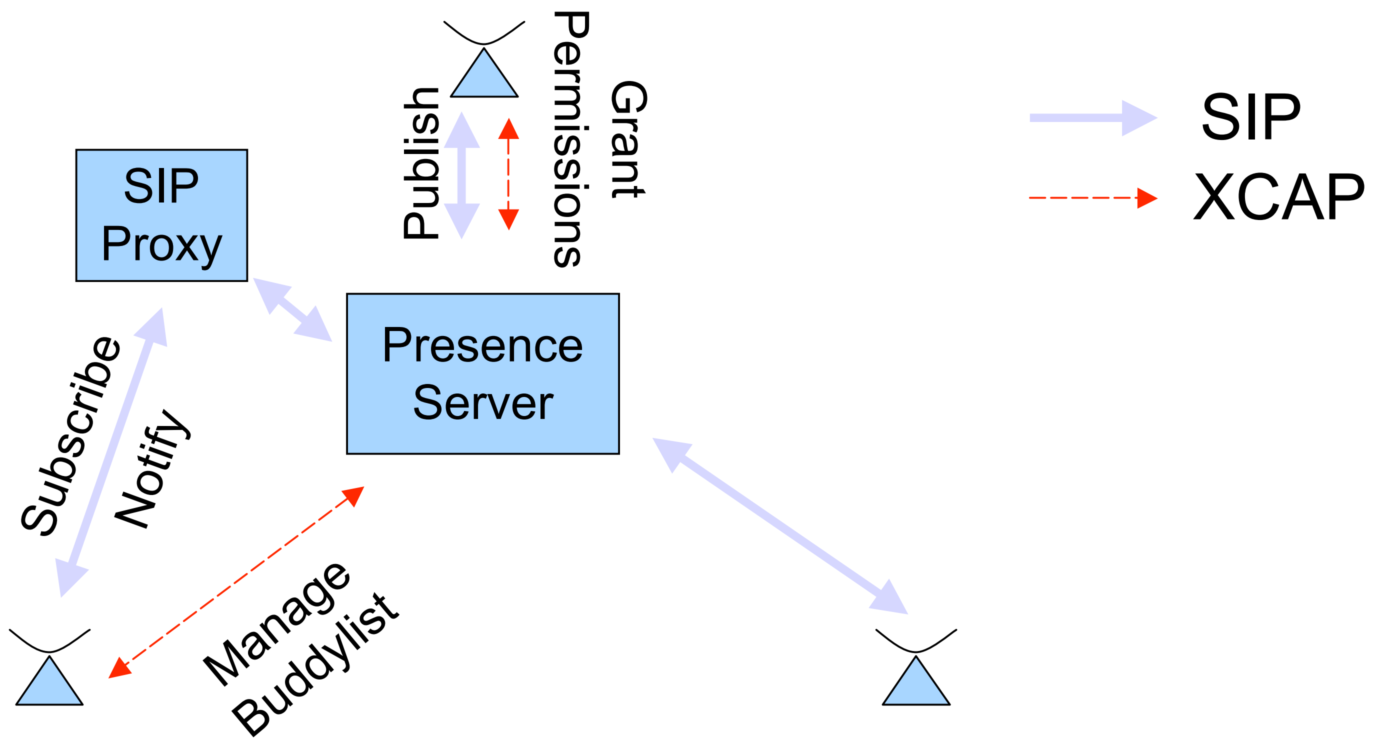
Ecosystem Members

Conference Servers

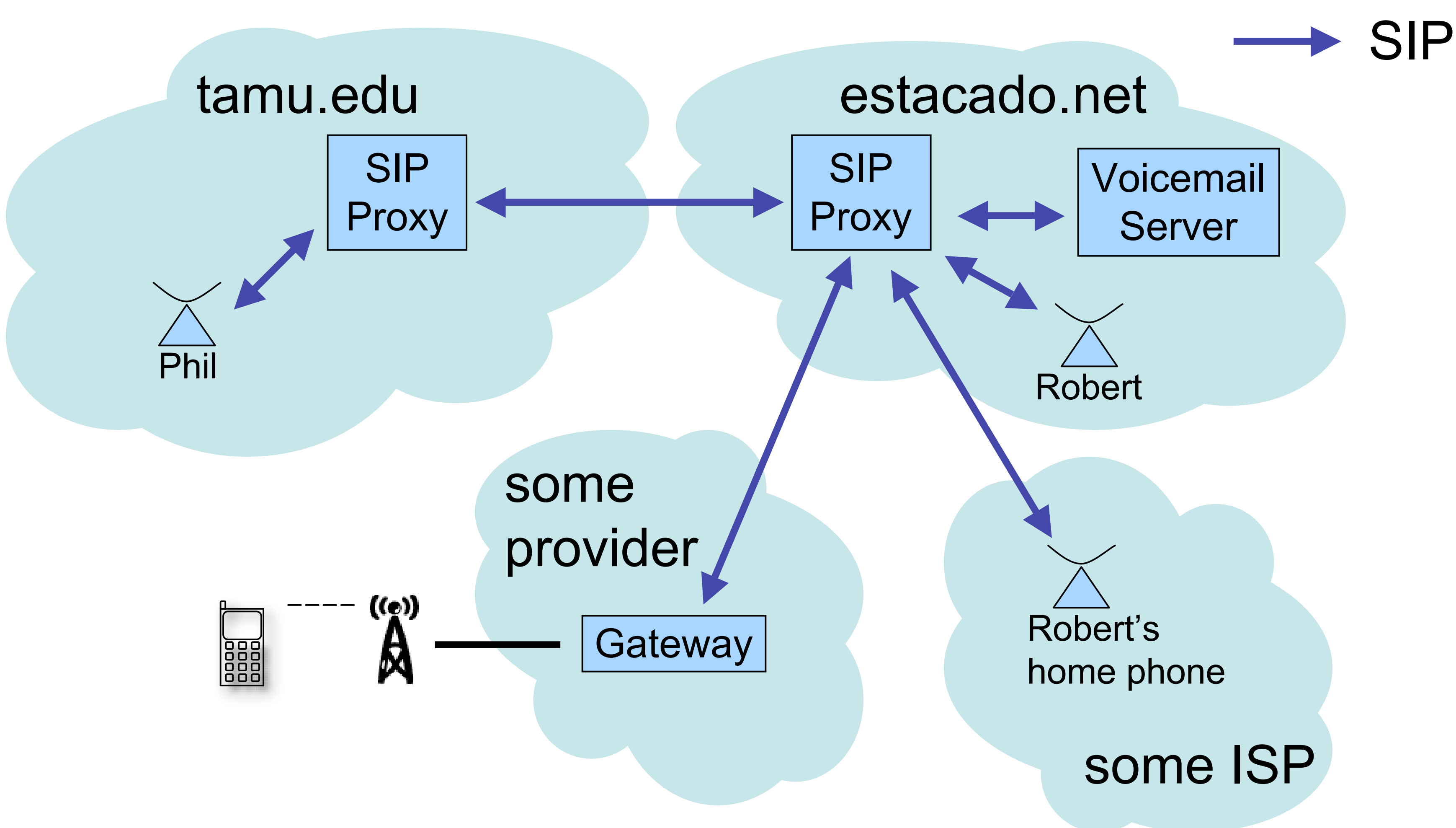


Ecosystem Members

Presence Servers



Forking


SIP
tamu.edu
SIP Proxy
Phil
estacado.net
SIP Proxy
Voicemail Server
Robert
some provider
Gateway
Robert's home phone
some ISP

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
  - "SIP: Understanding the Session Initiation Protocol, 2nd Edition", A. Johnston
  - "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