VoIP – Catalyst for Disruption

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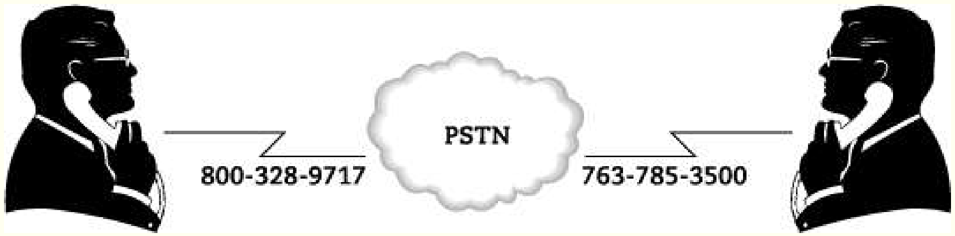
Outline

- Technology Introduction
  - History
  - What can we change?
  - Model
    - Coding
    - Marshalling
    - Session control
  - SIP Example
  - Location
    - ENUM
    - LDAP
- Disruption
  - SIP.edu
  - fwdOUT
- Resources

Traditional Voice Networks

- Voice/fax is sent over dedicated circuits
- Voice/fax calls routed using telephone numbers
- Telephone switches know how to get to every public phone number



Source: Multitech

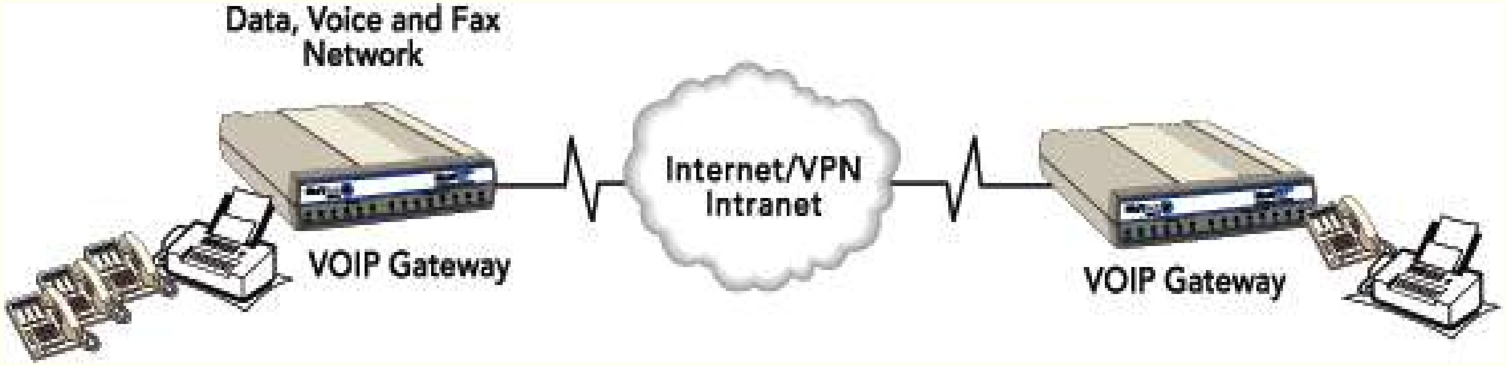
How Data Networks Work

- Data is sent in packets
- IP is the standard protocol
- Data is routed using IP addresses
- Routers know how to get to any public IP address, or at least how to ask

Source: Multitech

What is Voice over IP (VOIP)?

- Voice/fax are converted to IP packets
- Voice/fax/data MAY share the same IP network
- Voice/fax "ride free" on the data network
- Public or Private network using Internet Technology



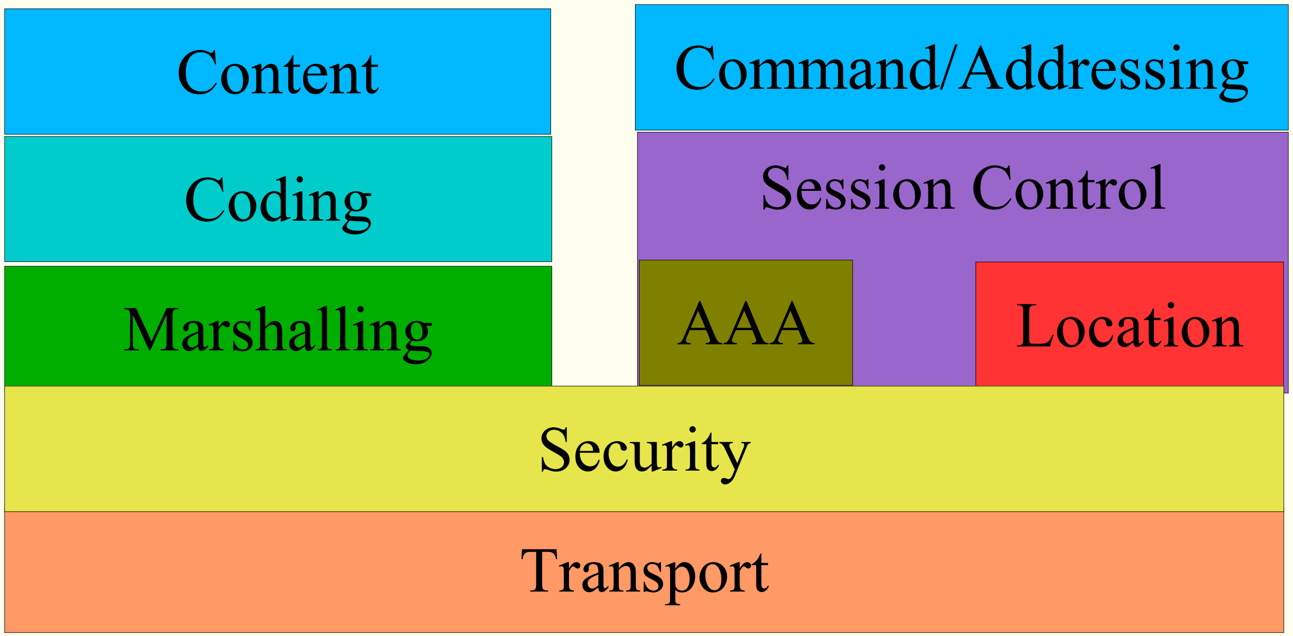
- Soft Phone
- IP Phone
- ATA
- Gateway

Original Source: Multitech

Many Mutable Characteristics of xoIP

- Device & Controlling terminal
- Time Shifting and Scheduling
- Mode
  - Text, voice, video images and streams, presence
- Fidelity
- Transport
  - DSL, OC, WiFi, 3G, cellular…
- Routing
- Number of session participants
  - Spatial Cues
  - Cocktail Party
- CRM/ Directory Integration
  - Single Agent
- Presence Enabled Features
  - Cooperating Agents

Layered Model of VoIP



(Preliminary)

Control and media planes

Alphabet Soup

- VoIP
- VON
- SIP
- ENUM
- CODEC
- SIMPLE *SIP for Instant Messaging and Presence Leveraging Extensions;*
- IETF

- SIP Session Initiation Protocol
- RFC
- Soft Phone
- UA SIP User Agent
- URI Uniform Resource Identifier;
- VoIP Voice over IP (Internet Protocol)
- Gateway

VoIP Media Encoding

CODEC

– the **CO**mpression and **DEC**ompression of audio information.

- G.711 - Pulse code modulation (PCM) on a 64 kbps channel.
- G.722 - 7 kHz audio-coding within 64 kbit/s
- G.726 - 40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)
- G.729 - Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)

VoIP Content Marshalling

- Most Common approach is 'RTP'

  (RFC 3550 - RTP: A Transport Protocol for Real-Time Applications)

  - Delivery – not 'reliable'
  - Reordering
  - Packet Loss
  - RTCP for quality monitoring

- Jitter Management

VoIP Session Control

Who (or what) is communicating

- with whom ?
- at what time?
- With what media types and coding?
- And, will you answer?

- Protocols:
  - **SIP**
  - H323
  - MGCP/MEGACO/H248

Basic SIP Call Architecture

1.User A enters user B's number. User A's SIP client sends an INVITE to FWD.

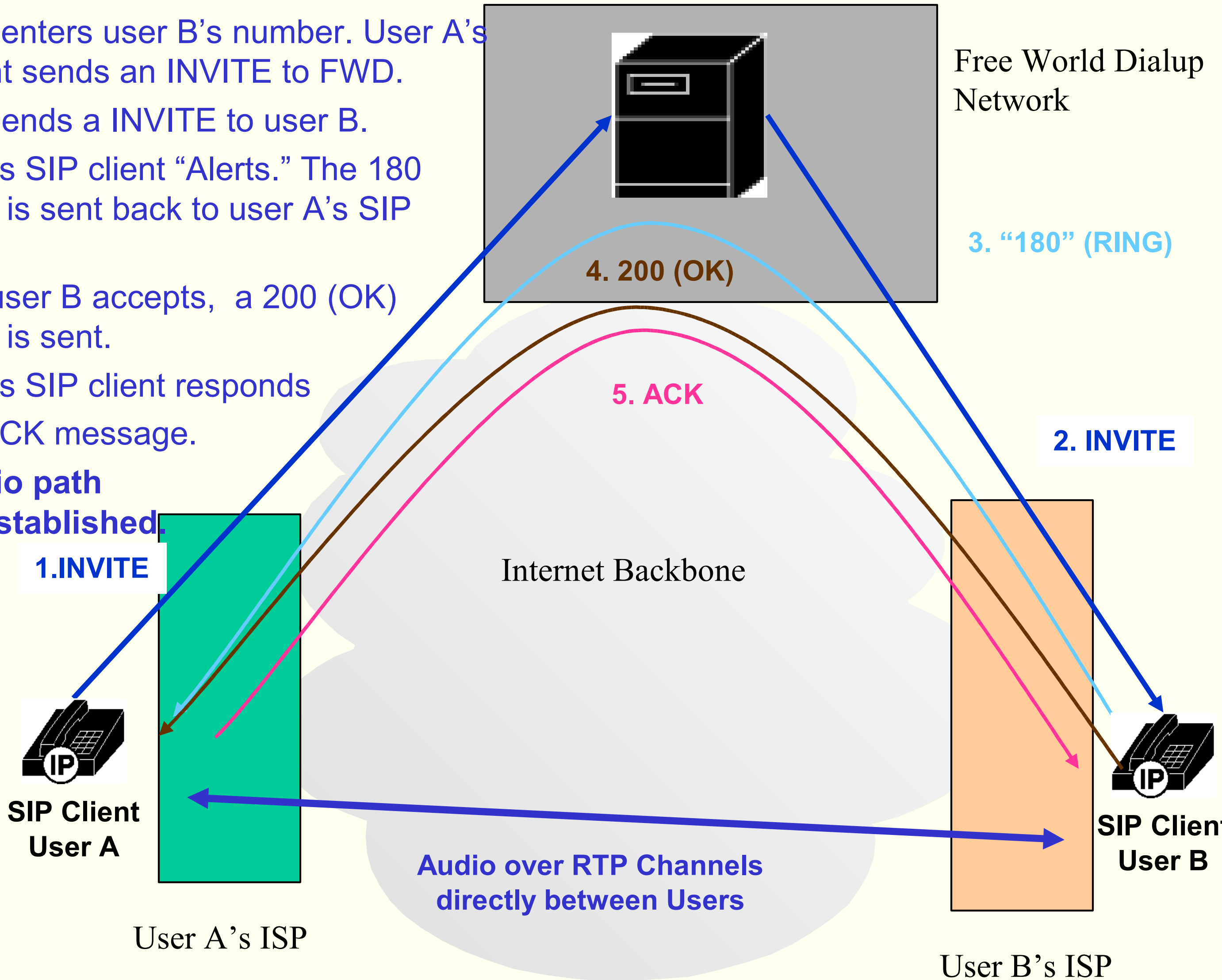
2. FWD sends a INVITE to user B.

3.User B's SIP client "Alerts." The 180 message is sent back to user A's SIP client.

4.When user B accepts, a 200 (OK) message is sent.

5.User A's SIP client responds with an ACK message.

The Audio path is now established.



SIP Session Start

INVITE sip:14551@fwd.pulver.com SIP/2.0
Via: SIP/2.0/UDP 64.81.202.47:5070
From: Ed Guy <sip:19020@fwd.pulver.com:5070>
;tag=791606717
To: <sip:14551@fwd.pulver.com>
Contact: <sip:19020@64.81.202.47:5070>
Call-ID: 8071FDF0-F384-4260-8E53-
824528212BFC@192.168.1.127
CSeq: 19171 INVITE
Content-Type: application/sdp
Content-Length: 269

v=0
o=19020 1090725218 1090725218 IN IP4 64.81.202.47
s=xPhone
c=IN IP4 64.81.202.47
t=0 0
m=audio 8000 RTP/AVP 4 0 8 3 101
a=rtpmap:4 G723/8000
a=rtpmap:0 pcmu/8000
a=rtpmap:8 pcma/8000
a=rtpmap:3 gsm/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

IP/2.0 100 Trying
Via: SIP/2.0/UDP 64.81.202.47:5070
From: Ed Guy <sip:19020@fwd.pulver.com:5070>;tag=791606717
To: <sip:14551@fwd.pulver.com:5060>
Call-ID: 8071FDF0-F384-4260-8E53-824528212BFC@192.168.1.127
CSeq: 19171 INVITE
Content-Length: 0

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 64.81.202.47:5070
From: Ed Guy <sip:19020@fwd.pulver.com:5070>;tag=791606717
To: <sip:14551@fwd.pulver.com:5060>;tag=846932532
Call-ID: 8071FDF0-F384-4260-8E53-824528212BFC@192.168.1.127
CSeq: 19171 INVITE
Record-Route:
<sip:14551@192.246.69.223:5060;maddr=192.246.69.223>,<sip:14551@192.246.69.223:5060;maddr=192.246.69.223>
Contact: sip:63.175.28.50:5060
User-Agent: eDial Server
Content-Length: 0

SIP Session Start (2)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 64.81.202.47:5070
From: Ed Guy<sip:19020@fwd.pulver.com:5070>
;tag=791606717
To:
<sip:14551@fwd.pulver.com:5060>;tag=8469325
32
Call-ID: 8071FDF0-F384-4260-8E53-
824528212BFC@192.168.1.127
CSeq: 19171 INVITE
Record-Route: <sip:14551@192.246.69.223:5060
;
maddr=192.246.69.223>,<sip:14551@192.246.69.
223:5060;maddr=192.246.69.223>
Contact: sip:63.175.28.50:5060
User-Agent: eDial Server
Content-Type: application/sdp
Content-Length: 207

v=0
o=edial_ivr 846932532 846932534 IN IP4 63.175.28.50
s=phone-call
c=IN IP4 63.175.28.50
t=0 0
m=audio 12002 RTP/AVP 0 101
a=rtpmap:0 pcmu/8000/1
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

ACK sip:14551@192.246.69.223:5060 SIP/2.0
Via: SIP/2.0/UDP 64.81.202.47:5070
From: Ed Guy <sip:19020@fwd.pulver.com:5070>;tag=791606717
To: <sip:14551@fwd.pulver.com:5060>;tag=846932532
Contact: <sip:19020@64.81.202.47:5070>
Route: <sip:14551@192.246.69.223:5060 ;
maddr=192.246.69.223>,<sip:63.175.28.50:5060>
Call-ID: 8071FDF0-F384-4260-8E53-824528212BFC@192.168.1.127
CSeq: 19171 ACK
Max-Forwards: 70
Content-Length: 0

BYE sip:14551@192.246.69.223:5060 SIP/2.0
Via: SIP/2.0/UDP 64.81.202.47:5070
From: Ed Guy <sip:19020@fwd.pulver.com:5070>;tag=791606717
CSeq: 19172 BYE
…

SIP/2.0 200 OK
Via: SIP/2.0/UDP 64.81.202.47:5070
CSeq: 19172 BYE
...

ACK sip:14551@192.246.69.223:5060 SIP/2.0
Via: SIP/2.0/UDP 64.81.202.47:5070
From: Ed Guy <sip:19020@fwd.pulver.com:5070>;tag=791606717
To: <sip:14551@fwd.pulver.com:5060>;tag=846932532
Contact: <sip:19020@64.81.202.47:5070>
CSeq: 19172 ACK
…

NAT Problem – Private Addresses

When SIP Clients are Behind NAT:

1) Addresses and Ports are not routable
2) Frequent Communication needed to keep open

```
INVITE sip:14551@fwd.pulver.com SIP/2.0
Via: SIP/2.0/UDP 192.168.1.26:5070
From: Ed Guy <sip:19020@fwd.pulver.com:5070>;tag=791606717
To: <sip:14551@fwd.pulver.com>
Contact: <sip:19020@192.168.1.26:5070>
Call-ID: 8071FDF0-F384-4260-8E53-824528212BFC@192.168.1.26
CSeq: 19171 INVITE
Content-Type: application/sdp
Content-Length: 269

v=0
o=19020 1090725218 1090725218 IN IP4 192.168.1.26
s=Cisco/4
c=IN IP4 192.168.1.26
t=0 0
m=audio 8000 RTP/AVP 4 0 8 3 101
a=rtpmap:4 G723/8000
a=rtpmap:0 pcmu/8000
a=rtpmap:8 pcma/8000
a=rtpmap:3 gsm/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

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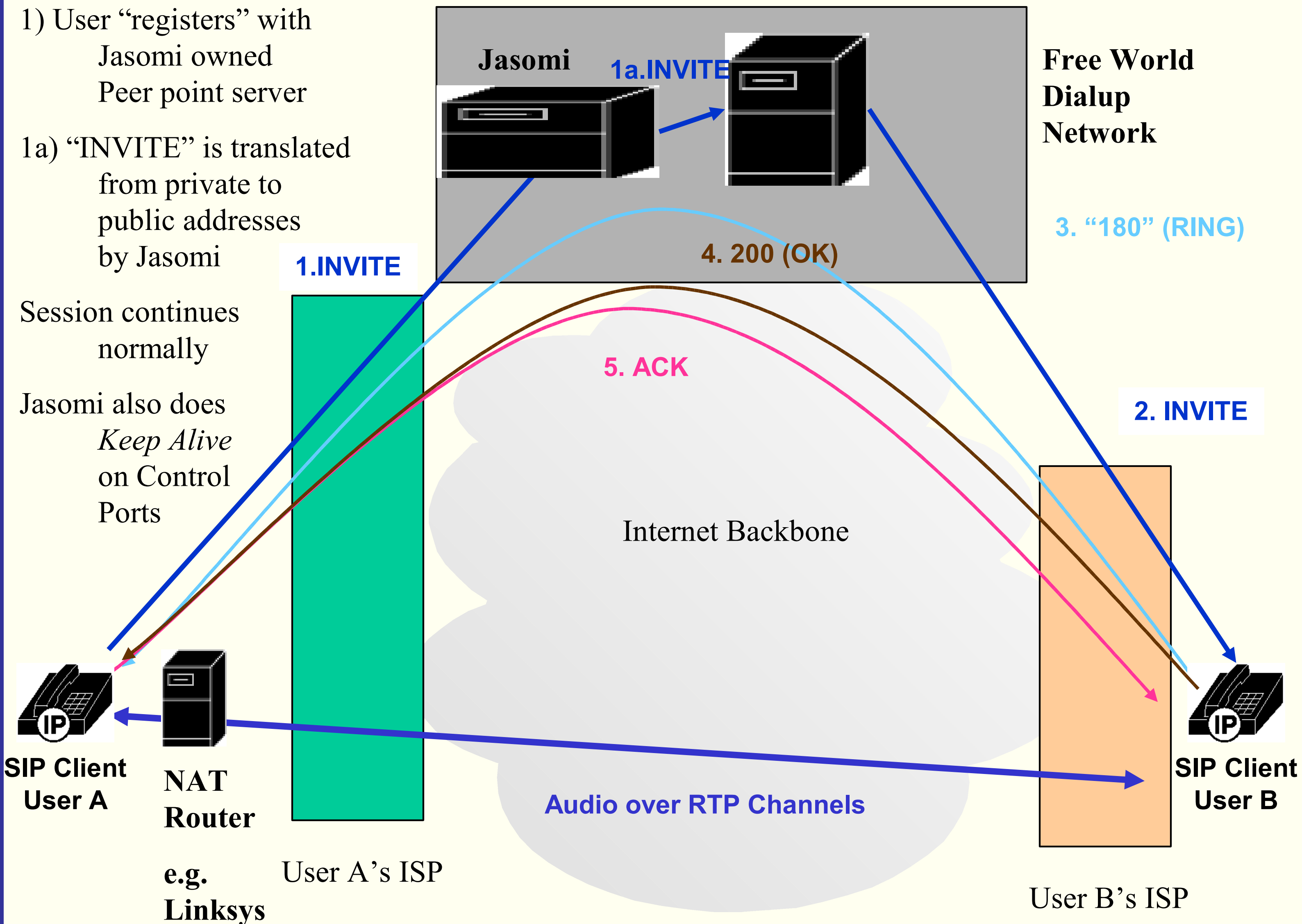
NAT Solution – Jasomi Peer point

1) User "registers" with Jasomi owned Peer point server

1a) "INVITE" is translated from private to public addresses by Jasomi

Session continues normally

Jasomi also does *Keep Alive* on Control Ports



SIP Drivers

- Simplicity
- Rich
  - Media independent
  - Presence
  - IM
- Private club membership not required to build applications
- Media does not need to be determined *a priori*
- Edges MAY propose and enforce call control & media policies

- Open Standard with many vendor Choices

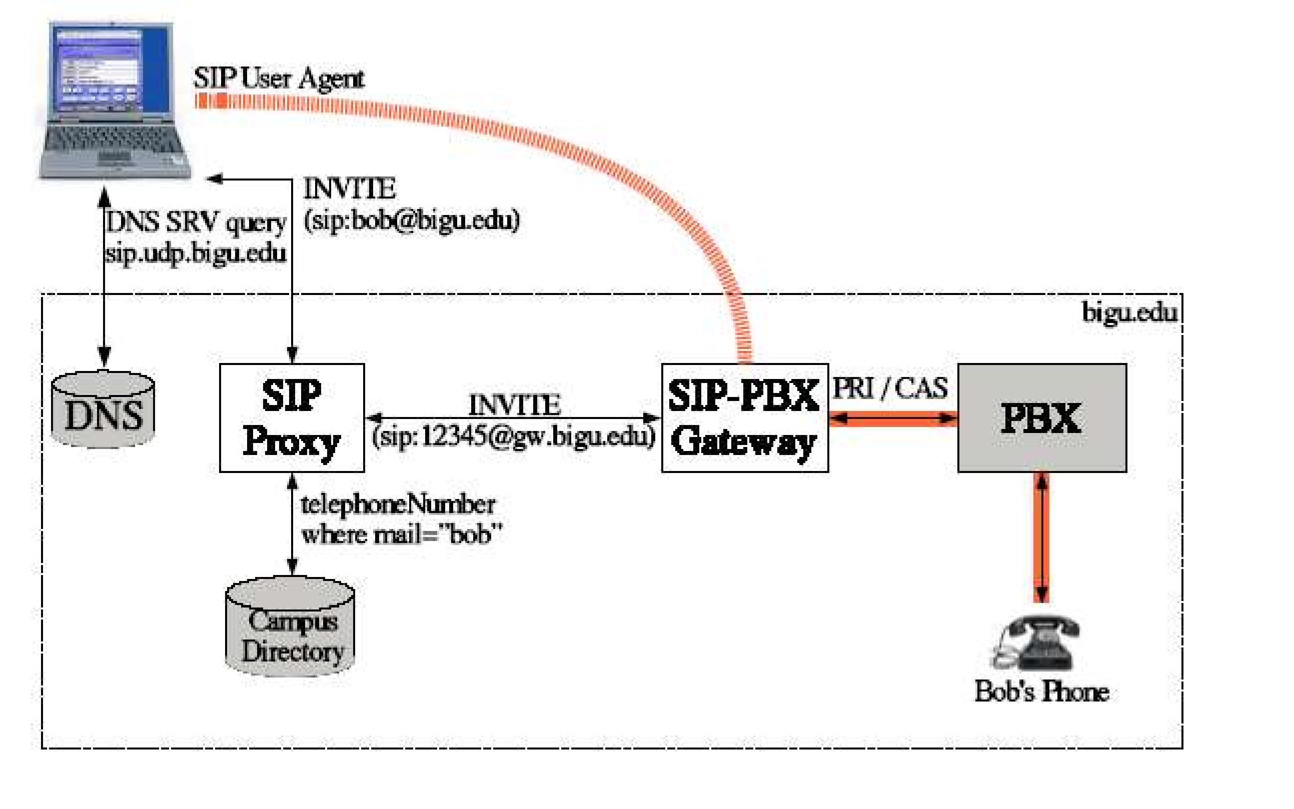
Number Location with ENUM

- Joe Dials +1(315)351.1234
  - Implied or explicit Country Code.
  - Area/City Code
  - Exchange
  - Number
- Convert to form suitable for our favorite distributed data base: DNS
- Lookup 4.3.2.1.1.5.3.5.1.3.1.sipEduEnum.pulver.com
- DNS NAPTR – sip:1234@cs.sunyit.edu

Name Location with LDAP

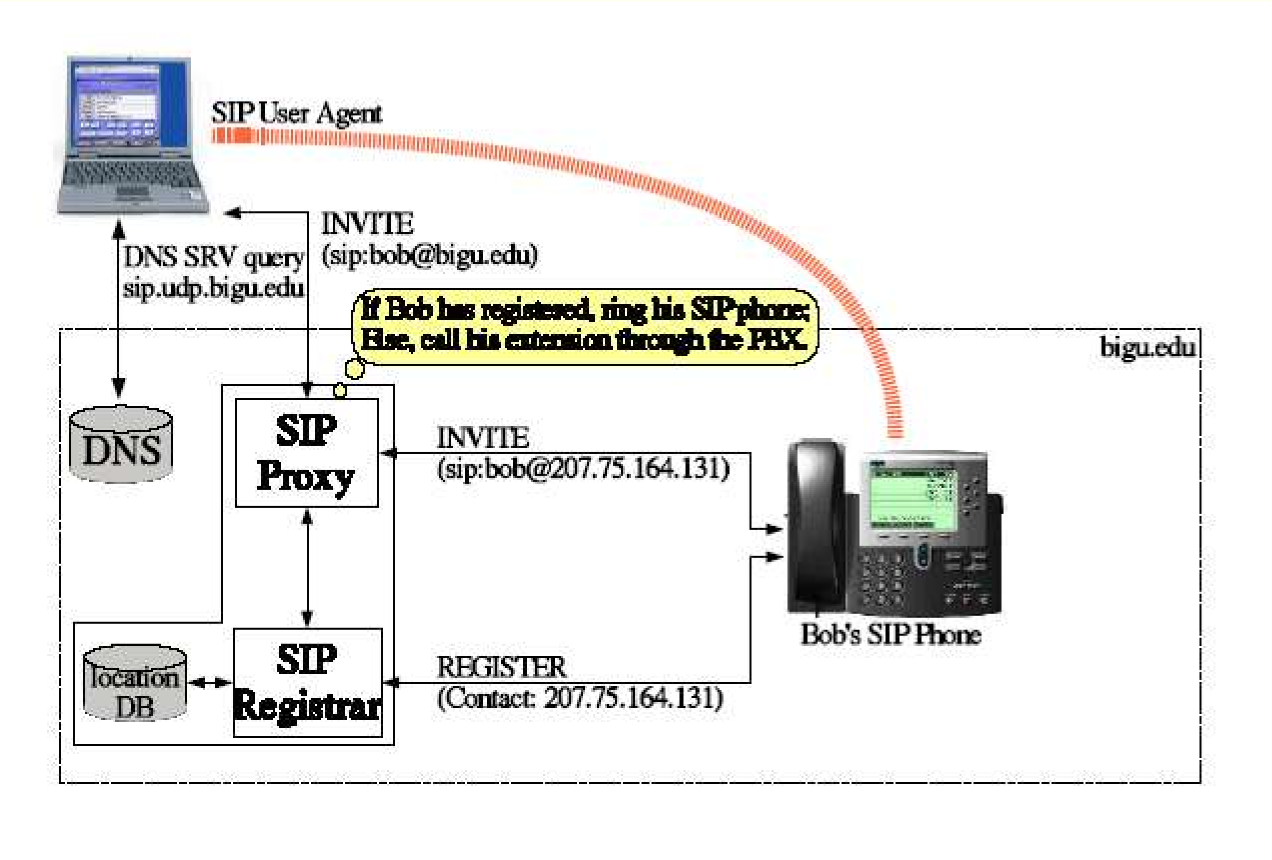
- Joe calls sip:BabsJensen@sunyit.edu
- Caller's service asks where is the server for sunyit.edu??
  - DNS Lookup for SRV record _udp._sip.sunyit.edu
  - Answer: ipbx.sunyit.edu port 5060
  - Call is Routed to ipbx.sunyit.edu
- Ipbx asks Does Babs have a SIP phone?
  - No ;-(
- Ipbx asks what is Bab's line?
  - Use LDAP to Search the student directory
  - 351-1212
- Ipbx finds a gateway to PBX
  - Call completes

SIP.edu USE Case



Source: D Baron. MIT

SIP.edu on a SIP phone



Source: D Baron. MIT

Where's the Money?

- *"It's a race to the bottom [price] and the biggest idiot wins!"*
  - Attributed to Bob Lucky, Telcordia
  - *Beating the other guy's price is not a long term strategy for success.*
- *Current Commercial Services simply replace your phone for less money*
  - *Provide Service Portability.*
  - *Some offer video (40 years after initial introduction!)*
- *The winner will consist of Innovative combinations of*
  - *various rich media*
  - *with intelligent agents*

It's now a social problem

Some Player Services

- .Com
  - 'Over the Top'
    - Vonage
    - Packet 8
    - Broadvoice
    - Voicepulse
  - Incumbents
  - Cable Operators

Drivers:
Access to other markets
Parasite play
Extend network

PSTN REPLACEMENT
For LE$$

Disruptors

skype

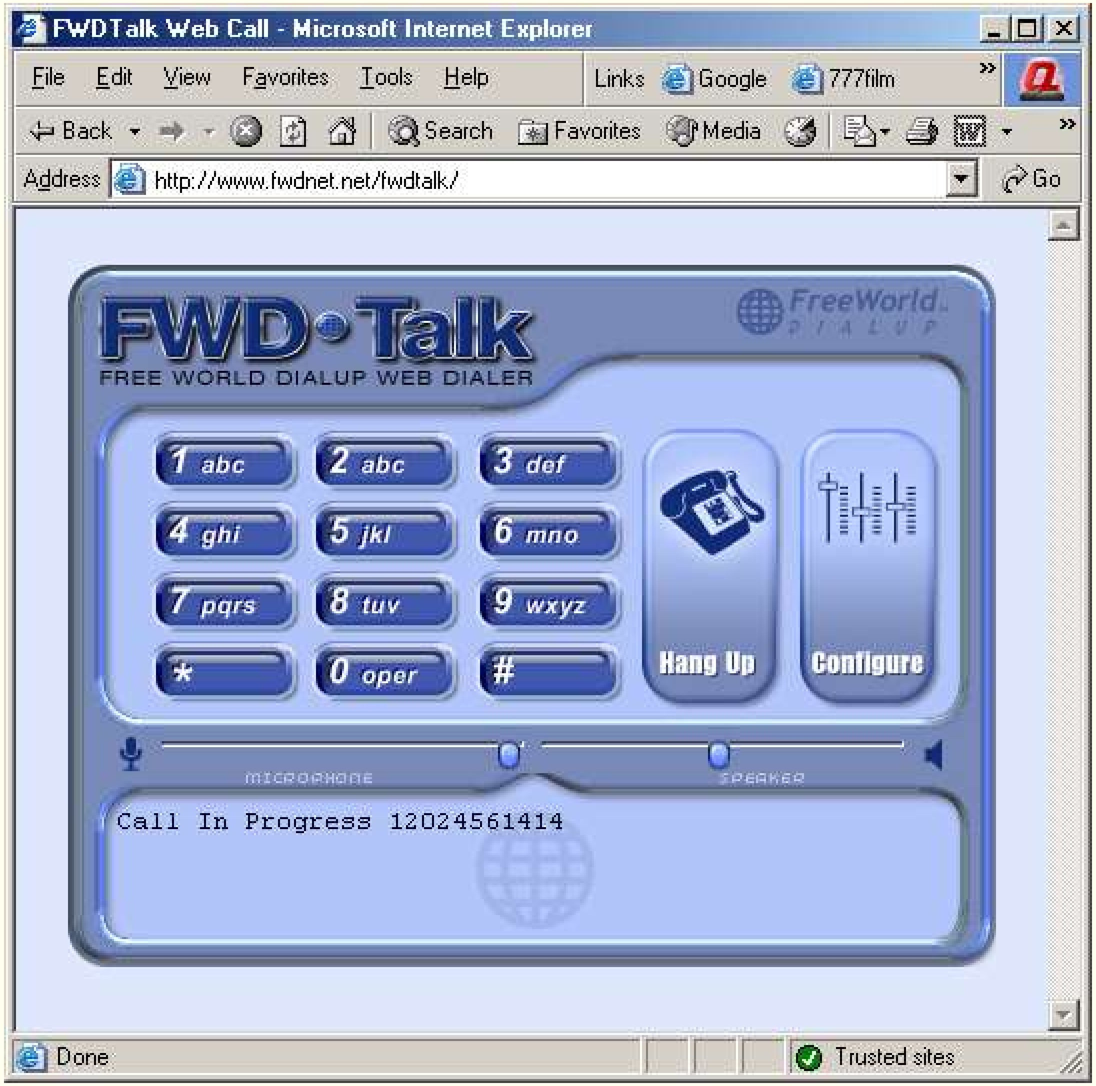
.Org
FWD +
IAXTEL
iptel.org

.Edu
Internet2/SIP.edu

fwdOUT

- Napster for Phone lines!
- Use phone lines of others
  - *Catch*: you need to share, too
- Based on Asterisk open source software
  - Linux
  - Old PC and
  - a special modem.

FWD Talk


FWDTalk Web Call - Microsoft Internet Explorer
File Edit View Favorites Tools Help
Links Google 777film
Back Search Favorites Media
Address http://www.fwdnet.net/fwdtalk/
Go
FWD Talk
FREE WORLD DIALUP WEB DIALER
FreeWorld DIALUP
1 abc
2 abc
3 def
4 ghi
5 jkl
6 mno
7 pqrs
8 tuv
9 wxyz
*
0 oper
#
Hang Up
Configure
MICROPHONE
SPEAKER
Call In Progress 12024561414
Done
Trusted sites

“Stupid” Phone Tricks

- User control
  - Click to dial
  - Advanced call logic – one number, time of day routing.
- IVR
  - Math Quiz in Perl
  - Jeff's blog read to you
- House Control
  - Lights, Status, HVAC, Phones

Online Resources for SIP

- Henning Schulzrinne's SIP Page
  http://www.cs.columbia.edu/sip
- ietf.org and supplementary SIP and SIPPING archives:
  http://www.softarmor.com/sipwg/drafts/
  http://www.softarmor.com/sipping/drafts/
- IP Telephony Web Page
  http://iptel.org
- SIP Forum
  http://www.sipforum.com
- SIP Center
  http://www.sipcenter.com
- SIP Products at Pulver.com
  http://www.pulver.com/sip/products.html
- Books
  - "Internet Communications Using SIP"
    by Henry Sinnreich and Alan B. Johnston, Wiley, 2001.
  - "SIP: Understanding the Session Initiation Protocol,"
    Alan B. Johnston, Artech House, 2001.
  - "SIP Demystified" by Gonzalo Camarillo, McGraw-Hill, 2001.

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IETF Resources on SIP

Finding a draft	http://search.ietf.org/
Finding an RFC	http://www.rfc-editor.org/rfcsearch.html
SIP WG	http://ietf.org/html.charters/sip-charter.html
SIPPING WG	http://ietf.org/html.charters/sipping-charter.html
SIMPLE WG	http://ietf.org/html.charters/simple-charter.html
XCON WG	http://ietf.org/html.charters/xcon-charter.html
MMUSIC WG	http://ietf.org/html.charters/mmusic-charter.html

Thank You

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