VoIP Basics
emil.ivov@sip-communicator.org
How did it all start?
How did it all start?
Encoding Media

- **Pulse Code Modulation** - a digital representation of an analog signal where the magnitude of the signal is sampled regularly at uniform intervals, then quantized to a series of symbols in a digital (usually binary) code.

- PCM has been widely used in:
  - digital telephone systems
  - compact disc red book format
  - Computer systems (wav files).
  - DVD or DVR
  - Many Blu-ray Disc and HD-DVD movies
  - Audio transmission within LANs

- Not used in real-time communication over the Internet due to high bandwidth consumption.
<table>
<thead>
<tr>
<th>Codec</th>
<th>Bit-Rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>G711</td>
<td>64Kbs</td>
</tr>
<tr>
<td>G723.1</td>
<td>6,3Kbs</td>
</tr>
<tr>
<td>G729A</td>
<td>8Kbs</td>
</tr>
<tr>
<td>iLBC</td>
<td>15.2 Kbs</td>
</tr>
<tr>
<td>Speex</td>
<td>variable</td>
</tr>
</tbody>
</table>
Transporting Media - TCP vs. UDP

- Using UDP
  - No way to detect loss.
  - Order of delivery does not necessarily reflect the order of sending

- Using TCP
  - Loss detection
  - Respects order
  - Loss recovery – inefficient for CoIP
    - Retransmission of lost segments – increases jitter
    - Decreasing window size – causes lower bandwidth
So, if both s**k, what do we do?
- Design a new transport protocol
- Design an application protocol that would compensate deficiencies of the transport protocol.

A widespread solution
- Using an application protocol (RTP) over UDP
What next?

... 1100110101100111001011 ...
The basics of IP telephony.

Alice
Address: A
Port: Pa

Bob
Address: B
Port: Pb

network core
(registrars, proxies, ...)

Call:
To: A
Media: B:Pb
The basics of IP telephony.

network core
(registrars, proxies, ...)

Bob
Address: \textit{B}
Port: \textit{Pb}

Alice
Address: \textit{A}
Port: \textit{Pa}

Answer:
To: \textit{B}
Media: \textit{A}:Pa
The basics of IP telephony.

Bob
Address: B
Port: Pb

Alice
Address: A
Port: Pa

network core
(registrars, proxies, ...)

... MEDIA ...

VoIP Basics, X/Str, Oct 2, 2008
emil.ivov@sip-communicator.org
Signaling

- Say why it is important to separate signalling from data.
- H.323
- MGCP
- SIP
- XMPP/Jingle

- Other signaling protocols:
  - IAX
  - Skype, ICQ, Yahoo
MGCP - Media Gateway Control Protocol

- Defines protocols for control of gateways that handle media flow conversion
  - Example: transcoding analogous voice (PSTN) into digital signal IP.
- This approach is based on the notion of separating signaling from multimedia support.
Media Gateway Controller

PSTN

PCM Flow 64 Kb/s

Media Gateway

IP

RTP Flow
H.323

- The ITU solution for video conferencing on data networks: IP, ATM, ...
  - Strongly inspired by the RNIS H320 standards for conferencing.
  - Multiple ITU PSTN low band protocols are employed by H.323
    - Q.931
    - Supplementary services coming from Q.932
• ITU Recommendations
  • Version 1 1996: Video telephony system for LANs with no QoS
  • Version 4 November 2000

• Pros and Cons:
  • Compatible with H320 (PSTN)
  • High complexity, difficult to adapt to the Internet (Firewall, NATs, QoS)
H.323 Entities

Terminal

Gateway

MCU

Terminal

Gatekeeper

IP Network

RTC RNIS
Architectures and protocols

RAS : Registration Admission Status
- Gatekeeper registration

Q.931 : signaling call
- Allows opening an H.245 connection

H.245 : control call
- Information exchange (codec, address, RTP and RTCP port numbers)
- Activates channels
<table>
<thead>
<tr>
<th>Port</th>
<th>Type</th>
<th>Used for</th>
</tr>
</thead>
<tbody>
<tr>
<td>389</td>
<td>static – TCP</td>
<td>ILS Registration (LDAP)</td>
</tr>
<tr>
<td>1300</td>
<td>static – TCP</td>
<td>H.235 Secure Signaling</td>
</tr>
<tr>
<td>1503</td>
<td>static – TCP</td>
<td>T120</td>
</tr>
<tr>
<td>1718</td>
<td>static – UDP</td>
<td>Gatekeeper Discovery</td>
</tr>
<tr>
<td>1719</td>
<td>static – UDP</td>
<td>Gatekeeper RAS</td>
</tr>
<tr>
<td>1720</td>
<td>static – TCP</td>
<td>Q.931 Call Setup</td>
</tr>
<tr>
<td>1024-65553 5</td>
<td>dynamic – TCP</td>
<td>H245 Control Channel</td>
</tr>
<tr>
<td>1024-65553 5</td>
<td>dynamic – UDP</td>
<td>RTP/RTCP – Audio/Video Streams</td>
</tr>
</tbody>
</table>
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emil.ivov@sip-communicator.org

A basic 2-party call

Q.931 port TCP 1720
H.245 port TCP > 1024
RTP – G7xx
RTP – H26x
RTP – G7xx
RTP – H26x

UDP
A Single Administrative Domain

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Multiple Administrative Domains

Clearing House

Packet Network

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XMPP/Jingle
The Session Initiation Protocol

SIP
The Session Initiation Protocol
(Some of the People Behind It)

Henning Schulzrinne
Department of Computer Science
Columbia University, New York, USA

Jonathan Rosenberg
Cisco Systems
A Very Basic SIP Call Flow

Alice

Bob

INVITE
100 Trying
180 Ringing
200 OK
ACK

... MEDIA ...

BYE
OK

CoIP with SIP. Rennes, November 2007
Emil Iovov, SIP Communicator
Example SIP Request

INVITE sip:barbara@b.com SIP/2.0
Via: SIP/2.0/UDP 10.43.122.3;branch=1
From: sip:alice@a.com;tag=4ad340f
To: sip:barbara@b.com
Contact: <sip:alice@10.43.122.3>
Call-ID: 1874630@10.43.122.3
Cseq: 12442 INVITE

v=0
o=user 14341433 14341433 IP4 10.43.122.3
s=.
t=0 0
c=IN IP4 10.43.122.3
m=audio 13222 RTP/AVP 0
a=rtpmap:0 PCMU/8000
### Example SIP Request

<table>
<thead>
<tr>
<th>Request line</th>
<th>INVITE sip:<a href="mailto:barbara@b.com">barbara@b.com</a> SIP/2.0</th>
</tr>
</thead>
<tbody>
<tr>
<td>Headers</td>
<td>Via: SIP/2.0/UDP 10.43.122.3;branch=1</td>
</tr>
<tr>
<td></td>
<td>From: sip:<a href="mailto:alice@a.com">alice@a.com</a>;tag=4ad340f</td>
</tr>
<tr>
<td></td>
<td>To: sip:<a href="mailto:barbara@b.com">barbara@b.com</a></td>
</tr>
<tr>
<td></td>
<td>Contact: <a href="">sip:alice@10.43.122.3</a></td>
</tr>
<tr>
<td></td>
<td>Call-ID: 1874630@10.43.122.3</td>
</tr>
<tr>
<td></td>
<td>Cseq: 12442 INVITE</td>
</tr>
<tr>
<td>Empty line</td>
<td></td>
</tr>
<tr>
<td>Body</td>
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</tr>
<tr>
<td></td>
<td>o=user 14341433 14341433 IP4 10.43.122.3</td>
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</tr>
<tr>
<td></td>
<td>a=rtpmap:0 PCMU/8000</td>
</tr>
</tbody>
</table>
SIP/2.0 404 Not Found
Via: SIP/2.0/UDP 10.43.122.3; branch=1
From: sip:alice@a.com;tag=4ad340f
To: sip:barbara@b.com;tag=4435211
Call-ID: 1874630@10.43.122.3
Cseq: 12442 INVITE
<table>
<thead>
<tr>
<th>Response line</th>
<th>SIP/2.0 404 Not Found</th>
</tr>
</thead>
<tbody>
<tr>
<td>Headers</td>
<td>Via: SIP/2.0/UDP 10.43.122.3; branch=1</td>
</tr>
<tr>
<td></td>
<td>From: sip:<a href="mailto:alice@a.com">alice@a.com</a>;tag=4ad340f</td>
</tr>
<tr>
<td></td>
<td>To: sip:<a href="mailto:barbara@b.com">barbara@b.com</a>;tag=4435211</td>
</tr>
<tr>
<td></td>
<td>Call-ID: 1874630@10.43.122.3</td>
</tr>
<tr>
<td></td>
<td>Cseq: 12442 INVITE</td>
</tr>
<tr>
<td>Empty line</td>
<td></td>
</tr>
</tbody>
</table>
1. Call Thomas - INVITE
2. Query “Where is Thomas@u-strasbg.fr?” (non-SIP)
3. Response “130.79.90.55” (non-SIP)
4. ‘Proxied’ Call - INVITE
5. Response - OK
6. Response - OK
7. Multimedia Channel Established – RTP Streams
LittleGuy

Registrar

REGISTER

From: LittleGuy sip:UserB@there.com
Contact: sip:UserB@110.111.112.113

200 OK

Contact: <sip:UserB@110.111.112.113>;expires=3600
LittleGuy

REGISTRER

From: LittleGuy sip:UserB@there.com
Contact: sip:UserB@110.111.112.113

401 Unauthorized

Contact: <sip:UserB@110.111.112.113>;expires=3600
WWW-Authenticate: <Authentication Challenge>

REGISTRER

From: LittleGuy sip:UserB@there.com
Contact: sip:UserB@110.111.112.113
Authorization: <Authentication Response>

200 OK

Contact: <sip:UserB@110.111.112.113>;expires=3600
An example REGISTER request

```
REGISTER sip:b.com SIP/2.0
Via: SIP/2.0/UDP 192.168.15.2
From: sip:barbara@b.com;tag=199257
To: sip:barbara@b.com
Contact: <sip:b@192.168.15.2>
Expires: 3600
Call-ID: 950398549@192.168.15.2
CSeq: 1 REGISTER
```
### An example REGISTER request

<table>
<thead>
<tr>
<th>Request-URI registration domain</th>
<th>REGISTER sip:b.com SIP/2.0</th>
</tr>
</thead>
<tbody>
<tr>
<td>Via:</td>
<td>Via: SIP/2.0/UDP 192.168.15.2</td>
</tr>
<tr>
<td>Who’s registering</td>
<td>From: sip:<a href="mailto:barbara@b.com">barbara@b.com</a>;tag=199257</td>
</tr>
<tr>
<td>AOR</td>
<td>To: sip:<a href="mailto:barbara@b.com">barbara@b.com</a></td>
</tr>
<tr>
<td>Contact</td>
<td>Contact: <a href="">sip:barbara@192.168.15.2</a></td>
</tr>
<tr>
<td>Duration in seconds</td>
<td>Expires: 3600</td>
</tr>
<tr>
<td></td>
<td>Call-ID: 950398549@192.168.15.2</td>
</tr>
<tr>
<td></td>
<td>CSeq: 1 REGISTER</td>
</tr>
</tbody>
</table>

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Emil Ivov, SIP Communicator
SIP/2.0 200 Ok
Via: SIP/2.0/UDP 192.168.15.2
From: sip:barbara@b.com;tag=199257
To: sip:barbara@b.com;tag=jjf223
Contact:<sip:barbara@192.168.15.2>;expires=3600
Contact:<sip:10.0.0.1>;expires=345
Contact:<sip:10.0.0.2>;expires=1000
Call-ID: 950398549@192.168.15.2
CSeq: 345435 REGISTER
### An example REGISTER response

<table>
<thead>
<tr>
<th>Who's registering</th>
<th>From: sip:<a href="mailto:barbara@b.com">barbara@b.com</a>;tag=199257</th>
</tr>
</thead>
<tbody>
<tr>
<td>AOR</td>
<td>To: sip:<a href="mailto:barbara@b.com">barbara@b.com</a>;tag=jjf223</td>
</tr>
<tr>
<td>List of all Contact headers for known AORs</td>
<td>Contact:<a href="">sip:barbara@192.168.15.2</a>;expires=3600</td>
</tr>
<tr>
<td></td>
<td>Contact:<a href="">sip:10.0.0.1</a>;expires=345</td>
</tr>
<tr>
<td></td>
<td>Contact:<a href="">sip:10.0.0.2</a>;expires=1000</td>
</tr>
<tr>
<td>Empty line</td>
<td>Call-ID: 950398549@192.168.15.2</td>
</tr>
<tr>
<td></td>
<td>CSeq: 345435 REGISTER</td>
</tr>
</tbody>
</table>

SIP/2.0 200 Ok

Via: SIP/2.0/UDP 192.168.15.2
- It is up to the user agent to refresh registrations of Contact addresses. In order to do so, a UA has to resend its initial REGISTER request.

- In order to cancel a Contact registration, a user agent has to set its “Expires” time to zero

  To: sip:barbara@b.com  
  Contact: <sip:barbara@192.168.15.2>  
  Expires: 0

- In order to cancel all contact address of records, a UA could use an asterisk (*)

  To: sip:barbara@b.com  
  Contact: *  
  Expires: 0

- Omitting the Contact header would not modify any AOR and the corresponding response would contain all existin AORs.
Inter Domain SIP Signaling

 INVITE 1
 TRYING 3
 RINGING 8
 OK 11

 INVITE 2
 TRYING 5
 RINGING 7
 OK 10

 ACK

 INVITE 4
 RINGING 6
 OK 9

 ... MEDIA ...

 BYE
 OK

CoIP with SIP. Rennes, November 2007
Emil Ivov, SIP Communicator
Hellooo ... did u forget about NATs?
Less standard NAT usage: End – to – end services
The basics of IP telephony.

VoIP Basics, X/Stra, Oct 2, 2008
emil.ivov@sip-communicator.org
The basics of IP telephony.

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The basics of IP telephony.

network core
(registrars, proxies, ...)

Alice
Address: A
Port: Pa

Bob
Address: B
Port: Pb

... MEDIA ...

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emil.ivov@sip-communicator.org
And then NATs were born …

Call: To: B
Media: Ap

NAT/Firewall
Address: F

Call: To: B
Media: Ap

ERROR

Alice
Private Address: Ap

Bob
Address: B
How do NATs work …

<table>
<thead>
<tr>
<th>Internal host:port</th>
<th>NAT port</th>
</tr>
</thead>
<tbody>
<tr>
<td>192.168.0.12 : 2368</td>
<td>8632</td>
</tr>
</tbody>
</table>

**Alice**

192.168.0.12

**MSG:**

Dst: 130.79.200.22 : 80
Src: 192.168.0.12 : 2368

**NAT**

Internal Address: 192.168.0.254

**MSG:**

Dst: 130.79.200.22 : 80
Src: 212.50.2.18 : 8632

**Server**

Address: 130.79.200.22

**NAT**

Public Address: 212.50.4.18
How do NATs work ...

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Alice

**MSG:**
- **Dst:** 192.168.0.12 : 2368
- **Src:** 130.79.200.22 : 80

NAT

- Internal Address: 192.168.0.254
- Public Address: 212.50.4.18

Server

- Address: 130.79.200.22
How do NATs work ... 

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Endpoint-Independent Mapping
Endpoint-Independent Filtering

Alice
192.168.0.12

Internal Address: 192.168.0.12

NAT
Internal Address: 192.168.0.254

NAT
Public Address: 212.50.4.18

Server
Address: 130.79.200.22

MSG: 
Dst: 192.168.0.12 : 2368
Src: 60.55.68.53 : 9595

Bob
Address: 60.55.68.53

MSG: 
Dst: 212.50.4.18 : 8632
Src: 60.55.68.53 : 9595
Basic Firewall and NAT Traversal

**STUN**

---

**What are my address and port?**

**Address:** $F$

**Port:** $P_f$

---

**Call:**

**To:** $B$

**Media:** $F:P_f$

---

**Answer:**

**To:** $A$

**Media:** $B$

---

**Alice**

Address: $A_p$

Port: $P_a$

---

**Bob**

Address: $B$

**STUN Server**

---

**STUN Server**

---

**VoIP Basics, X/Stra, Oct 2, 2008**

emil.ivov@sip-communicator.org
How do NATs work …
Address (and port) dependent filtering

<table>
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<tr>
<th>Internal host:port</th>
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<td>130.79.200.22 (: 80)</td>
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Alice
192.168.0.12

Internal Address: 192.168.0.254

192.168.0.12 : 2368

STUN Server
Address: 130.79.200.22

Internal host:port
NAT port
Active connections host:port

VoIP Basics, X/Stra, Oct 2, 2008
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**Alice**

192.168.0.12

**NAT**

Internal Address: 192.168.0.254

**STUN Server**

Address: 130.79.200.22

VoIP Basics, X/Stra, Oct 2, 2008
emil.ivov@sip-communicator.org
How do NATs work …

Address (and port) dependent filtering

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Endpoint-Independent Mapping

Endpoint-Dependent Filtering

Alice
- Internal Address: 192.168.0.12

NAT
- Internal Address: 192.168.0.254

Bob
- Address: 60.55.68.53

NAT
- Public Address: 212.50.4.18

STUN Server
- Address: 130.79.200.22

VoIP Basics, X/Stra, Oct 2, 2008
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How do NATs work ...
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NAT
Internal Address: 192.168.0.254
Public Address: 212.50.4.18

STUN Server
Address: 130.79.200.22

MSG:
Dst: 60.55.68.53 : 80
Src: 192.168.0.12 : 2368

Bob
Address: 60.55.68.53

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Address (and port) dependent filtering

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Endpoint-Independent Mapping
Endpoint-Dependent Filtering

**Alice**
- Internal Address: 192.168.0.12

**NAT**
- Internal Address: 192.168.0.254
- Public Address: 212.50.4.18

**Bob**
- Address: 60.55.68.53

**STUN Server**
- Address: 130.79.200.22

**MSG:**
- Dst: 192.168.0.12 : 2368
- Src: 60.55.68.53 : 80

VoIP Basics, X/stra, Oct 2, 2008
emil.ivov@sip-communicator.org
How do NATs work …

Endpoint dependent mapping

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Internal Address: 192.168.0.12

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**NAT**

Internal Address: 192.168.0.254

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<tr>
<th>STUN Server Address: 130.79.200.22</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>MSG:</strong></td>
</tr>
<tr>
<td>Dst: 130.79.200.22 : 80</td>
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**NAT**

Public Address: 212.50.4.18

VoIP Basics, X/stra, Oct 2, 2008
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How do NATs work … Endpoint dependent mapping

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Alice
192.168.0.12

MSG: Dst: 192.168.0.12:2368
Src: 130.79.200.22:80

NAT
Internal Address: 192.168.0.254

STUN Server
Address: 130.79.200.22

MSG: Dst: 212.50.4.18:8632
Src: 130.79.200.22:80

NAT
Public Address: 212.50.4.18
How do NATs work ... Endpoint dependent mapping

<table>
<thead>
<tr>
<th>Internal host:port</th>
<th>NAT port</th>
<th>Active connections host:port</th>
</tr>
</thead>
<tbody>
<tr>
<td>192.168.0.12 : 2368</td>
<td>8632</td>
<td>130.79.200.22 (: 80)</td>
</tr>
<tr>
<td>192.168.0.12 : 2368</td>
<td>9391</td>
<td>60.55.68.53 (: 80)</td>
</tr>
</tbody>
</table>

**Alice**
- Internal Address: 192.168.0.12
- STUN Server Address: 130.79.200.22
- MSG: Dst: 60.55.68.53 : 80, Src: 192.168.0.12 : 2368

**Bob**
- Internal Address: 192.168.0.254
- Public Address: 212.50.4.18

**MSG:**
- Dst: 60.55.68.53 : 80
- Src: 192.168.0.12 : 2368

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emil.ivov@sip-communicator.org
How do NATs work ...

Endpoint dependent mapping

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</tr>
</tbody>
</table>

Endpoint-Dependent Mapping
Endpoint-Dependent Filtering

MSG:
Dst: 192.168.0.12 : 2368
Src: 60.55.68.53 : 80

Alice
192.168.0.12

NAT
Internal Address: 192.168.0.254

NAT
Public Address: 212.50.4.18

Bob
Address: 60.55.68.53

STUN Server
Address: 130.79.200.22
Relaying Media

STUN Relay Server
Address: T
Port: Pt

Symmetric NAT/Firewall F1:P1

Alice
Address: Ap
Port: Pa

Reserve port

Symmetric NAT/Firewall F1:p2

STUN Relay Server
Address: T
Port: Pt

Call:
To: B
Media: T:Pt

Bob
Address: B

Call:
To: B
Media: T:Pt

Call:
To: B
Media: T:Pt

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Relaying Media

Symmetric NAT/Firewall $F1:P1$

STUN Relay Server
Address: $T$
Port: $Pt$

Alice
Address: $Ap$
Port: $Pa$

Bob
Address: $B$

Symmetric NAT/Firewall $F1:p2$
Relaying Media (The SIP Way)

Alice
Address: Ap
Port: Pa

Bob
Address: B

NAT/Firewall

SIP Server
Address: T
Port: Pt

Call:
To: B
Media: A:Pa

Call:
To: B
Media: T:Pt
Relaying Media (The SIP Way)

Alice
Address: Ap
Port: Pa

NAT/Firewall

SIP Server
Address: T
Port: Pt

 MEDIA

Bob
Address: B

 MEDIA

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Relaying Media

SIP clients behind a symmetric NAT/firewall

non-scalable expensive complex

symmetric firewall

Relay Server

SIP clients behind a symmetric NAT/firewall

symmetric NAT/firewall

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Could we please have IPv6 now?

... ok, it’s probably high time we moved to IPv6 ...
Could we please have IPv6 now?

... this should simplify VoIP

... shouldn’t it?
VoIP and IPv6 – demo version

Alice
2001:660::2

Bob
2001:660::1

Call:
To: 2001:660::1
Media: 2001:660::1

network core
(registrars, proxies, ...)

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VoIP and IPv6 – demo version

Bob
2001:660::1

Alice
2001:660::2

To: 2001:660::1
Media: 2001:660::2

network core
(registrars, proxies, ...)

Answer:

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VoIP and IPv6 – demo version

network core
(registrars, proxies, ...)

Bob
2001:660::1

Alice
2001:660::2

... MEDIA ...

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Reality check!
Alice: 2001:660::2, 192.168.0.6
Bob: 216.109.112.135

Alice’s list of addresses:
- 2001:660::2
- 192.168.0.6
- 172.16.0.9
- 130.79.12.64
- 64.233.187.99
- 212.50.4.12

VPN: Priv: 172.16.0.0, Pub: 64.233.187.99
NAT: 130.79.12.64
Stun Relay Server: 212.50.4.12
SIP network
How to avoid relaying?

Interactive Connectivity Establishment (ICE)

An IETF draft brought to you by Cisco’s Jonathan Rosenberg
Address management … … with ICE

Alice
192.168.0.6

Bob
216.109.112.135

VPN
Priv: 172.16.0.0
Pub: 64.233.187.99

Stun Relay Server
212.50.4.12

NAT
130.79.12.64

SIP network

Please try me on any of the following:
2001:660::2
192.168.0.6
172.16.0.9
130.79.12.64
64.233.187.99
212.50.4.12

Alice’s list of addresses:
2001:660::2
192.168.0.6
172.16.0.9
130.79.12.64
64.233.187.99
212.50.4.12
Address management ... 
... with ICE

Alice's list of addresses:
2001:660::2
192.168.0.6
172.16.0.9
130.79.12.64
64.233.187.99
212.50.4.12
Deployment
- Reliability – support for failover
- A lot of available resources
- Scalability
  - Simple Configurations
    ~ 30 calls Soekris board, no transcoding
    ~ 60 calls with transcode Pentium4
    2.4 = 80% busy
  - Clustering
  - Load sharing Combining with SER
- Codecs:
  - ADPCM, G.711 (A-Law & µ-Law), G.722, G.723.1 (pass through), G.726, G.729, GSM, iLBC, Linear, LPC-10, Speex

- Other features:
  - conference bridging
  - voice mail (including email delivery)
  - echo and MP3 plugins
  - call parking, queueing, recording, retrieving, snooping
  - caller id
  - call blocking
  - ENUM
  - fax transmit and receive
  - music on hold, and transfer ............
SIP Express Router (SER)

- Light weight
- Very fast
- Routing wise, more sophisticated than (*) (routing can be based on packet contents)
- Packet rewriting.
- Modules are available for: accounting, authentication, interaction with RADIUS, ENUM, SIMPLE, NAT Support, SMS gateway, web interface, stateless replies, presence agent, MySQL interaction, Jabber interaction
- Support for IPv6
SIP Express Media Server (SEMS)

- Especially well suited for work with SER
- Supports plugins
- Plugins shipped with SEMS:
  - Voicemail: record messages and mail them.
  - ISDN Gateway: support calls from and to the PSTN.
  - Conferencing: connect people within a conference room.
  - Announcement: plays an announcement.
  - Echo: test module echoing your voice.
- Codecs: G711.u, G711.a, GSM
Open SER

- Small footprint
- Plug&Play module interface - ability to add new extensions, without touching the core
- Support for UDP/TCP/TLS transport layers
- IPv4 and IPv6
- Flexibility of routing configuration
- Authentication, authorization and accounting (AAA)
- CPL - Call Processing Language (RFC3880)
- NAT traversal support for SIP and RTP traffic
- ENUM support

- Extension interfaces for PERL and Javaload balancing with failover, support for replication
- Interconnection with PSTN, XMPP, SMS
- Multiple database backends - MySQL, PostgreSQL, ...
- Over 70 extra modules in the OpenSER repository

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Open SER is no longer

OpenSIPS.org  Kamailio.org
Media relaying with RTP Proxy

- High performance RTP stream proxy-ing
- Works with
  - SER
  - OpenSER
  ... through their nat helper module

- Supports features such as:
  - Remote control mode
  - IPv4
  - IPv6
  - IPv4 to IPv6 relaying
the other side
SIP Communicator Overview

Audio/Video Calls with SIP

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SIP Communicator Overview
Instant Messaging

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Other?

- KPhone – [http://kphone.sf.net](http://kphone.sf.net)

- Hardware clients:
  - ...

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A Sample Deployment

IPv4 VoIP Phones

Linksys PAP

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VoIP Basics

emil.ivov@sip-communicator.org