ulp UNIVERSITÉ LOUIS PASTEUR STRASBOURG

VoIP Basics

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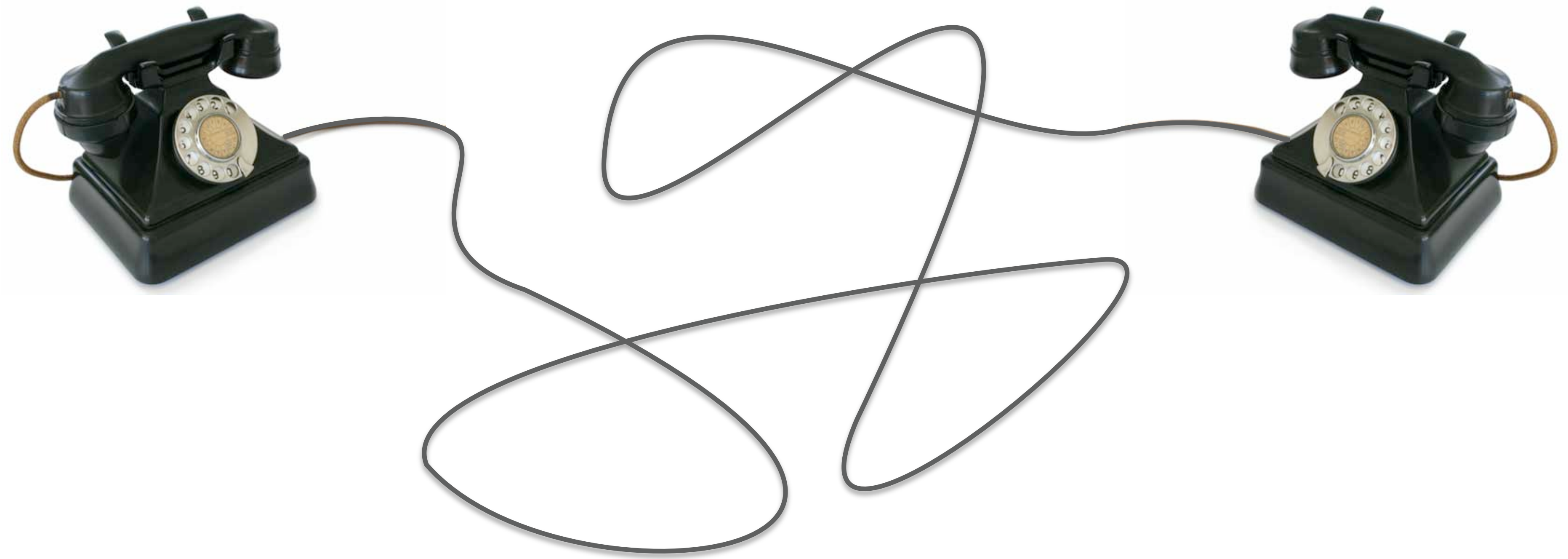
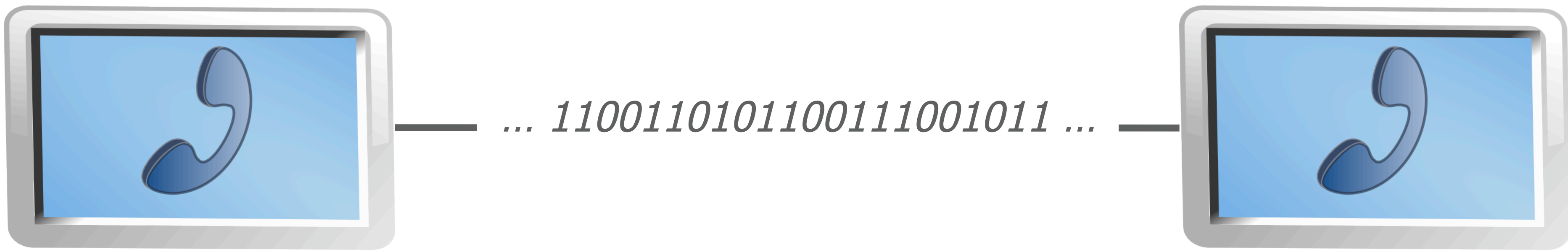
How did it all start?



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.

Voice Codecs

Codec	Bit-Rate
G711	64Kbs
G723.1	6,3Kbs
G729A	8Kbs
iLBC	15.2 Kbs
Speex	variable

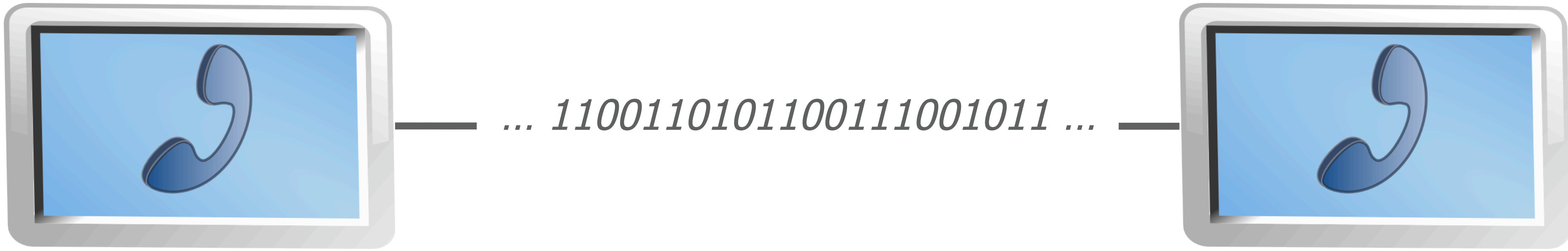
Transporting Media - TCP vs. UDP

- 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

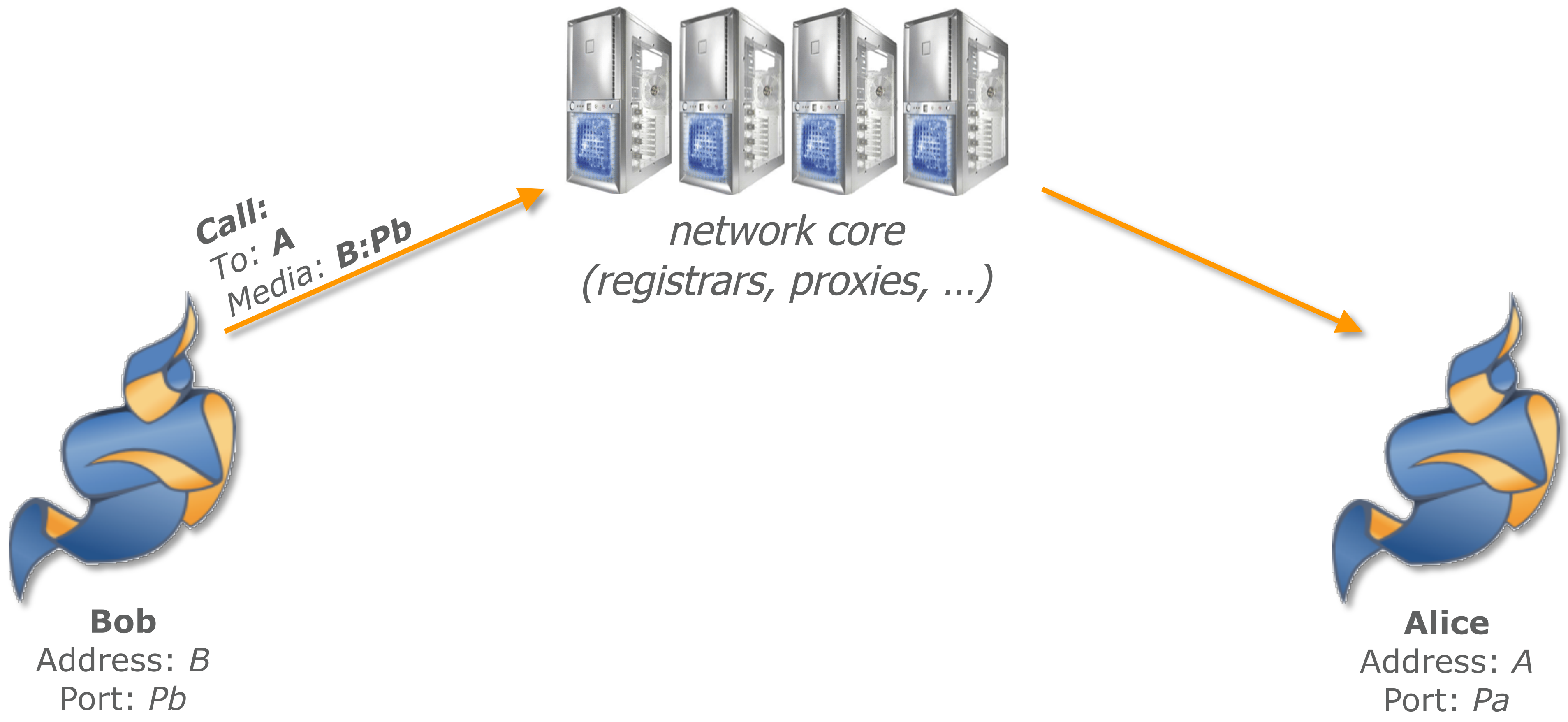
UDP + RTP

- 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?



The basics of IP telephony.



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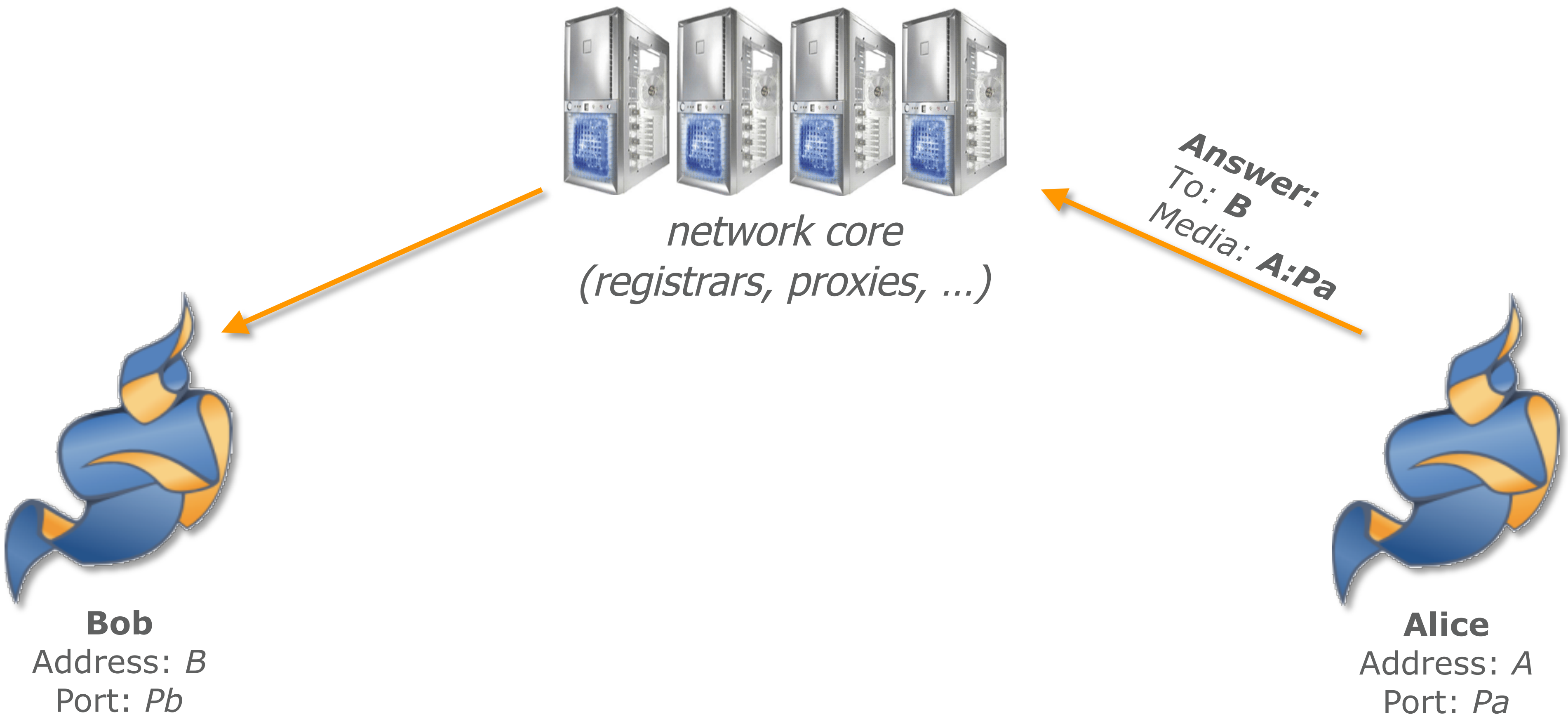


The basics of IP telephony.

Signaling

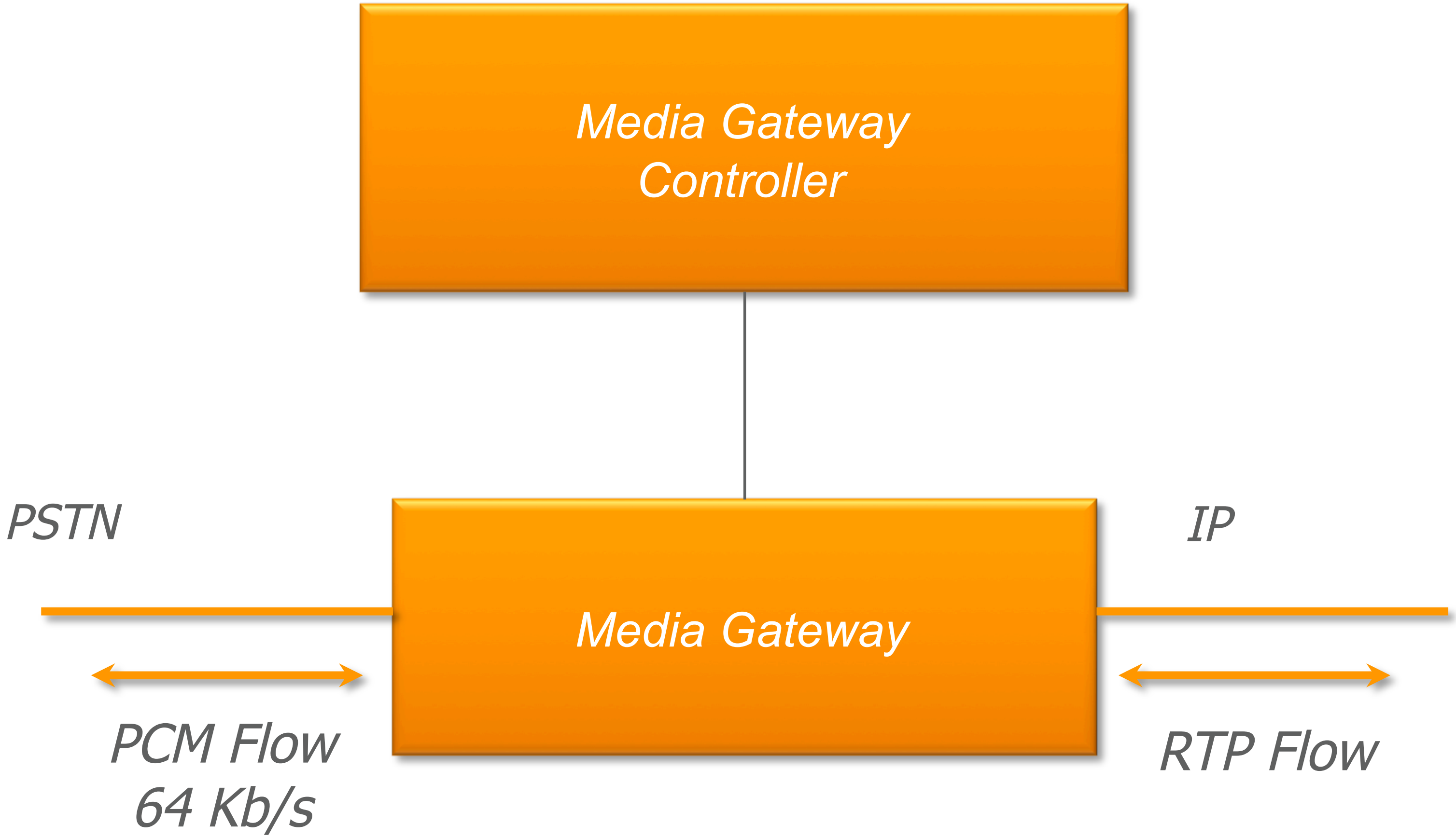
- Say why it is important to separate signalling from data.
- H.323
- MGCP
- SIP
- XMPP/Jingle

- Other signaling protocols:
  - IAX
  - Skype, ICQ, Yahoo

MEGACO- H248-MGCP

- 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.

H248-MGCP (MEGACO)



H.323

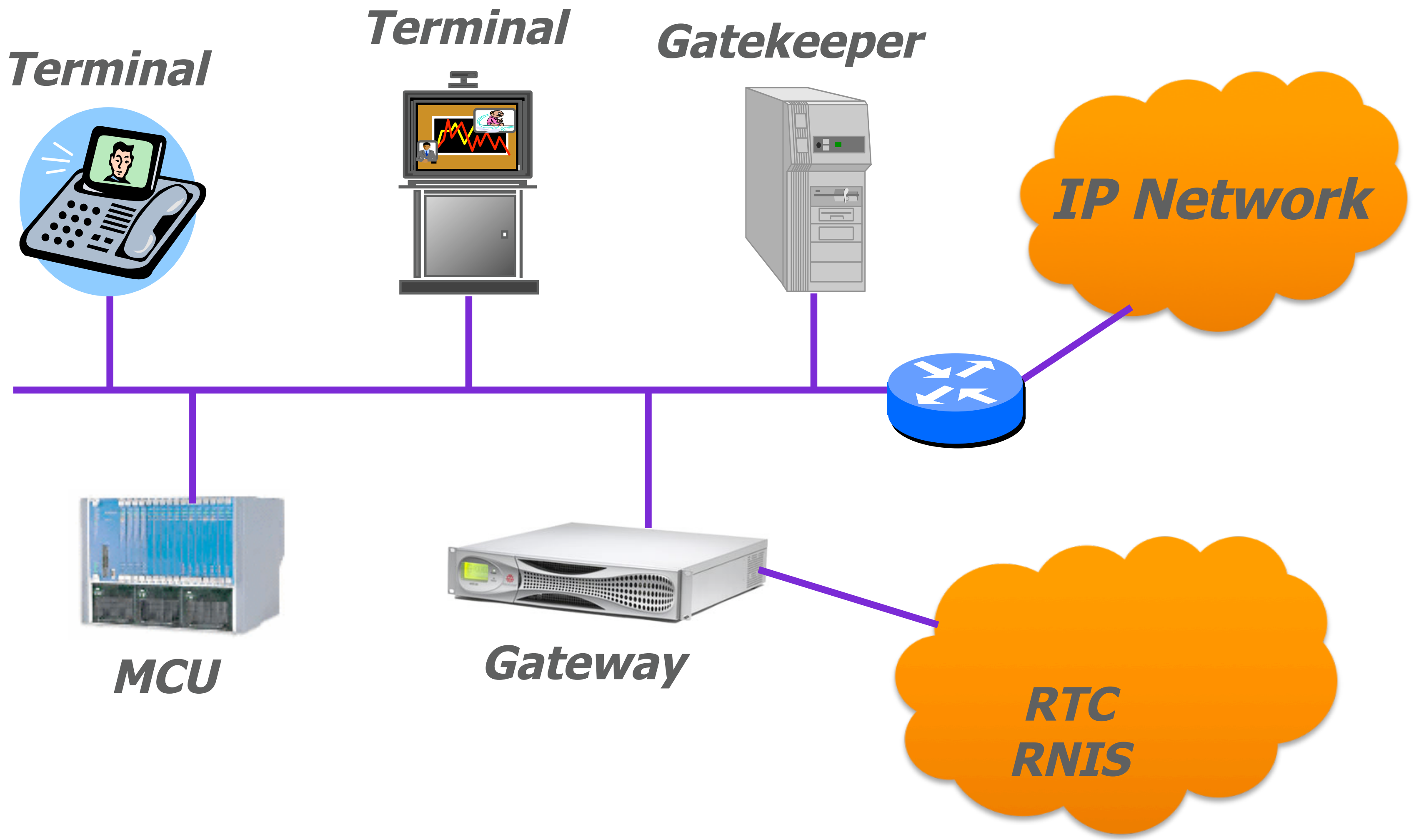
- The ITU solution for video conferencing on data networks: IP, ATM, …
  - Strongly inspired by the RNIS H320 standards for conferencing.
  - Multiple ITU PSTN low band protocols are employed by H.323
    - Q.931
    - Supplementary services coming from Q.932

H.323

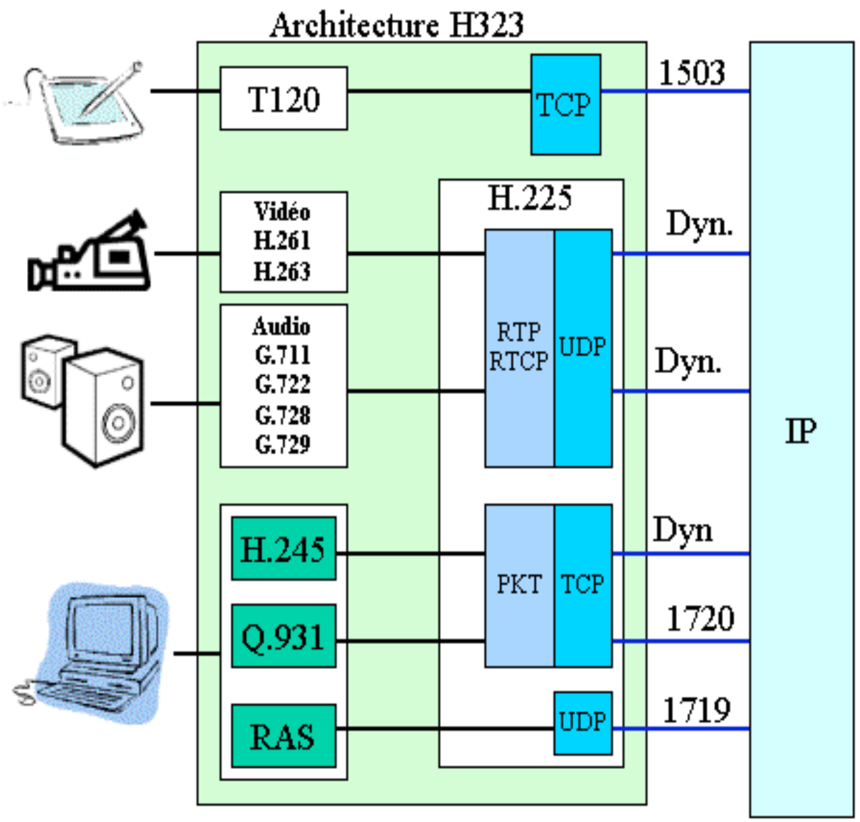
- ITU Recommendations
  - Version 1 1996 : Video telephony system for LANs with no QoS
  - Version 4 November 2000

- Pros and Cons:
  - Compatible with H320 (PSTN)
  - High complexity, difficult to adapt to the Internet (Firewall, NATs, QoS)

H.323 Entities



Architectures and protocols



RAS : Registration Admission Status

- Gatekeeper registration

Q.931 : signaling call

- Allows opening an H.245 connection

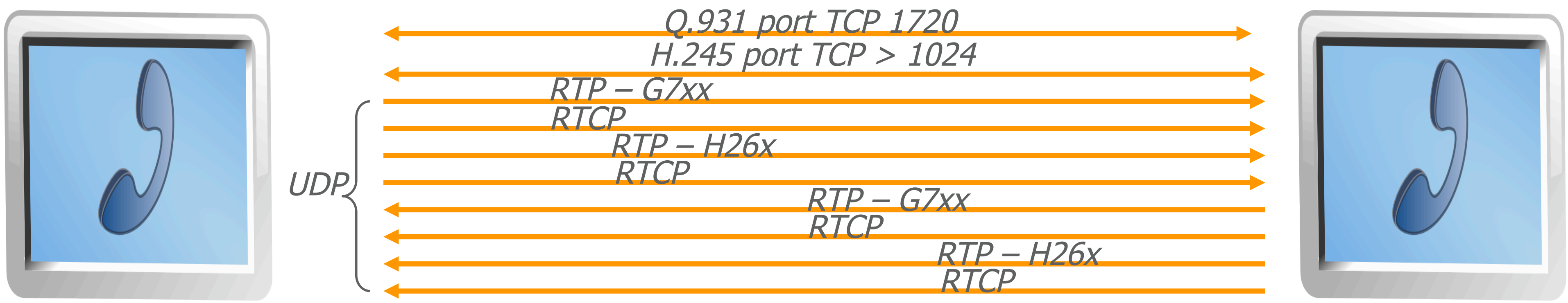
H.245 : control call

- Information exchange (codec, address, RTP and RTCP port numbers)
- Activates channels

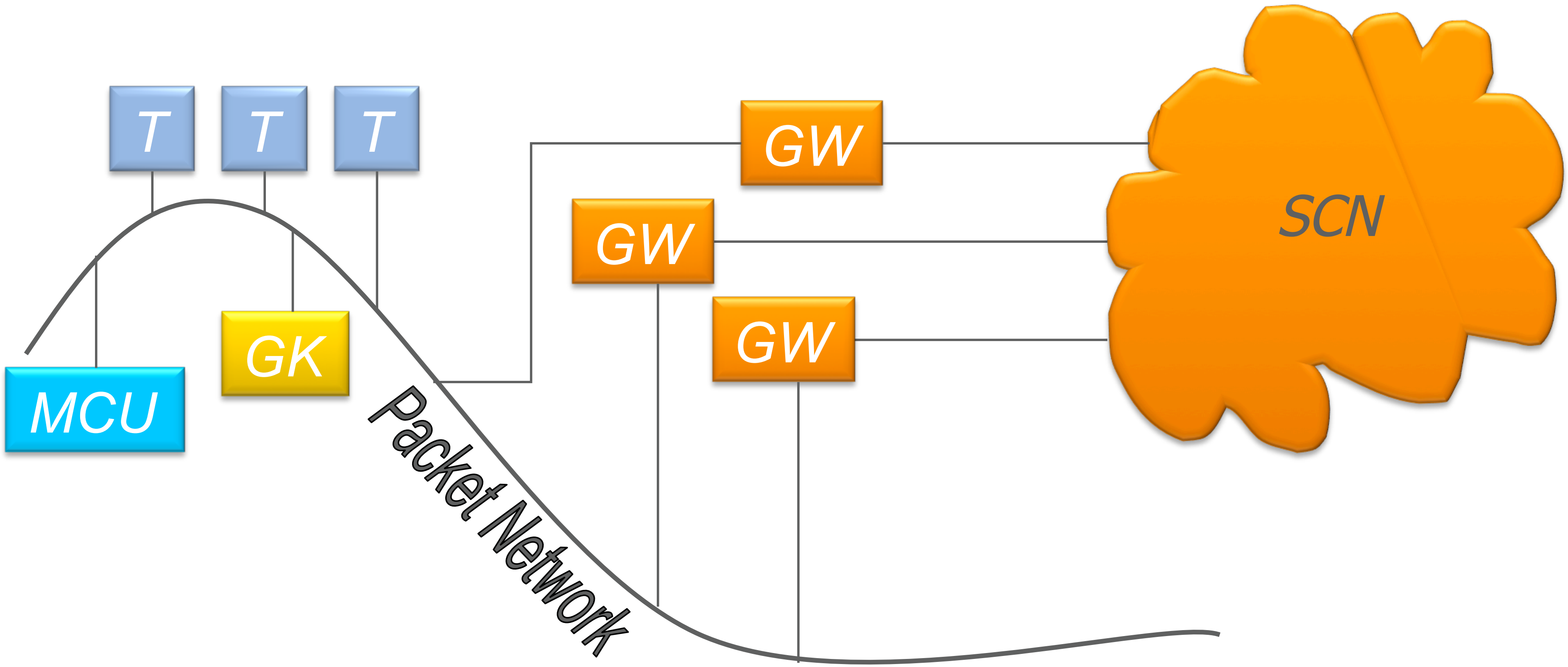
Ports

Port	Type	Used for
389	*static – TCP*	*ILS Registration (LDAP)*
1300	*static – TCP*	*H.235 Secure Signaling*
1503	*static – TCP*	*T120*
1718	*static – UDP*	*Gatekeeper Discovery*
1719	*static – UDP*	*Gatekeeper RAS*
1720	*static – TCP*	*Q.931 Call Setup*
1024-655535	*dynamic – TCP*	*H245 Control Channel*
1024-655535	*dynamic – UDP*	*RTP/RTCP – Audio/Video Streams*

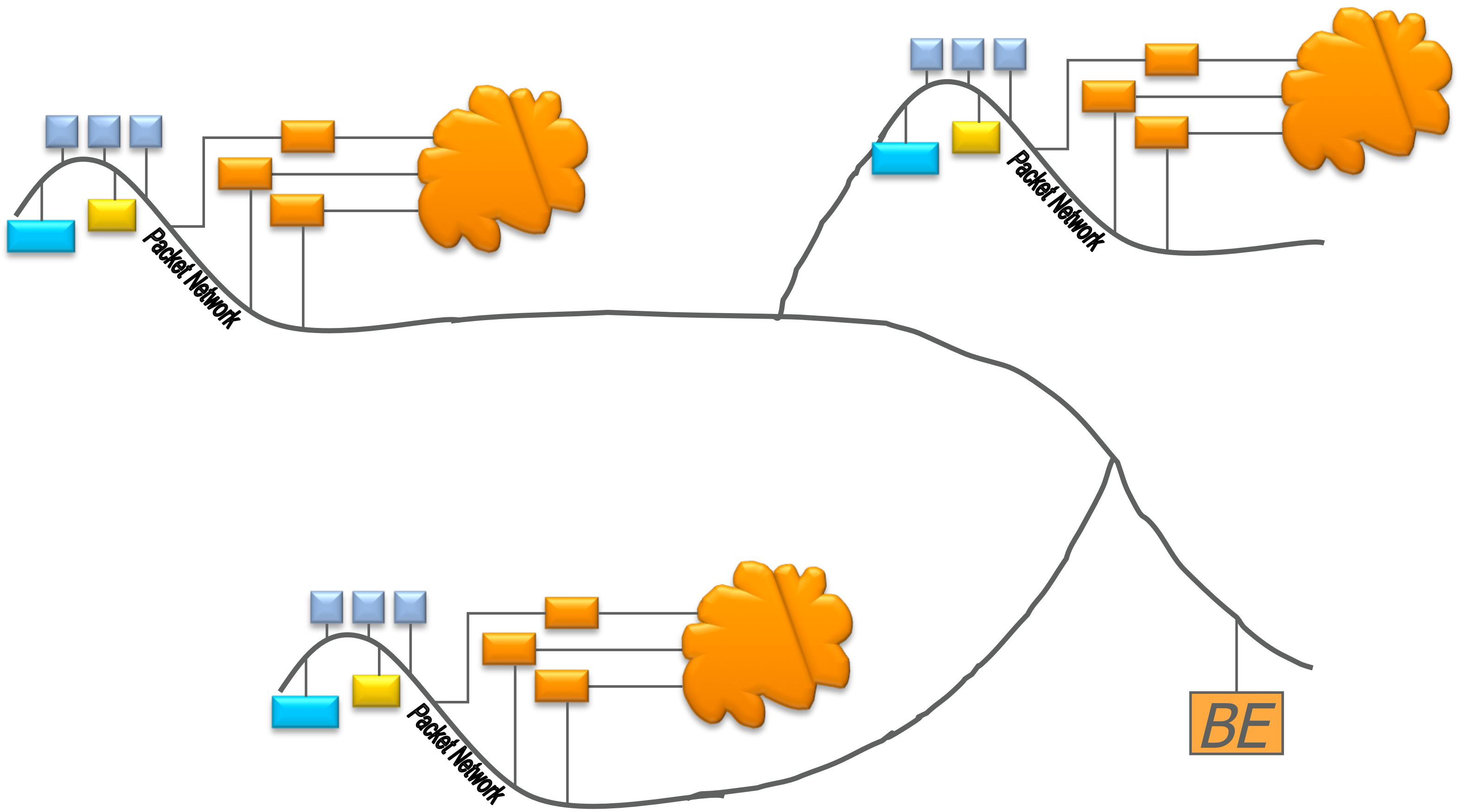
A basic 2-party call



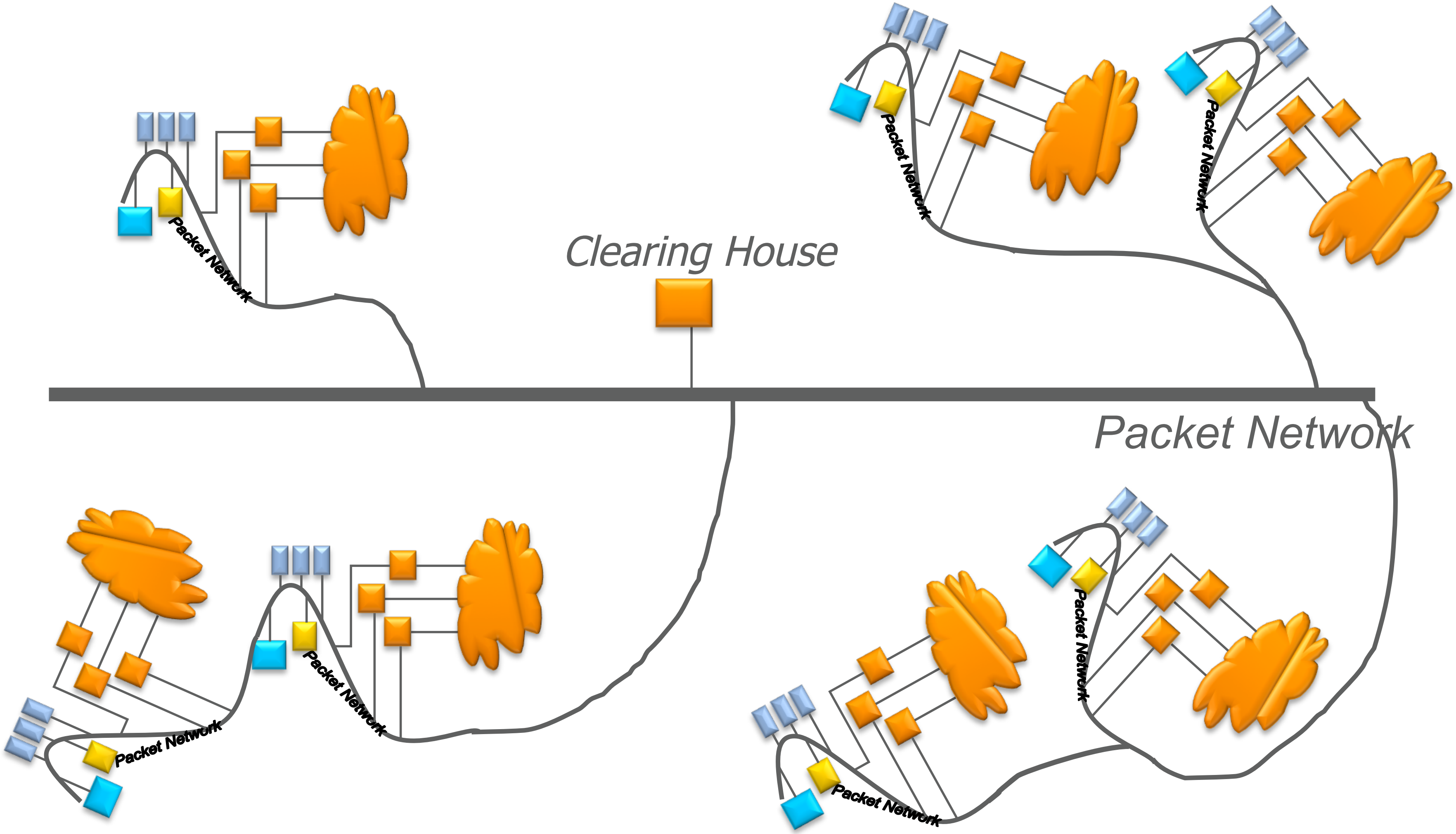
The Zone



A Single Administrative Domain



Multiple Administrative Domains



XMPP/Jingle

The Session Initiation Protocol

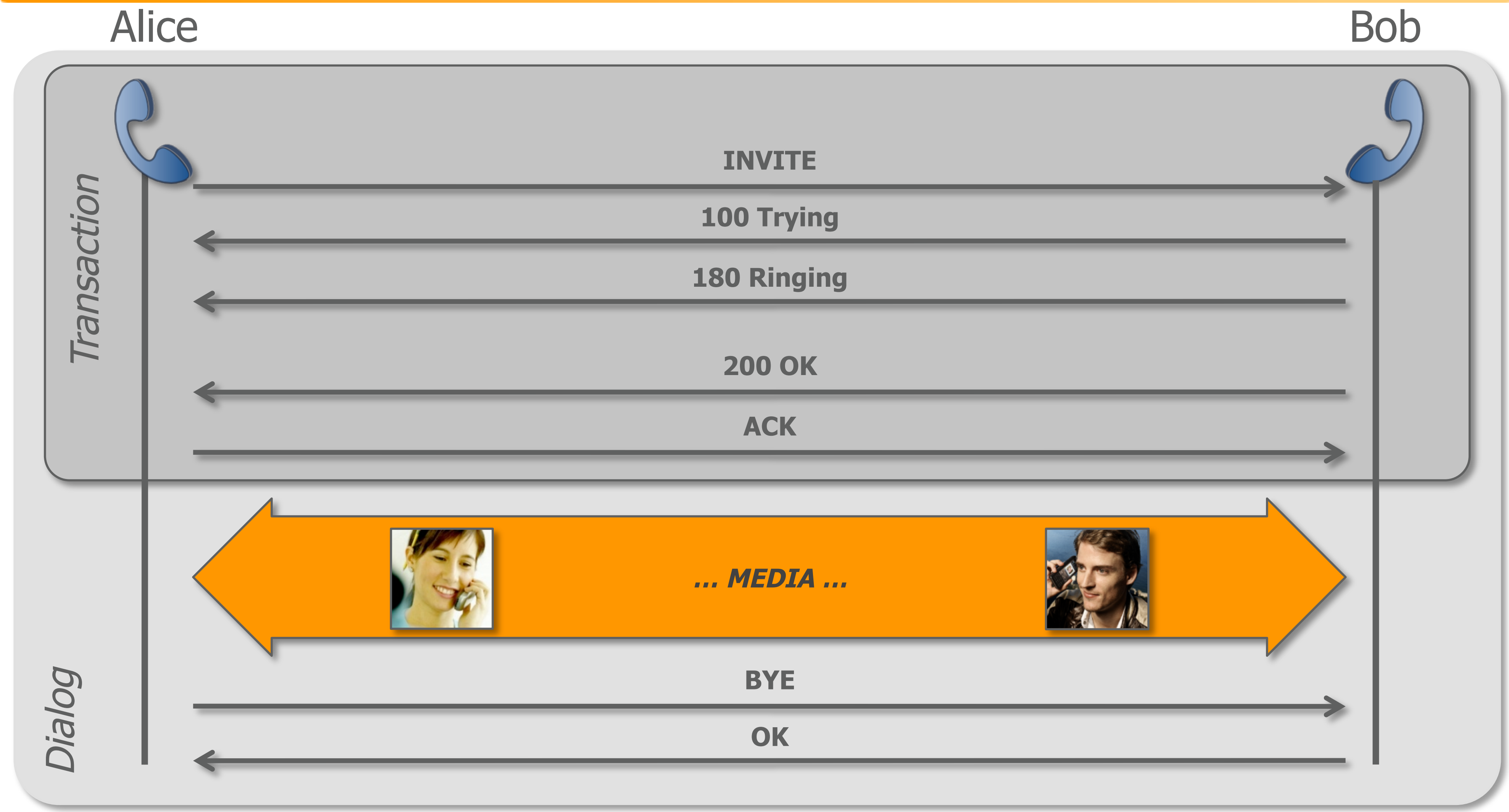
SIP

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



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

Request line	`INVITE sip:barbara@b.com SIP/2.0`

Headers	`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`

Empty line	

Body	`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 Response

```
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
```

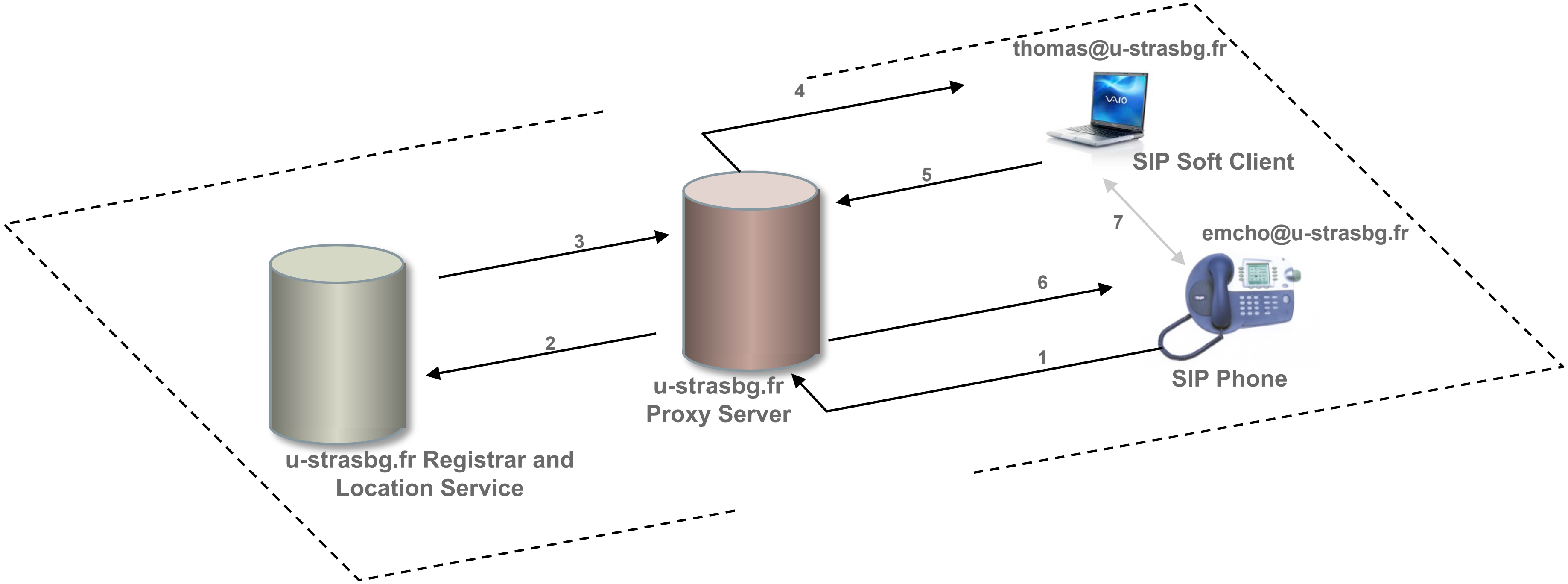
Example SIP Response

Response line	`SIP/2.0 404 Not Found`

Headers	`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`

Empty line	

Intra Domain SIP Signaling



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 Chanel Establised – RTP Streams

Registrars and Registrations

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

Registrars and Registrations

LittleGuy **Registrar**

REGISTER →

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>

REGISTER →

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

Request-URI registration domain	`REGISTER sip:b.com SIP/2.0`
	`Via: SIP/2.0/UDP 192.168.15.2`
Who's registering	`From: sip:barbara@b.com;tag=199257`
AOR	`To: sip:barbara@b.com`
Contact	`Contact: <sip:barbara@192.168.15.2>`
Duration in seconds	`Expires: 3600`
	`Call-ID: 950398549@192.168.15.2` `CSeq: 1 REGISTER`
Empty line	

An example REGISTER response

```
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>;expir
es=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

	SIP/2.0 200 Ok
	Via: SIP/2.0/UDP 192.168.15.2
Who’s regisering	From: sip:barbara@b.com;tag=199257
AOR	To: sip:barbara@b.com;tag=jjf223
List of all Contact headers for know AORs	Contact:<sip:barbara@192.168.15.2>;expir es=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
Empty line	

REGISTER: refresh, cancel, query

- It is up to the user agent to refresh registrations of Contact addresses. In order to do so, a UA has to resend its innitial REGISTER request.
- In order to cancel a Contact registration, a user agent has to set its “Expires” time to zero

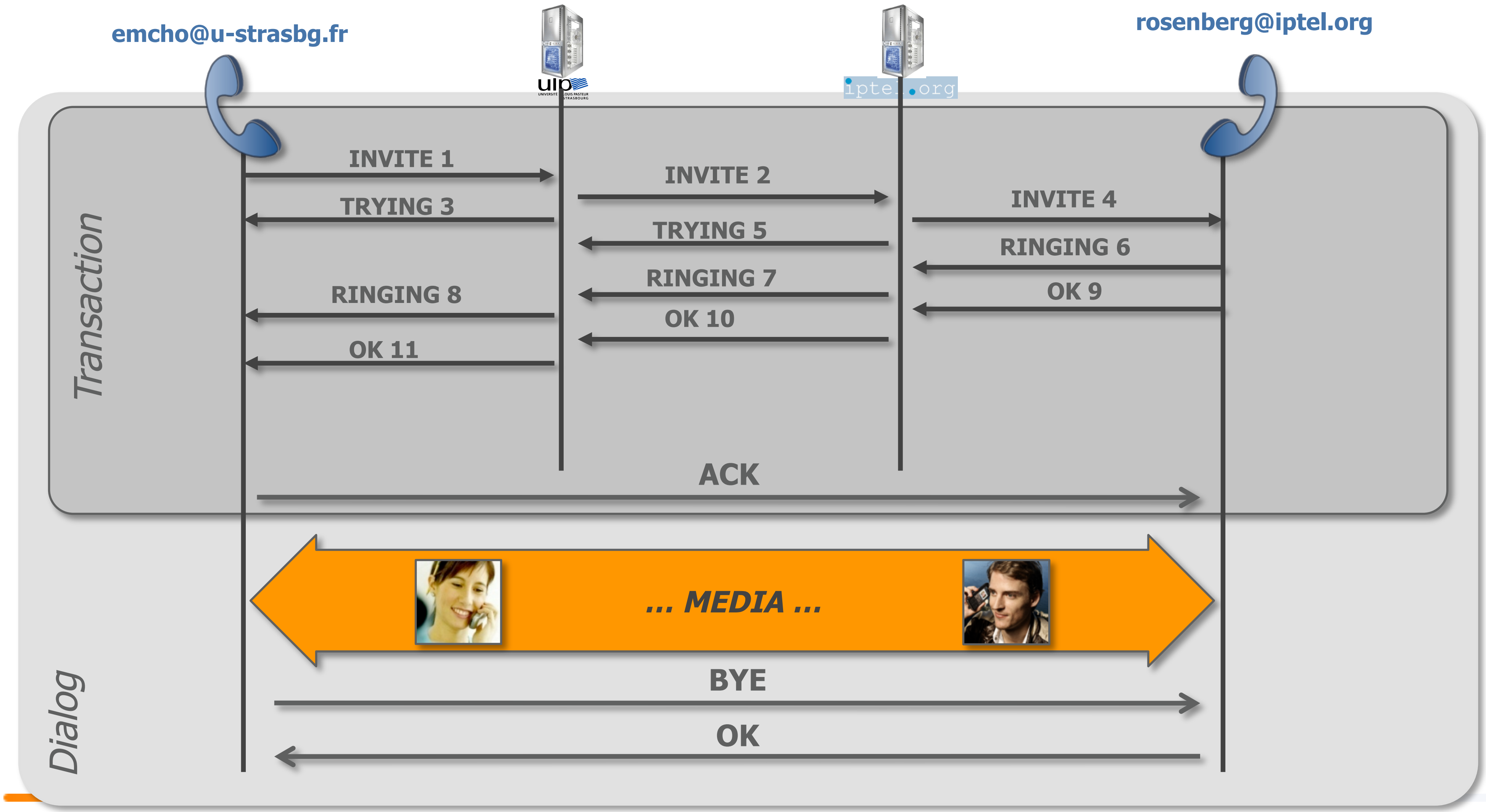
```
      To: sip:barbara@b.com
 Contact: <sip:barbara@192.168.15.2>
 Expires: 0
```

- 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.

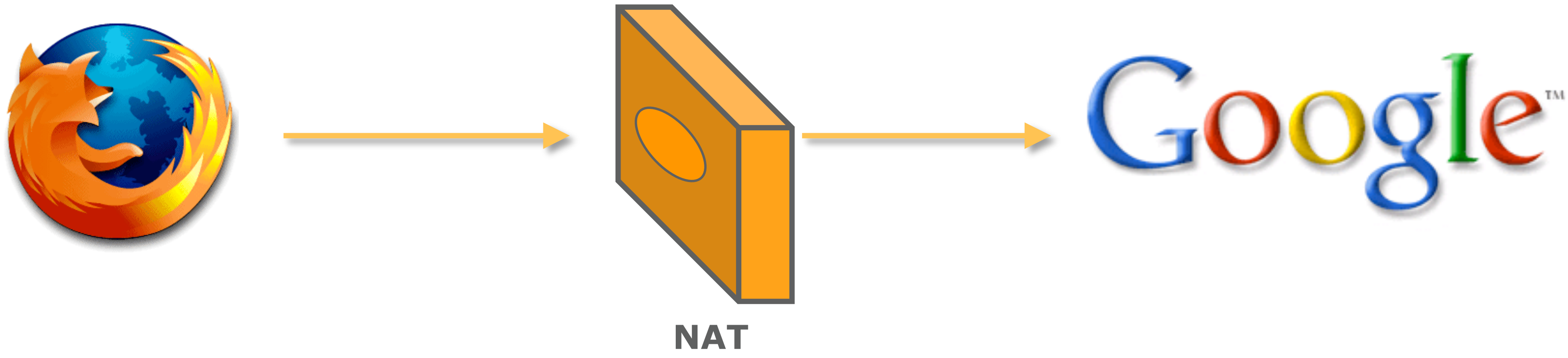
Inter Domain SIP Signaling



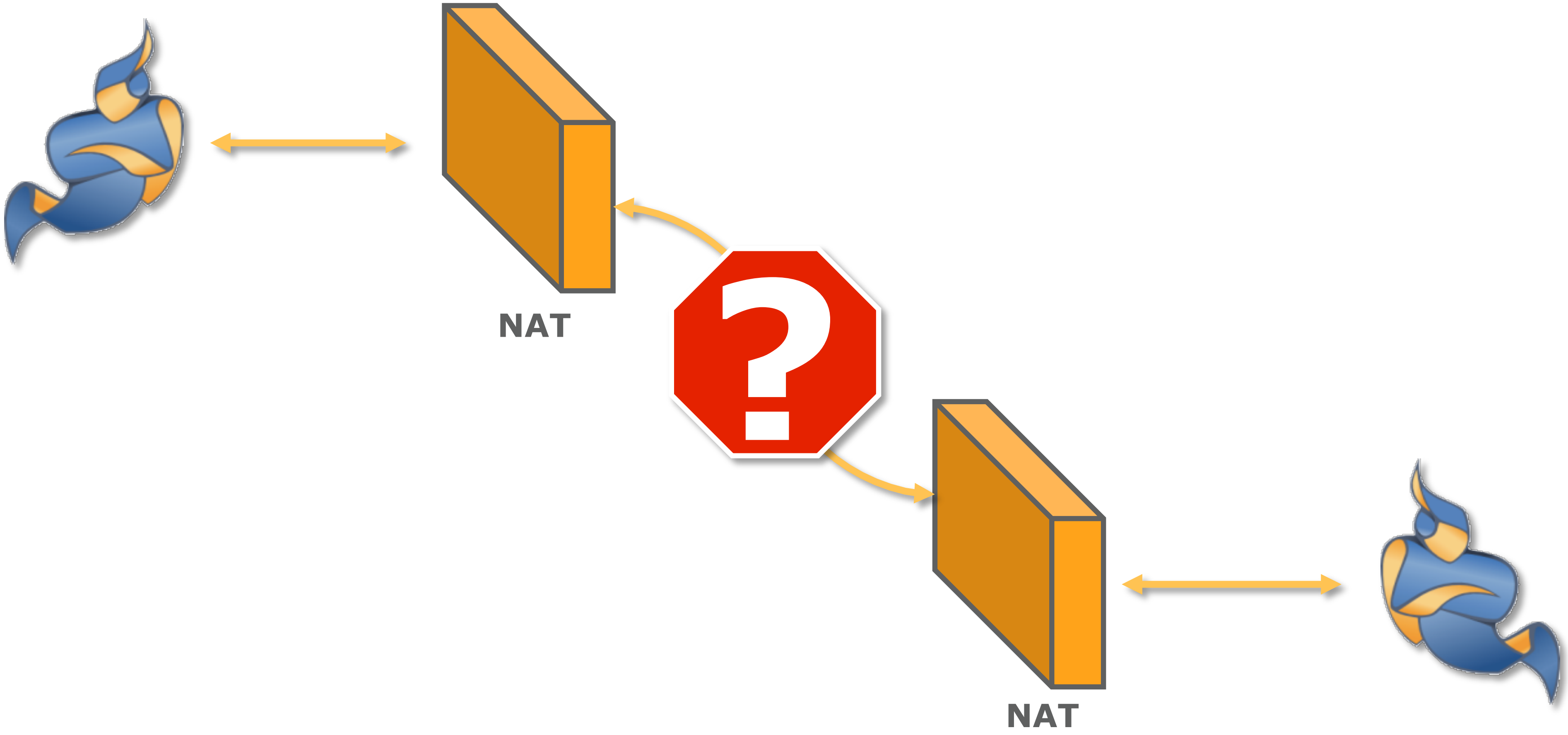
Reality Check

Hellooo … did u forget about NATs?

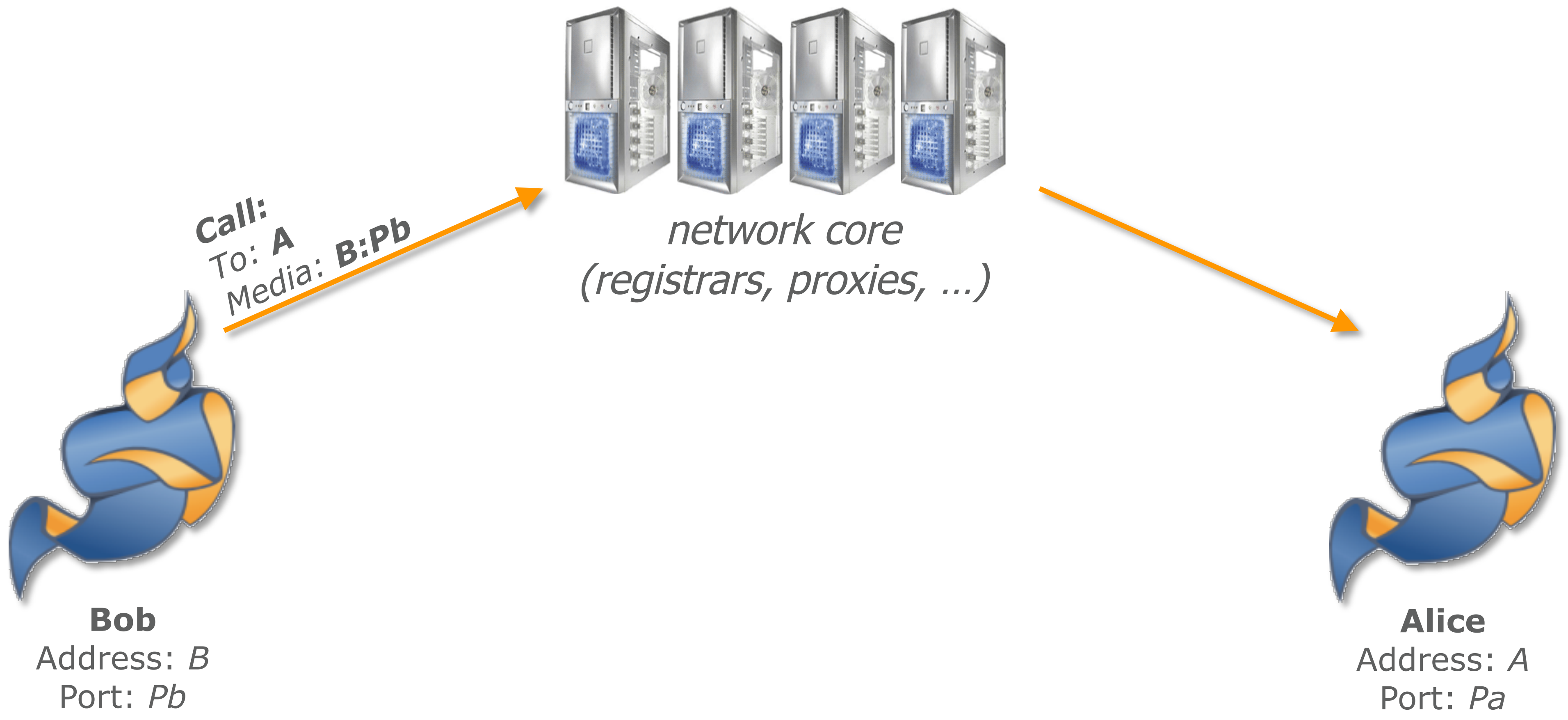
Standard NAT usage



Less standard NAT usage:
End – to –end services



The basics of IP telephony.



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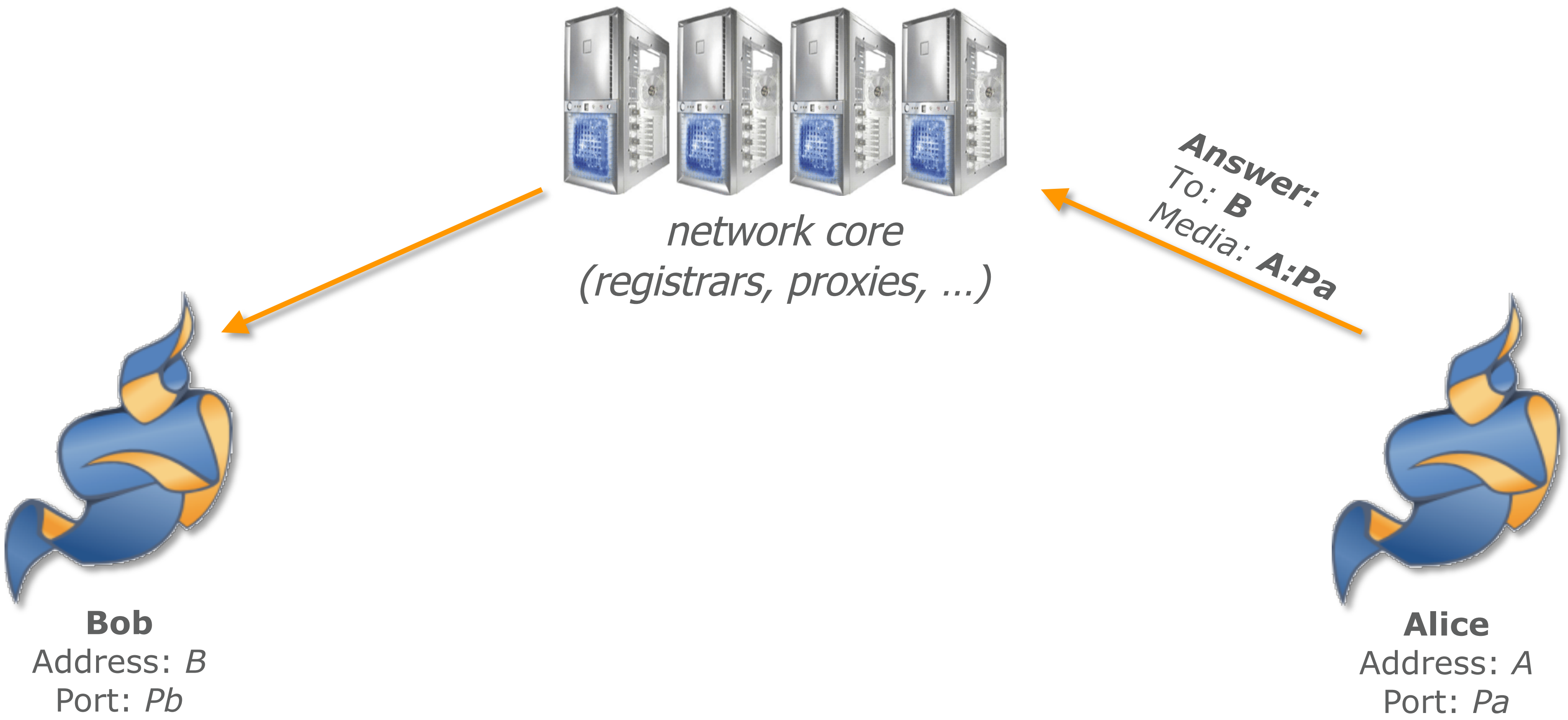
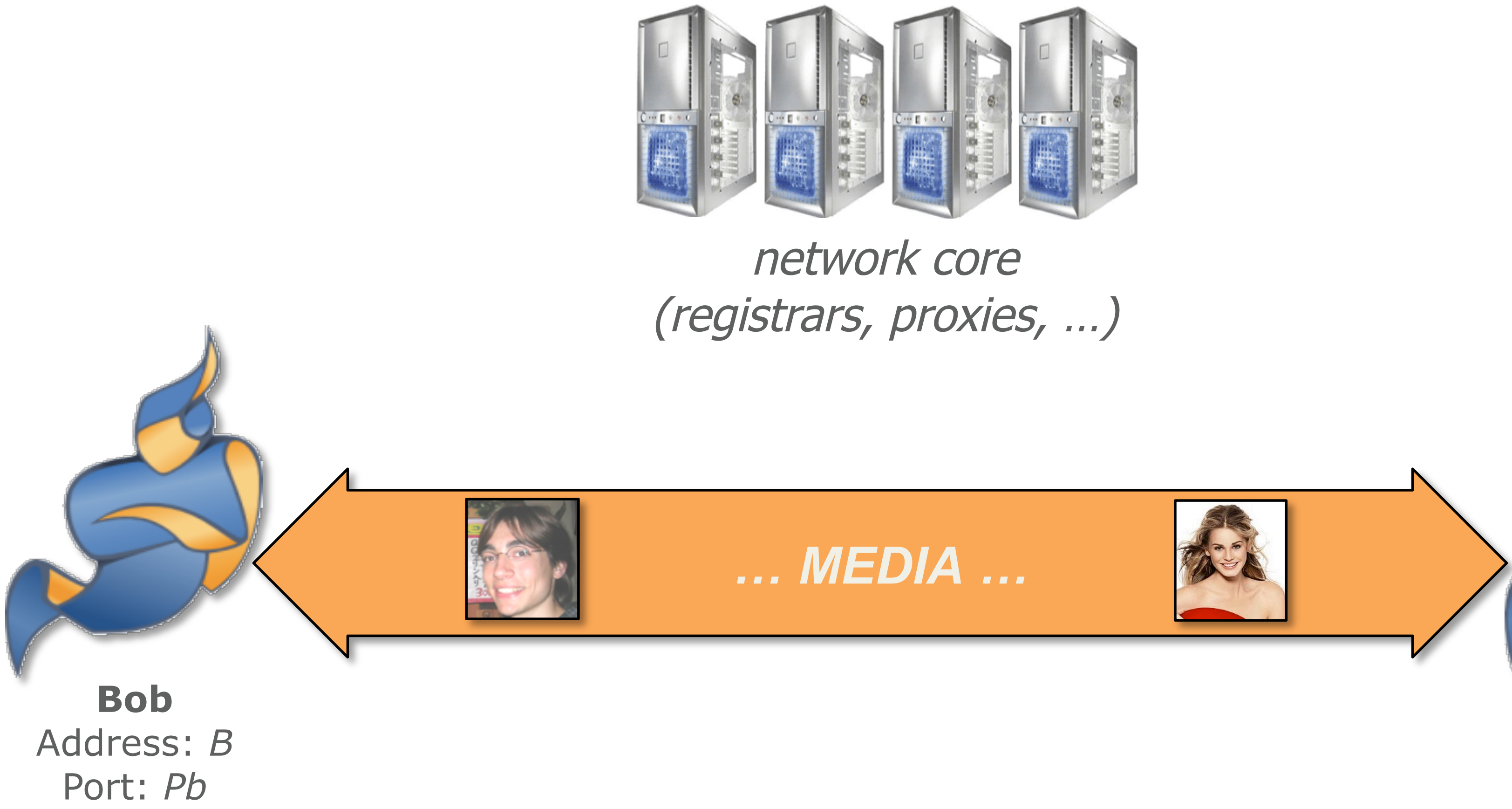
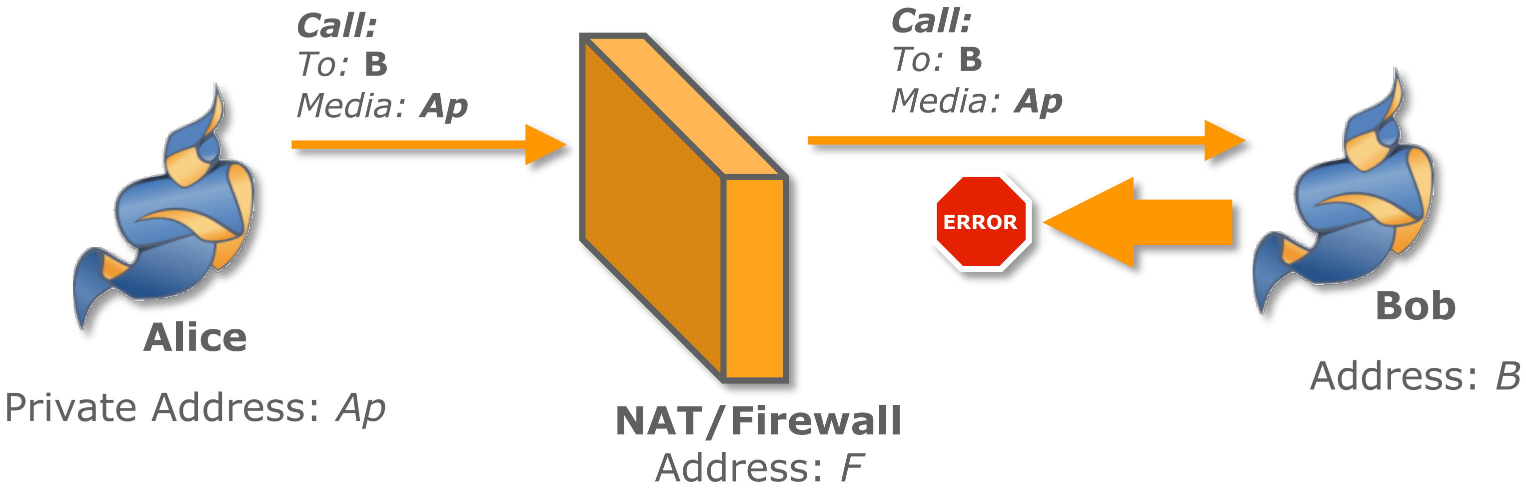


The basics of IP telephony.

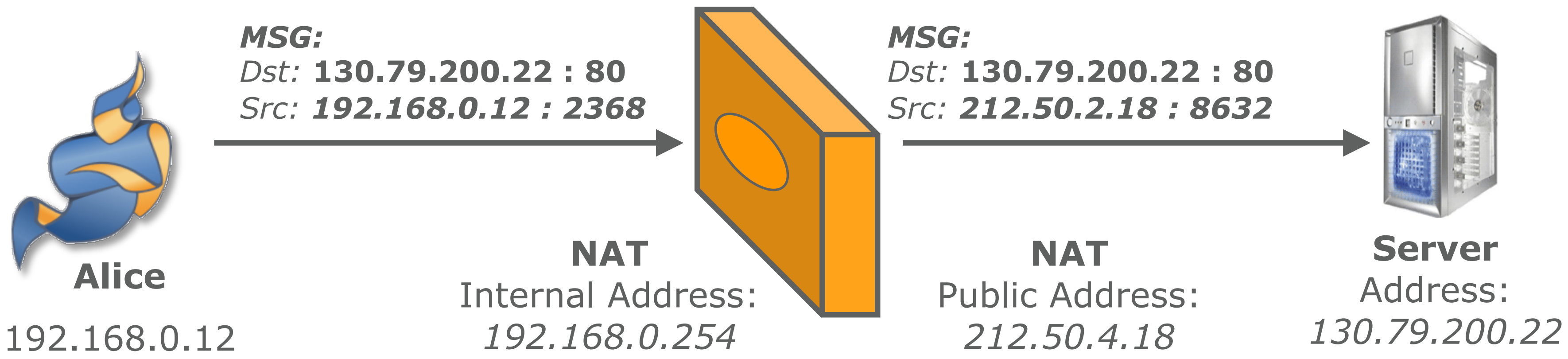


And then NATs were born …



How do NATs work …

Internal host:port	NAT port
192.168.0.12 : 2368	***8632***



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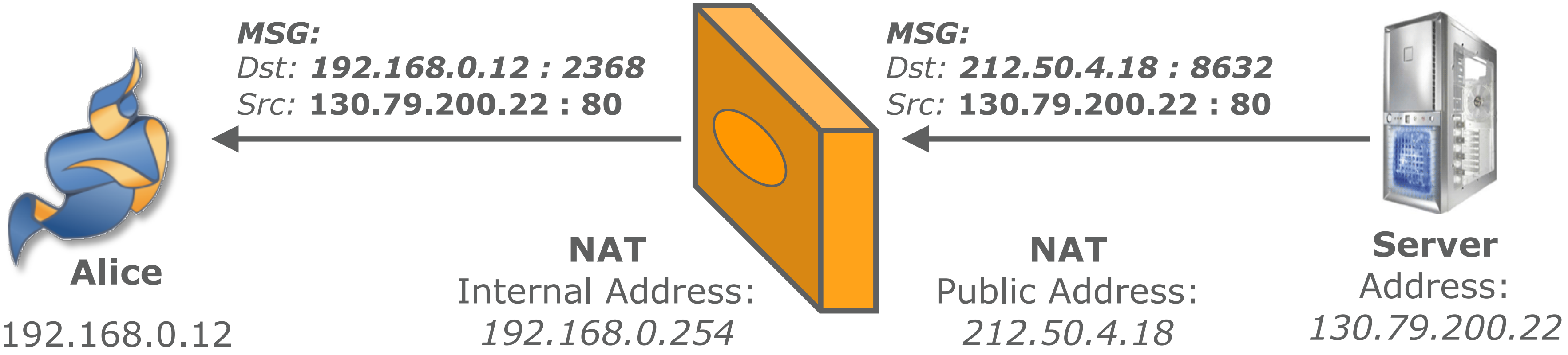
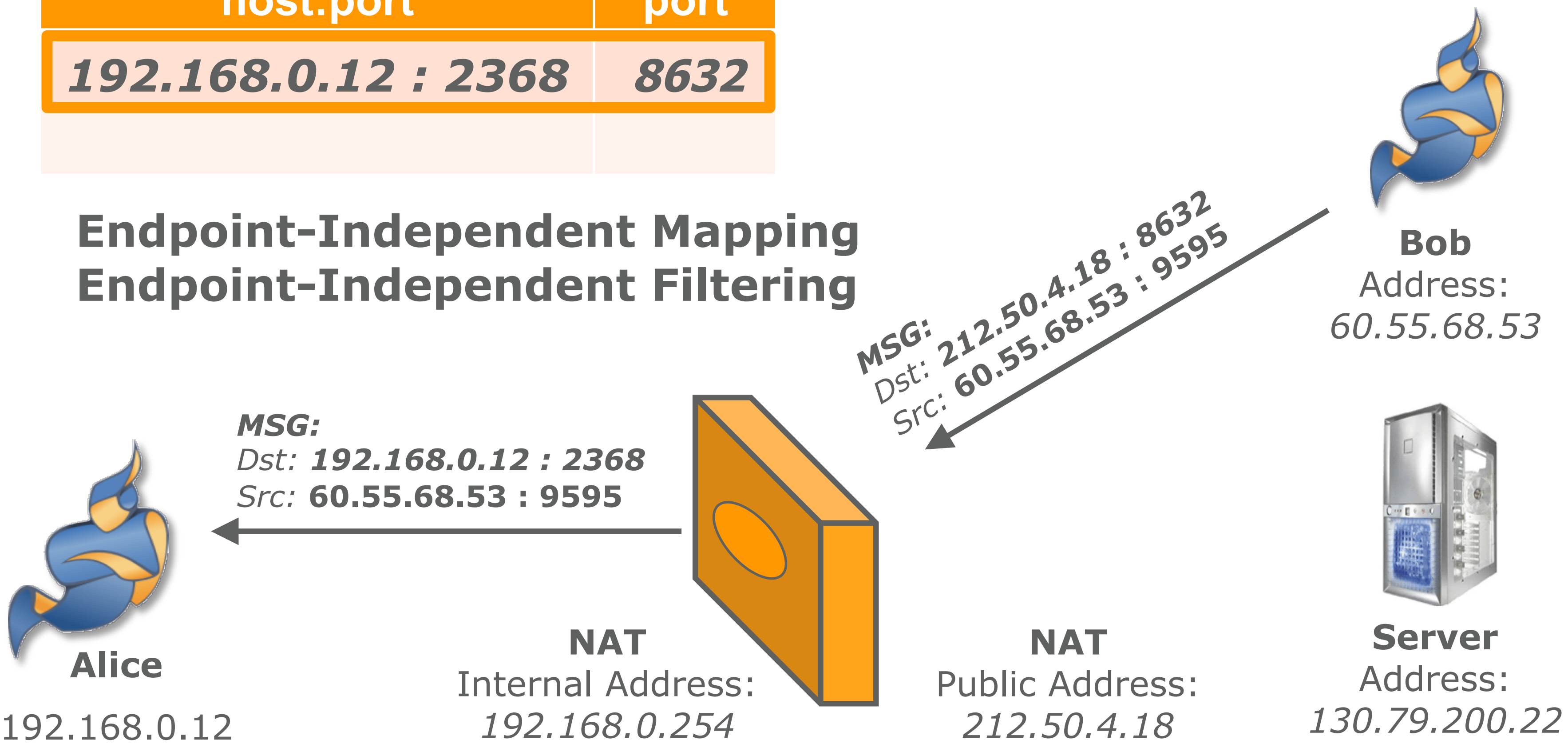


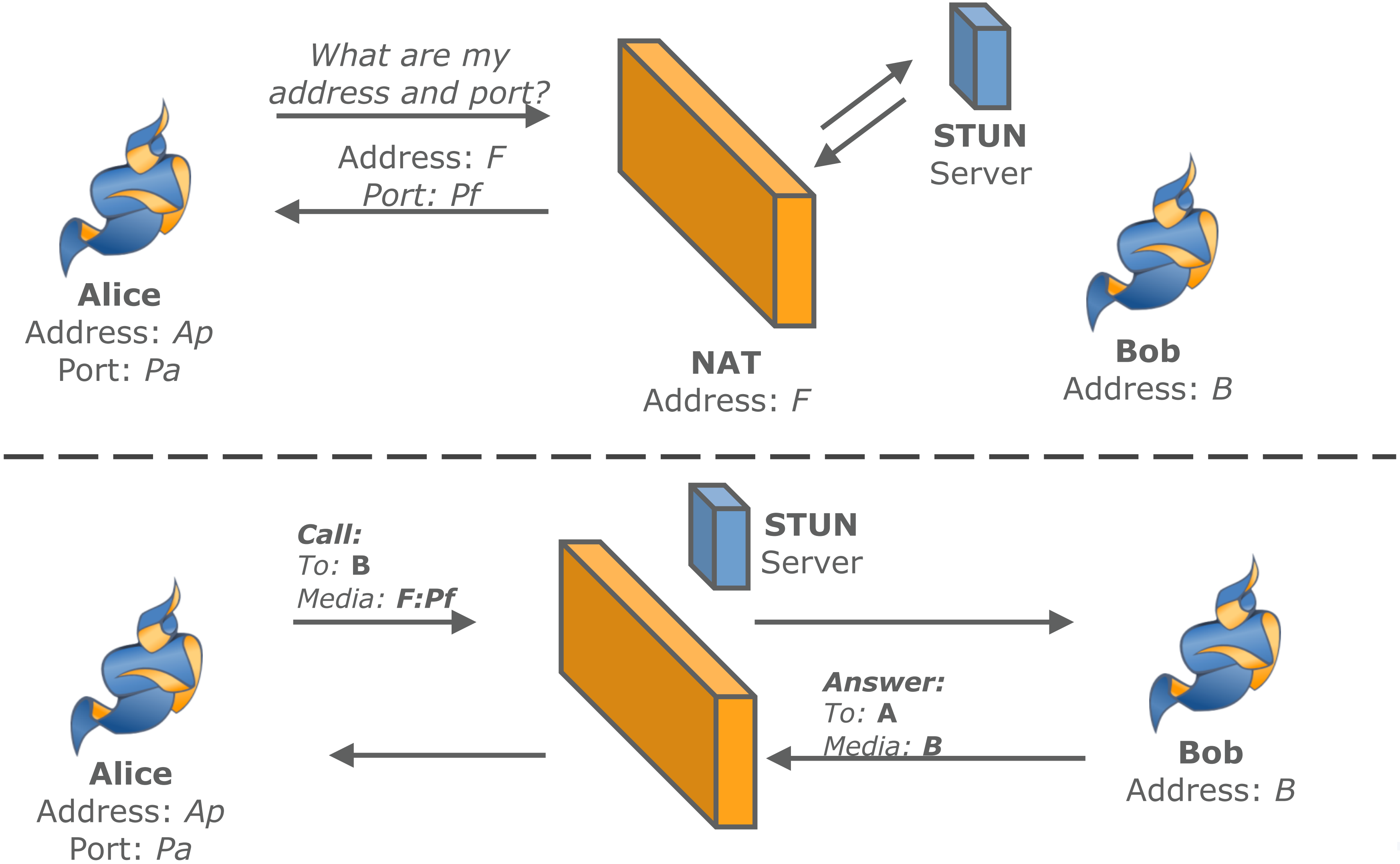
How do NATs work …

Internal **host:port**	NAT **port**
192.168.0.12 : 2368	***8632***

Endpoint-Independent Mapping
Endpoint-Independent Filtering

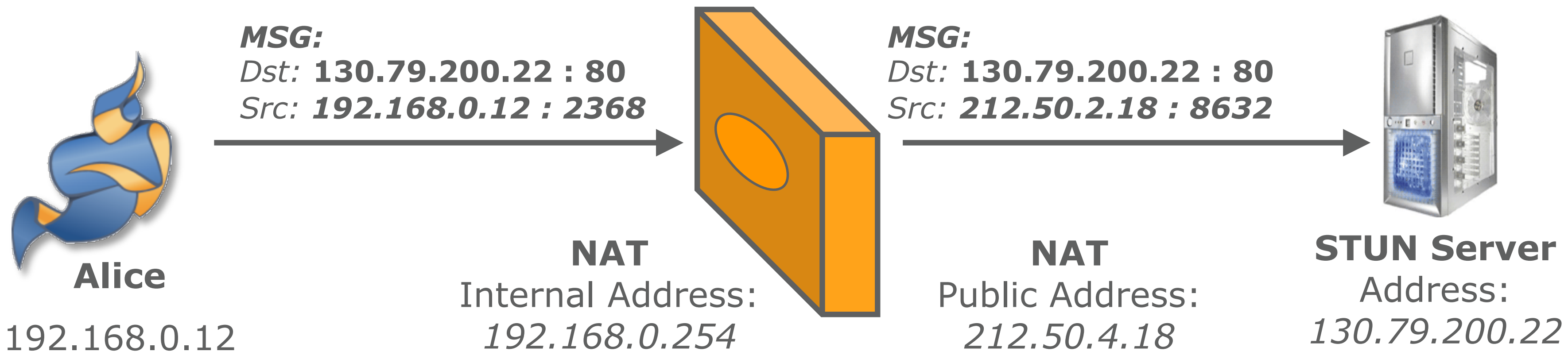


Basic Firewall and NAT Traversal
STUN



How do NATs work …
Address (and port) dependent filtering

Internal **host:port**	NAT **port**	Active connections **host:port**
192.168.0.12 : 2368	***8632***	**130.79.200.22 (: 80)**



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Address (and port) dependent filtering

Internal host:port	NAT port	Active connections host:port
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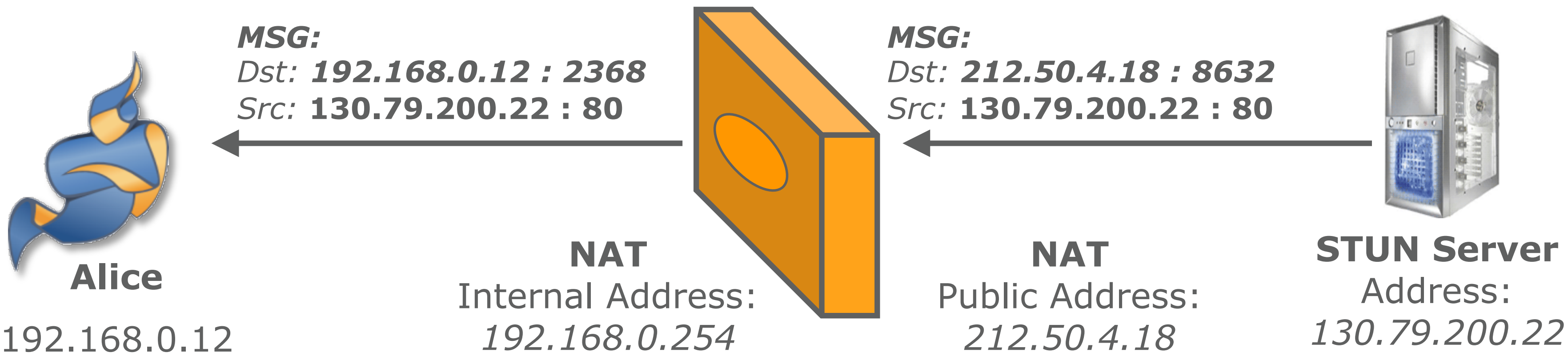
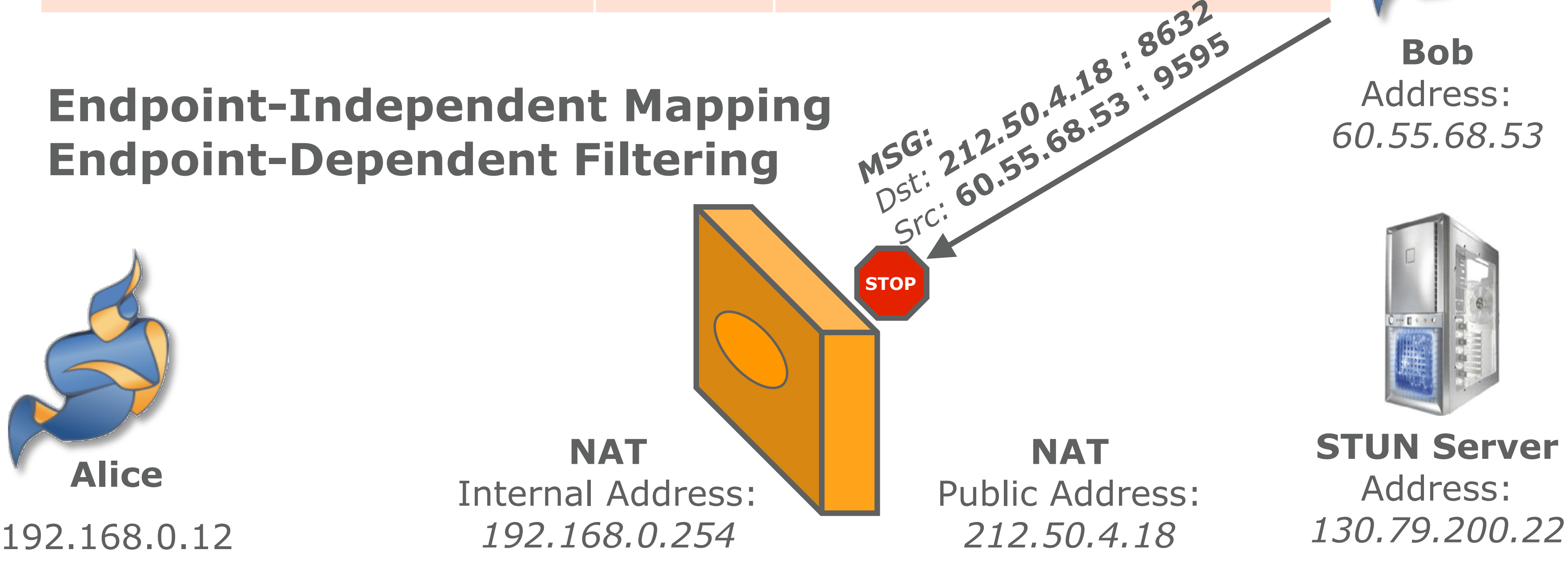


How do NATs work …
Address (and port) dependent filtering

Internal **host:port**	NAT **port**	Active connections **host:port**
192.168.0.12 : 2368	***8632***	**130.79.200.22 (: 80)**

Endpoint-Independent Mapping
Endpoint-Dependent Filtering



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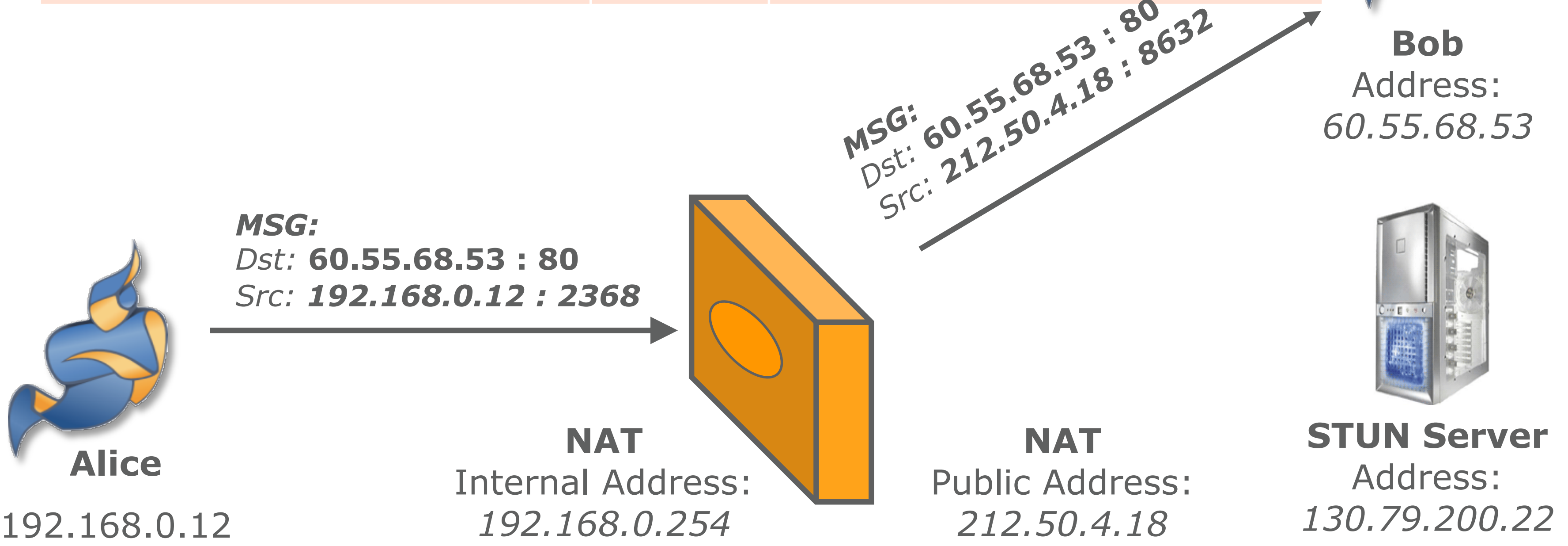
Address (and port) dependent filtering

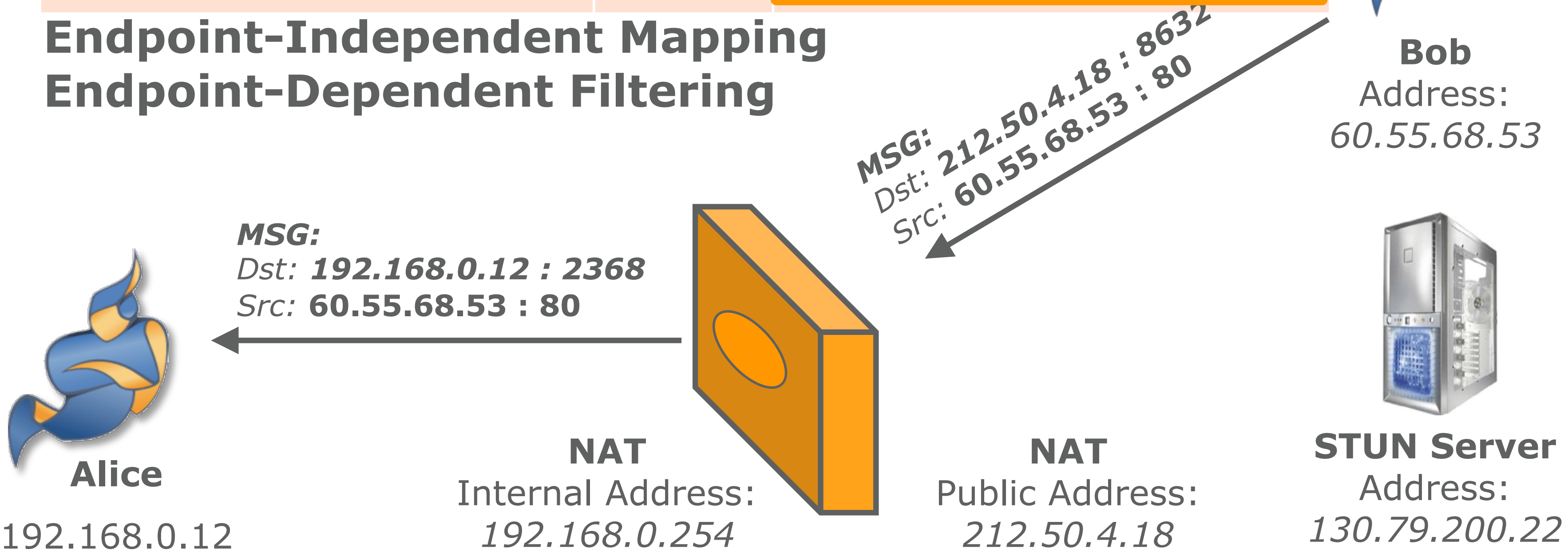
Internal **host:port**	NAT **port**	Active connections **host:port**
192.168.0.12 : 2368	***8632***	**130.79.200.22 (: 80)** **60.55.68.53 (: 80)**



How do NATs work …
Address (and port) dependent filtering

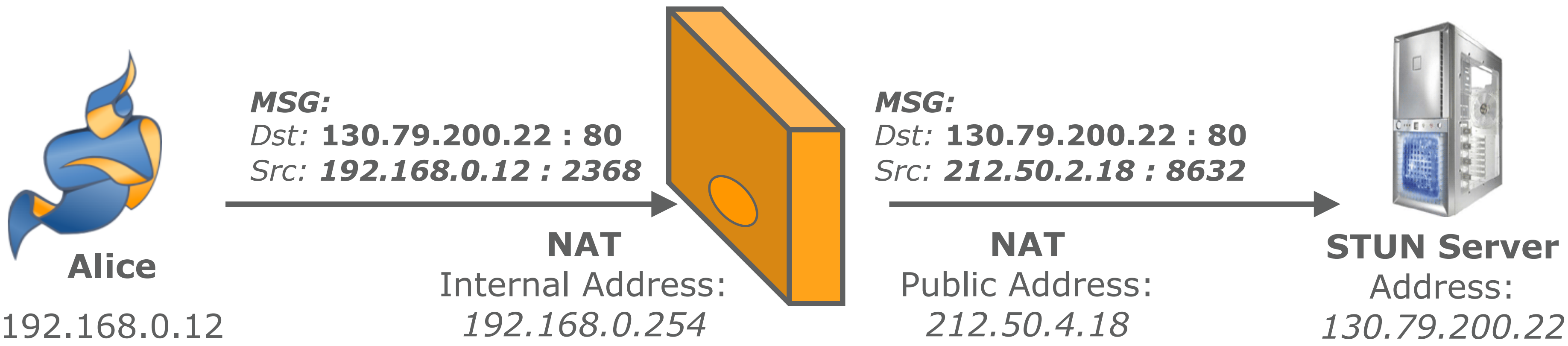
Internal **host:port**	NAT **port**	Active connections **host:port**
192.168.0.12 : 2368	***8632***	**130.79.200.22 (: 80)**
		60.55.68.53 (: 80)

Endpoint-Independent Mapping
Endpoint-Dependent Filtering



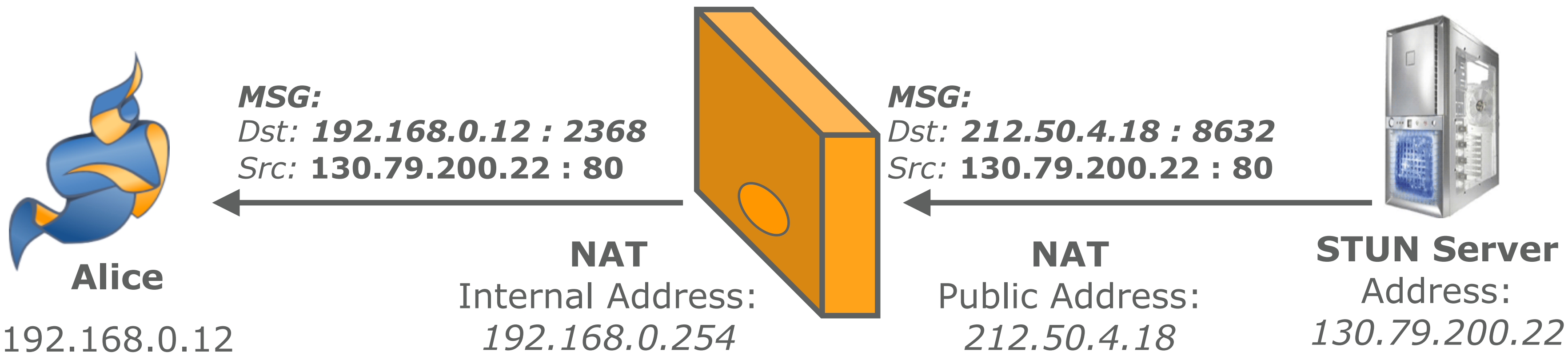
How do NATs work … Endpoint dependent mapping

Internal **host:port**	NAT **port**	Active connections **host:port**
192.168.0.12 : 2368	***8632***	**130.79.200.22 (: 80)**



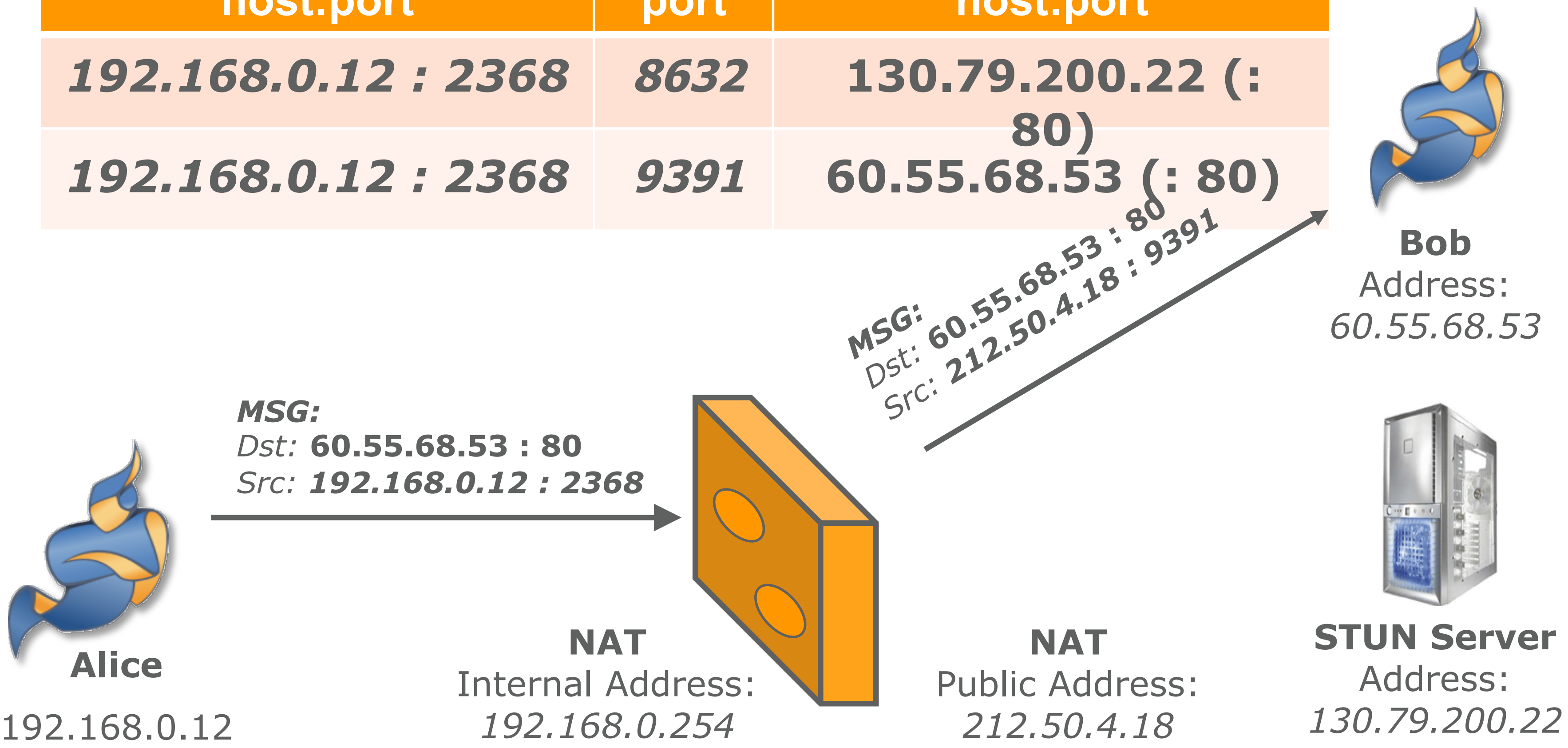
How do NATs work …
Endpoint dependent mapping

Internal **host:port**	NAT **port**	Active connections **host:port**
192.168.0.12 : 2368	***8632***	**130.79.200.22 (: 80)**



How do NATs work … Endpoint dependent mapping

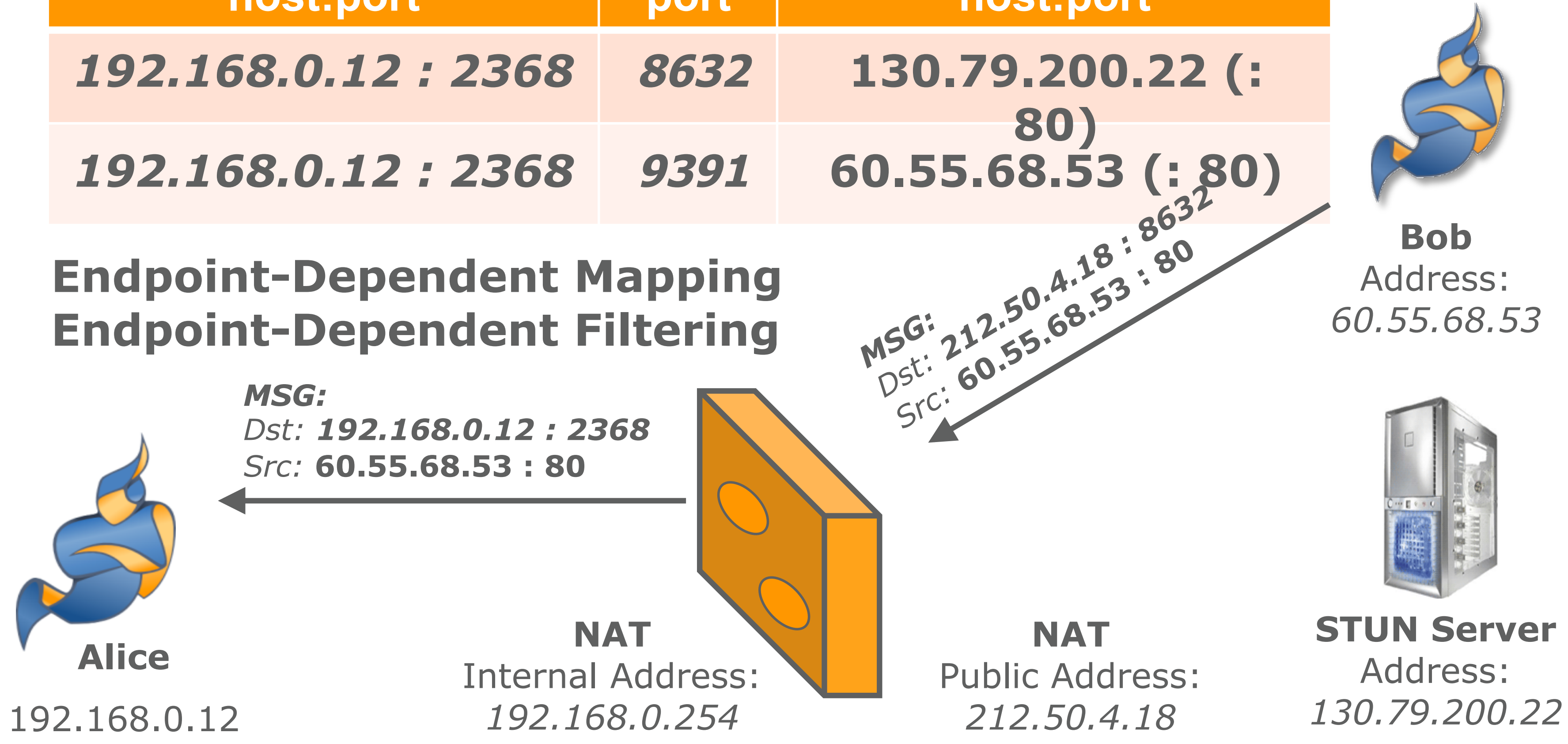
Internal **host:port**	NAT **port**	Active connections **host:port**
192.168.0.12 : 2368	***8632***	**130.79.200.22 (: 80)**
192.168.0.12 : 2368	***9391***	**60.55.68.53 (: 80)**



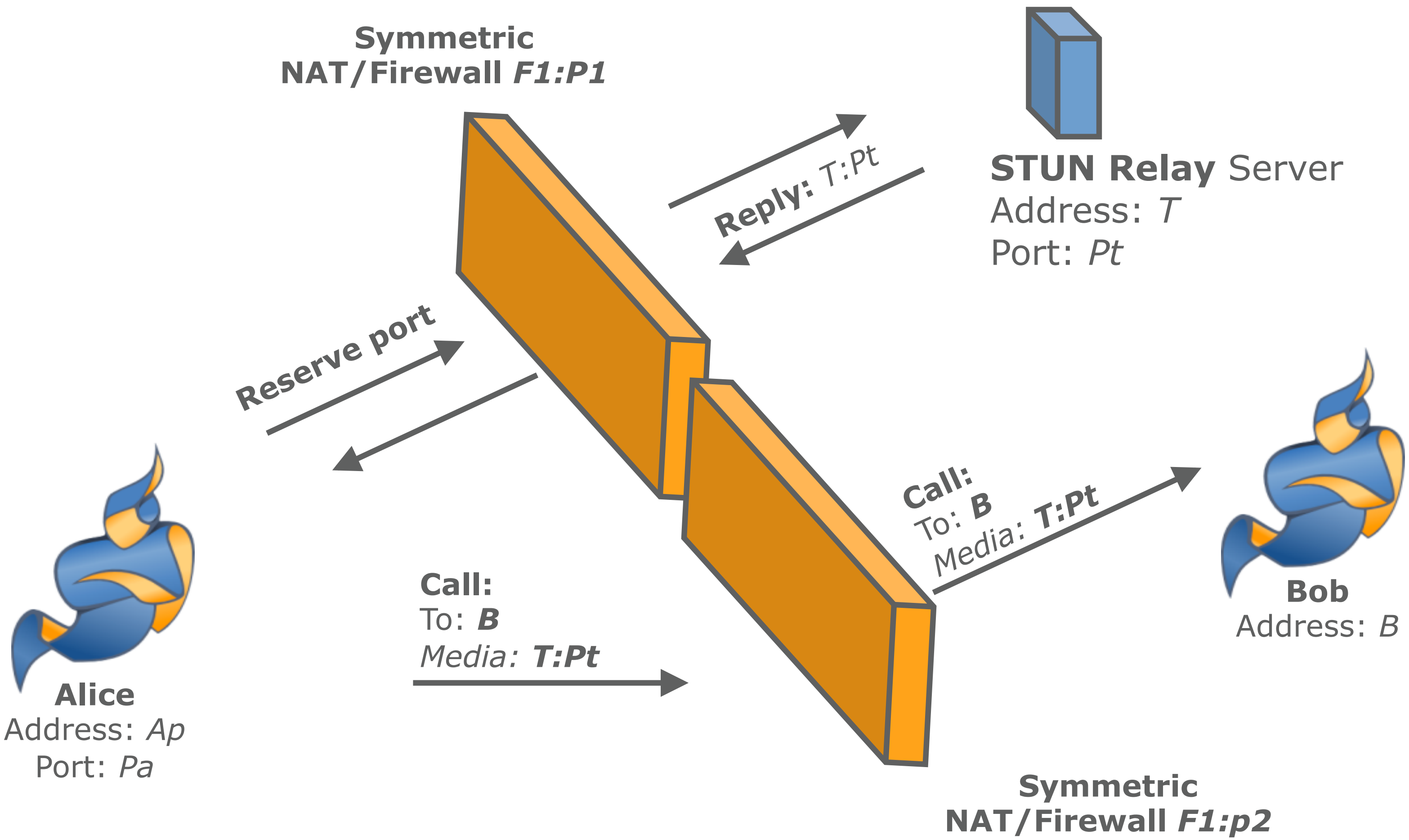
How do NATs work … Endpoint dependent mapping

Internal **host:port**	NAT **port**	Active connections **host:port**
192.168.0.12 : 2368	***8632***	**130.79.200.22 (: 80)**
192.168.0.12 : 2368	***9391***	**60.55.68.53 (: 80)**

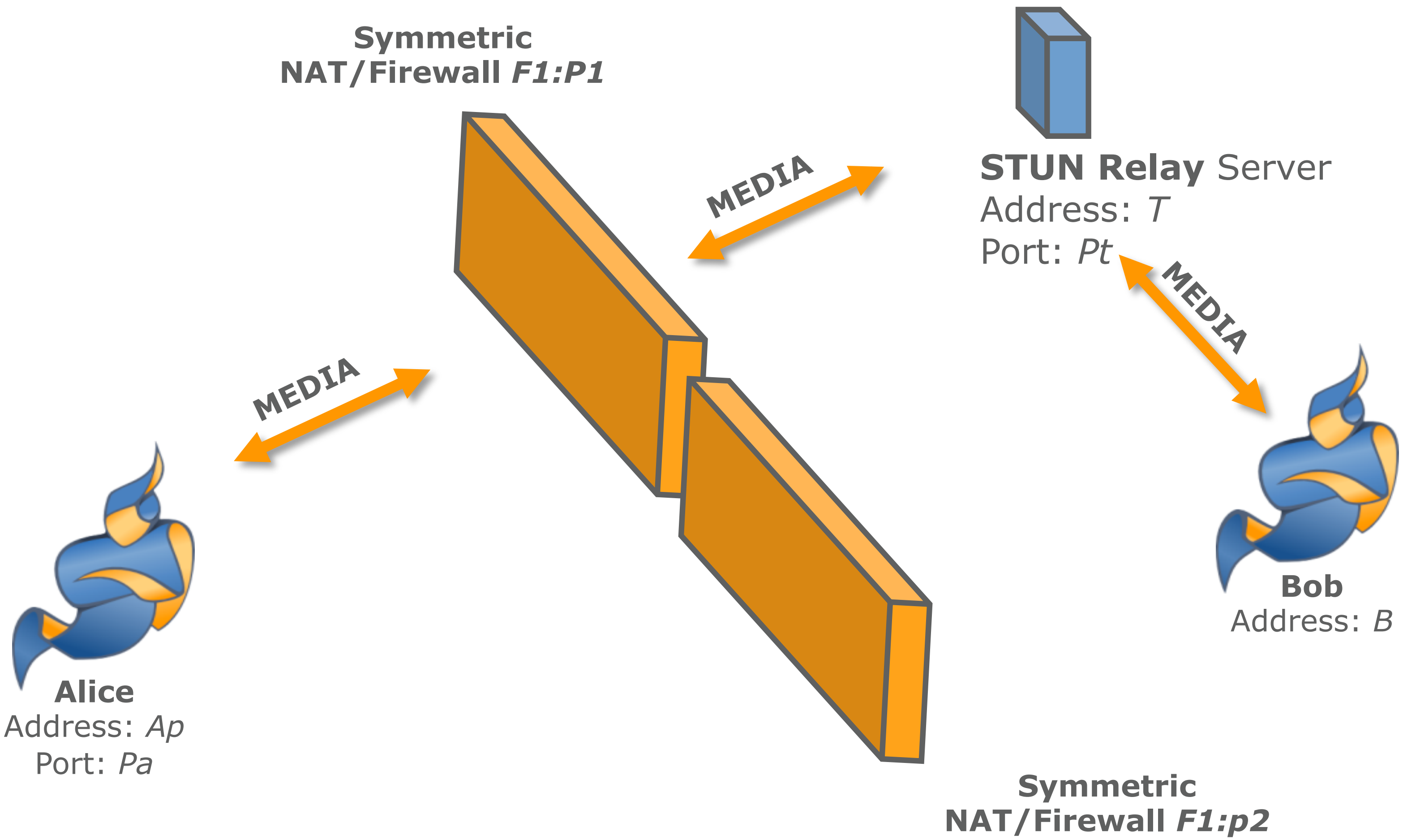
Endpoint-Dependent Mapping
Endpoint-Dependent Filtering



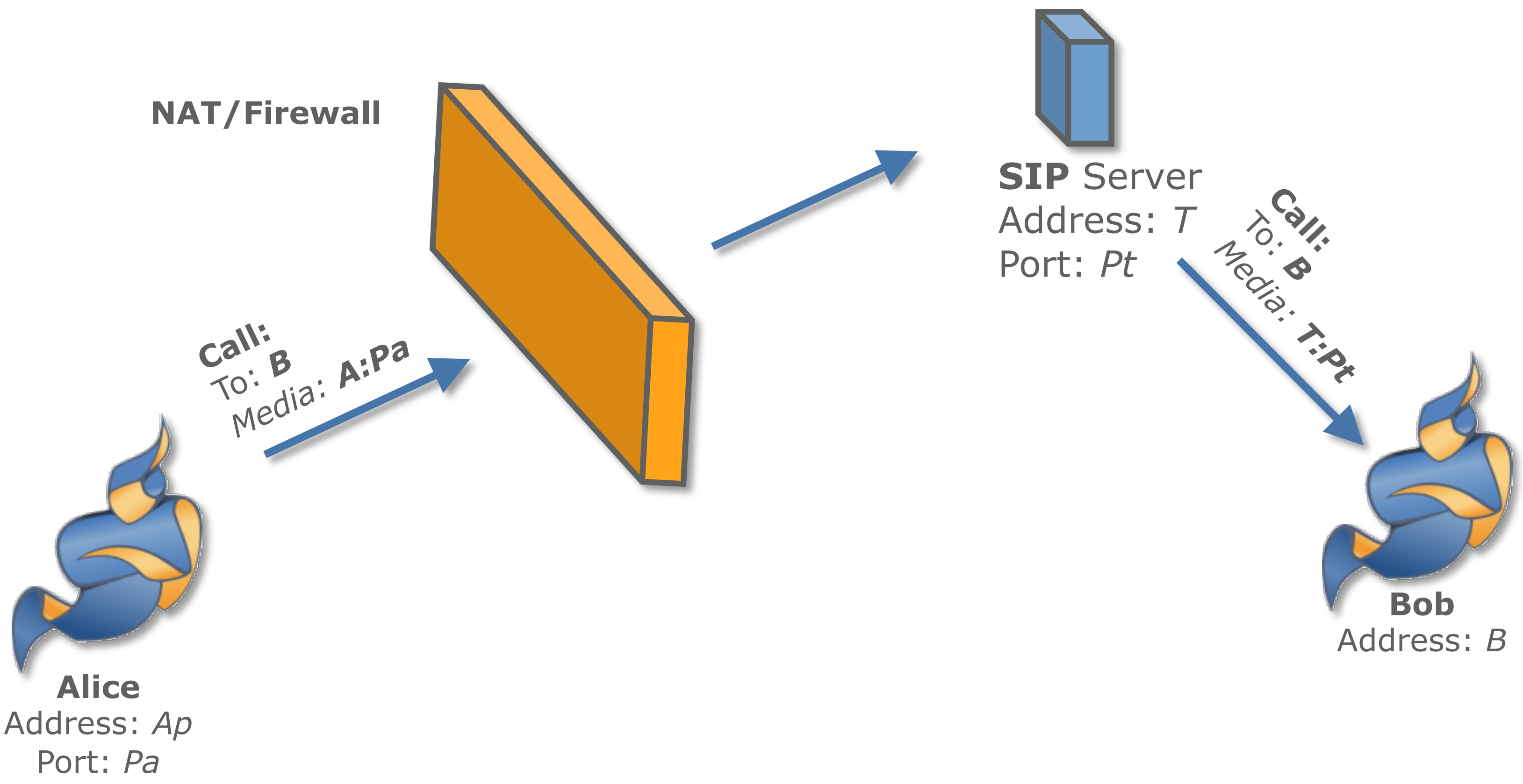
Relaying Media



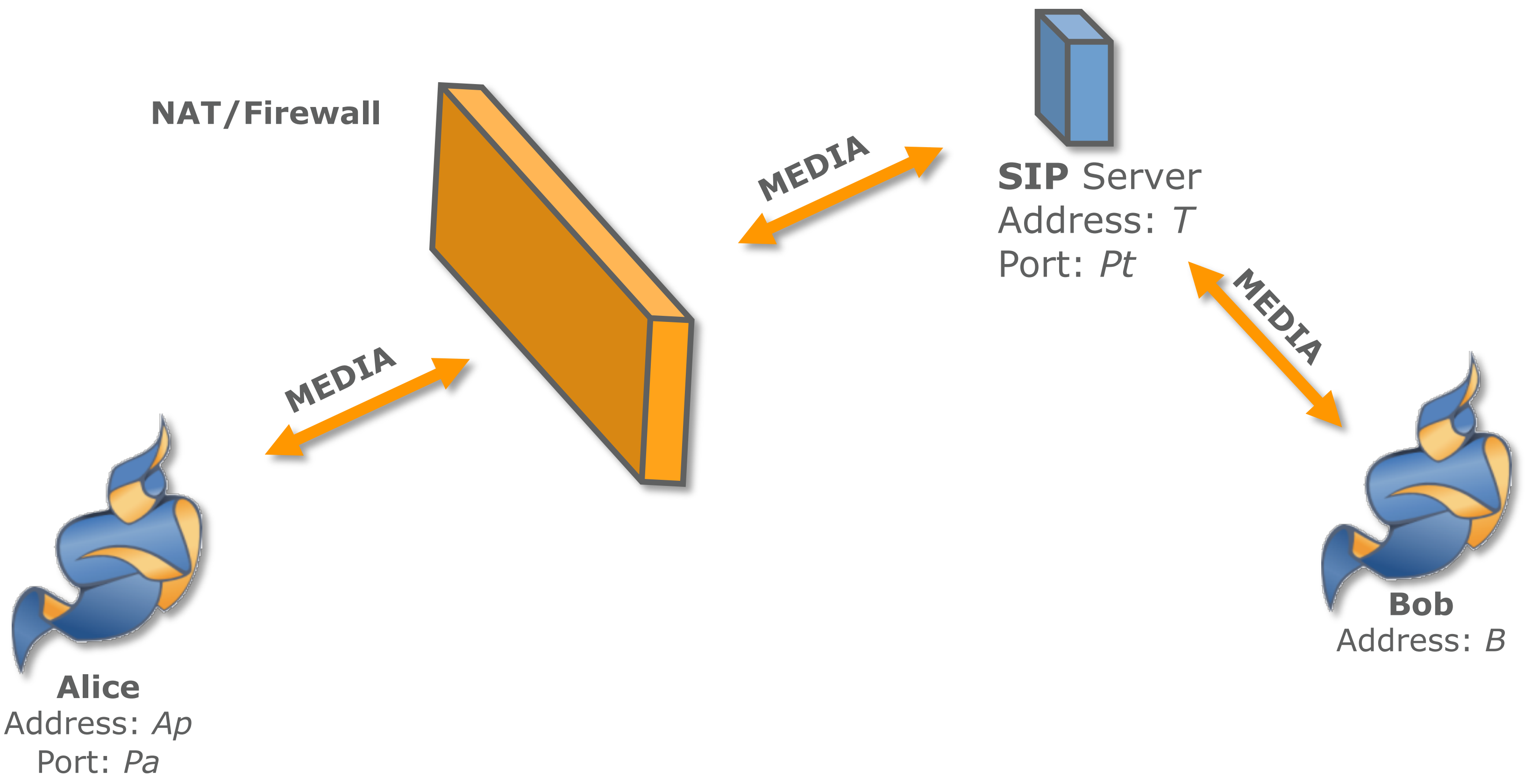
Relaying Media



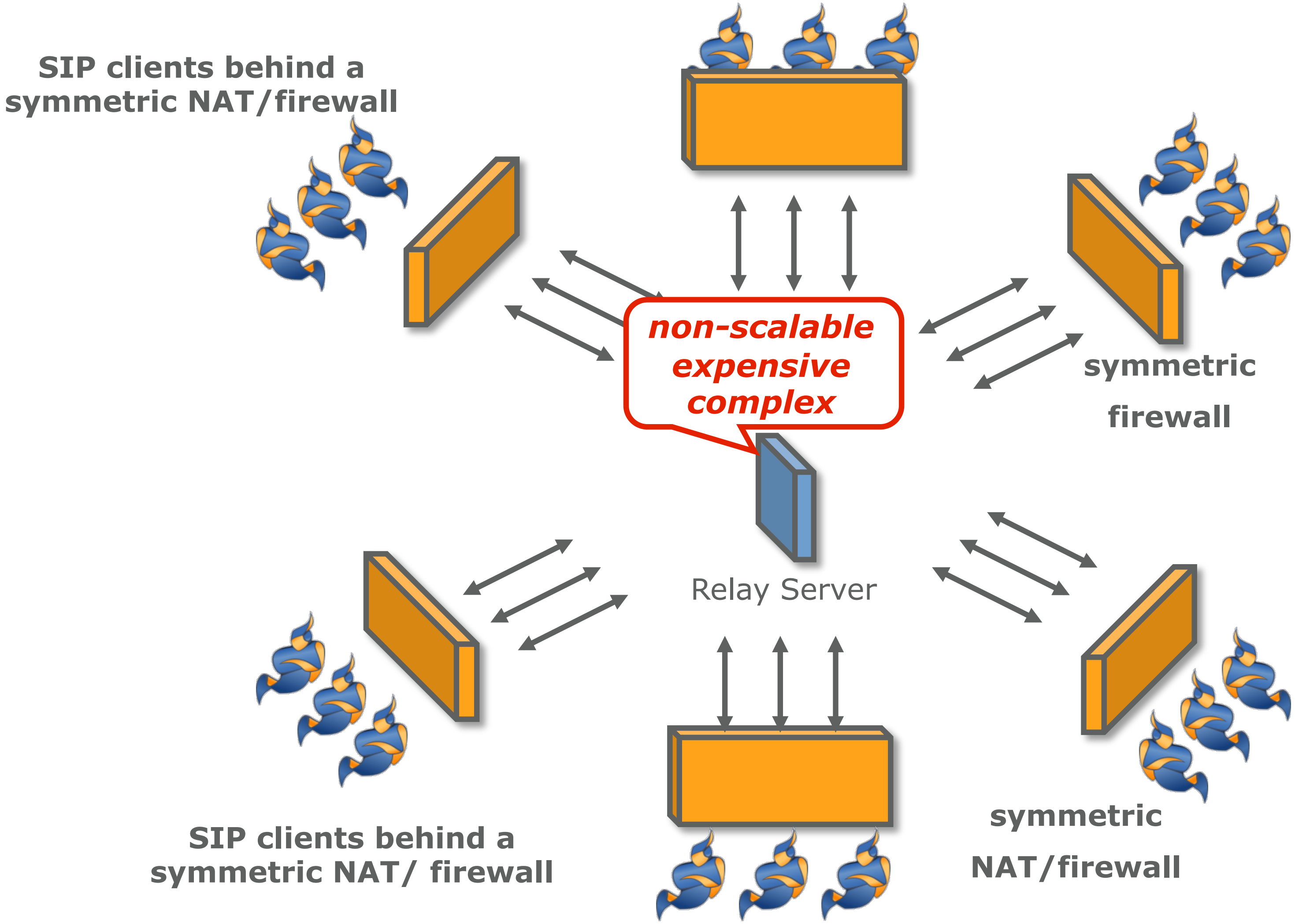
Relaying Media (The SIP Way)



Relaying Media (The SIP Way)



Relaying Media



Could we please have IPv6 now?

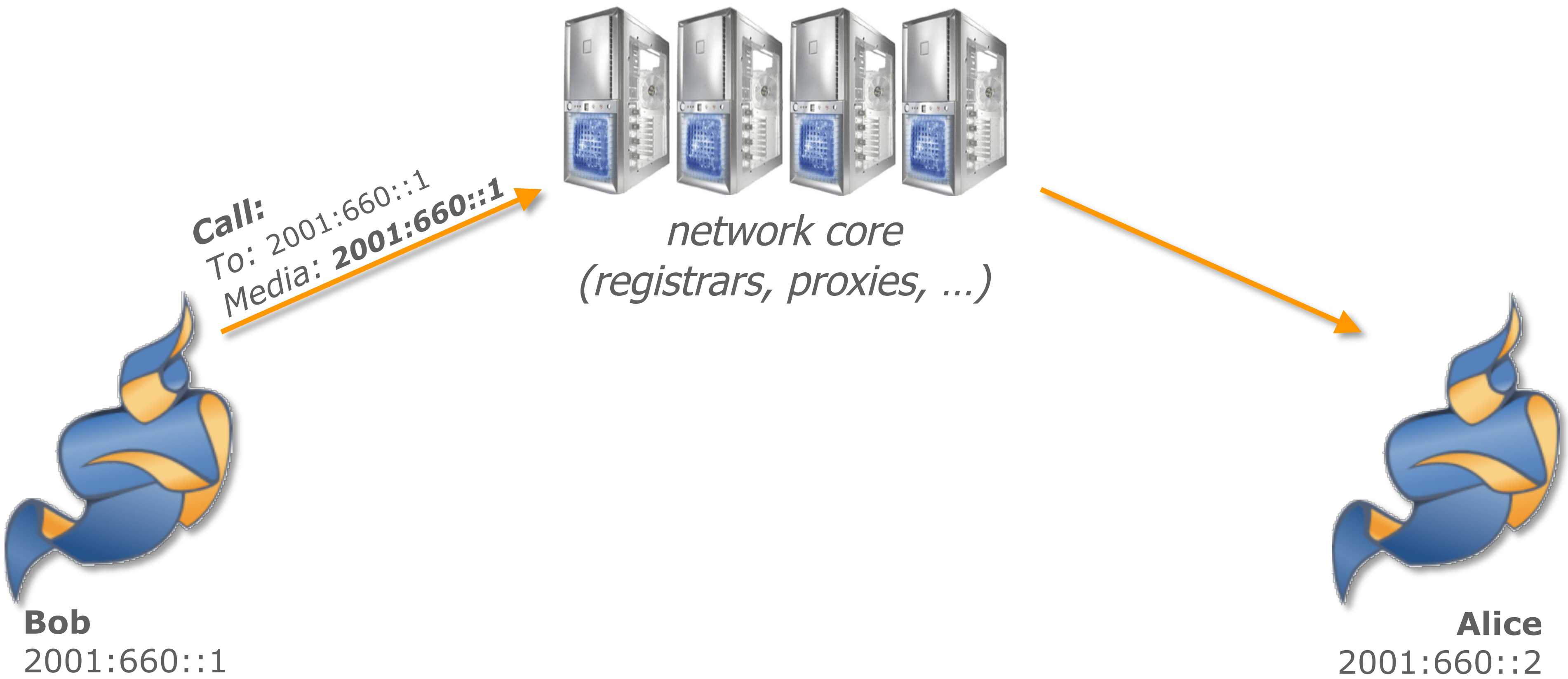
… ok, it’s probably high time we moved to IPv6 …

Could we please have IPv6 now?

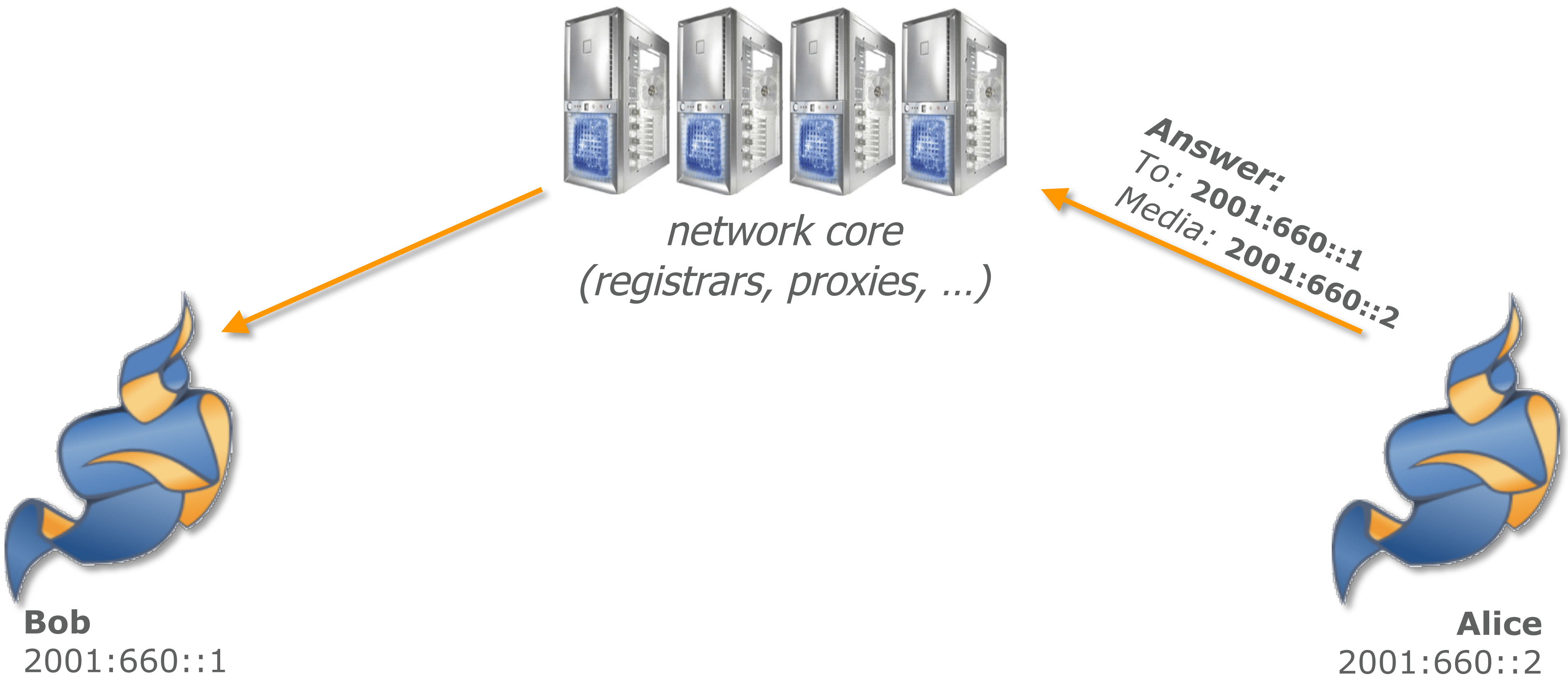
... this should simplify VoIP

... shouldn't it?

VoIP and IPv6 – demo version



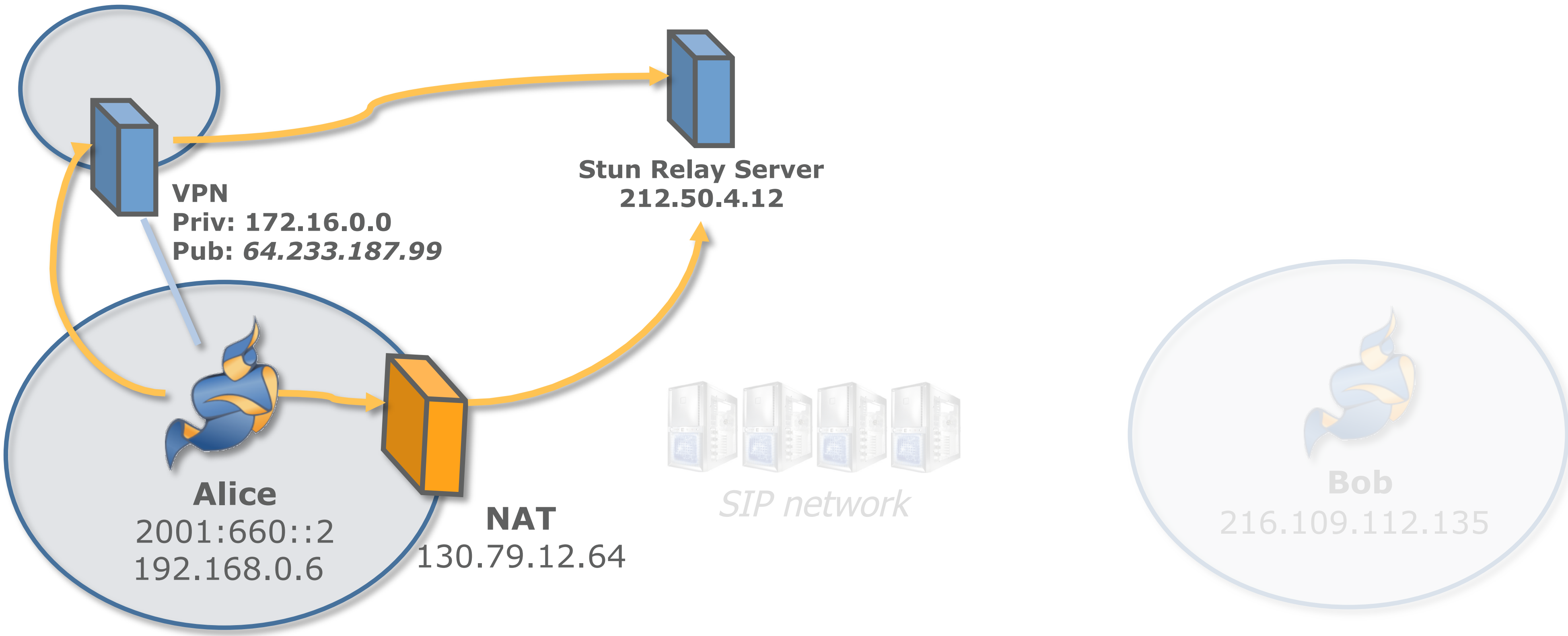
VoIP and IPv6 – demo version



VoIP and IPv6 – demo version

Reality check!

Reality check!

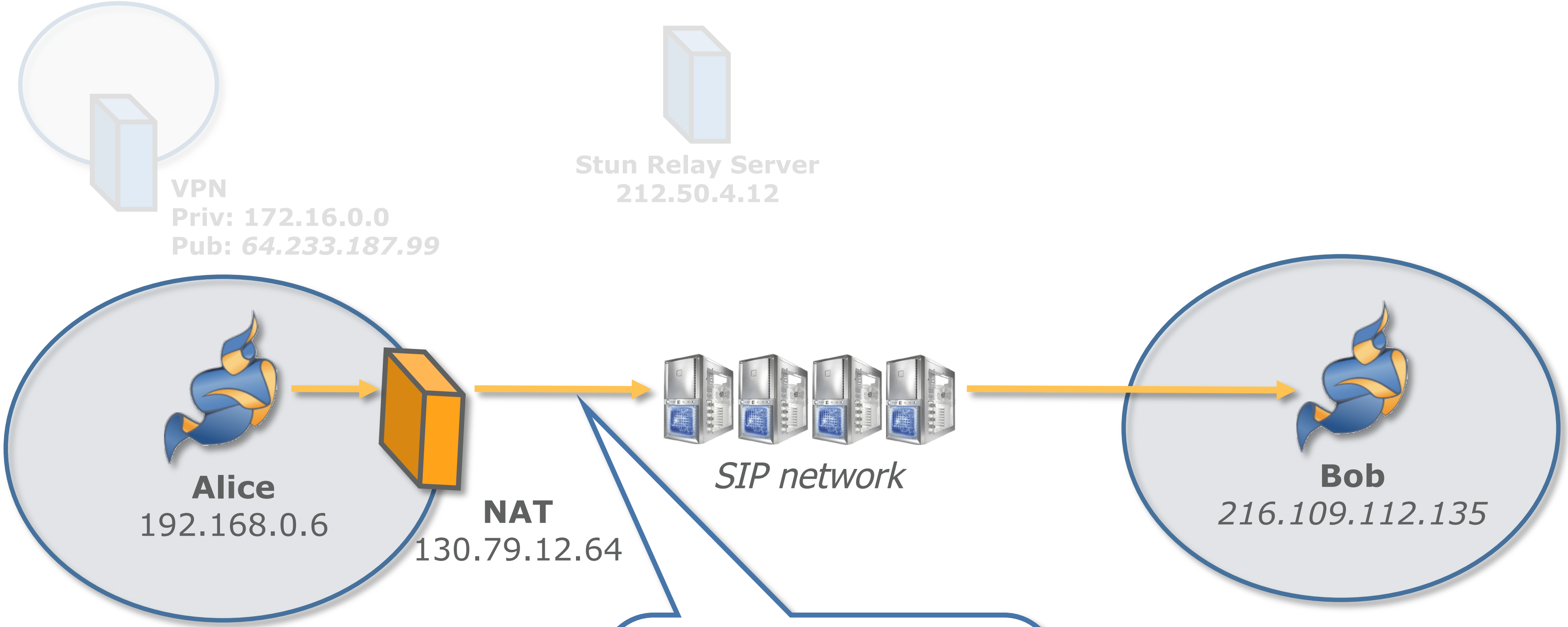


Alice's list of addresses:
2001:660::2
192.168.0.6
172.16.0.9
130.79.12.64
64.233.187.99
212.50.4.12

Interactive Connectivity Establishment (ICE)

An IETF draft brought to you by Cisco's Jonathan Rosenberg

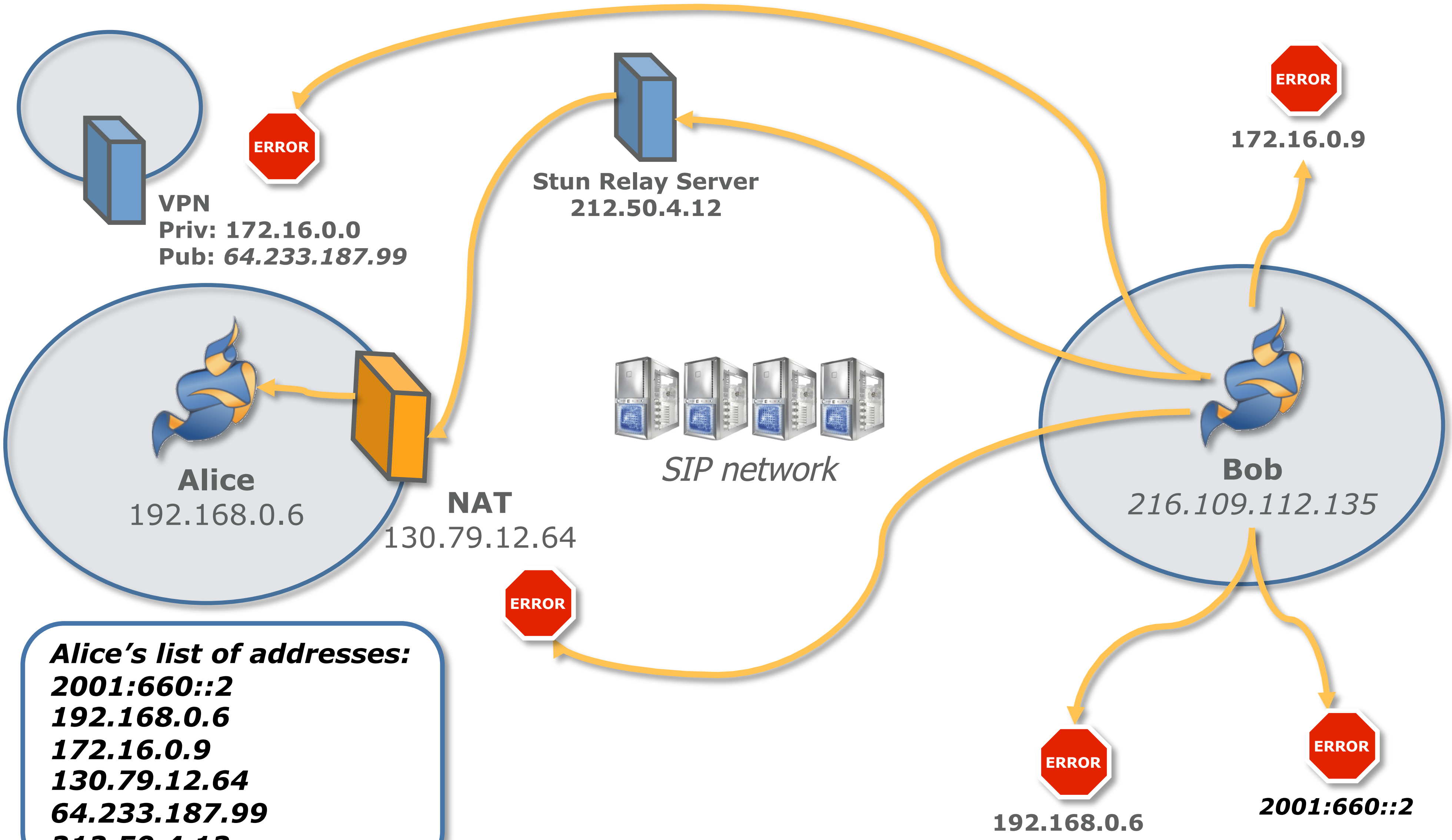
Address management …
… with ICE



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

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

Address management …
… with ICE



Alice's list of addresses:
2001:660::2
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Deployment

Asteriskv6.org (*)

- 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
    ~ Up to 250 lines through PRI Digium cards (zaptel limitation)
  - 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

Asterisk™

- 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

Open SER is no longer

OpenSIPS.org **Kamailio.org**

Media relaying with RTP Proxy

- High performance RTP stream proxy-ing
- Works with
  - SER
  - OpenSER

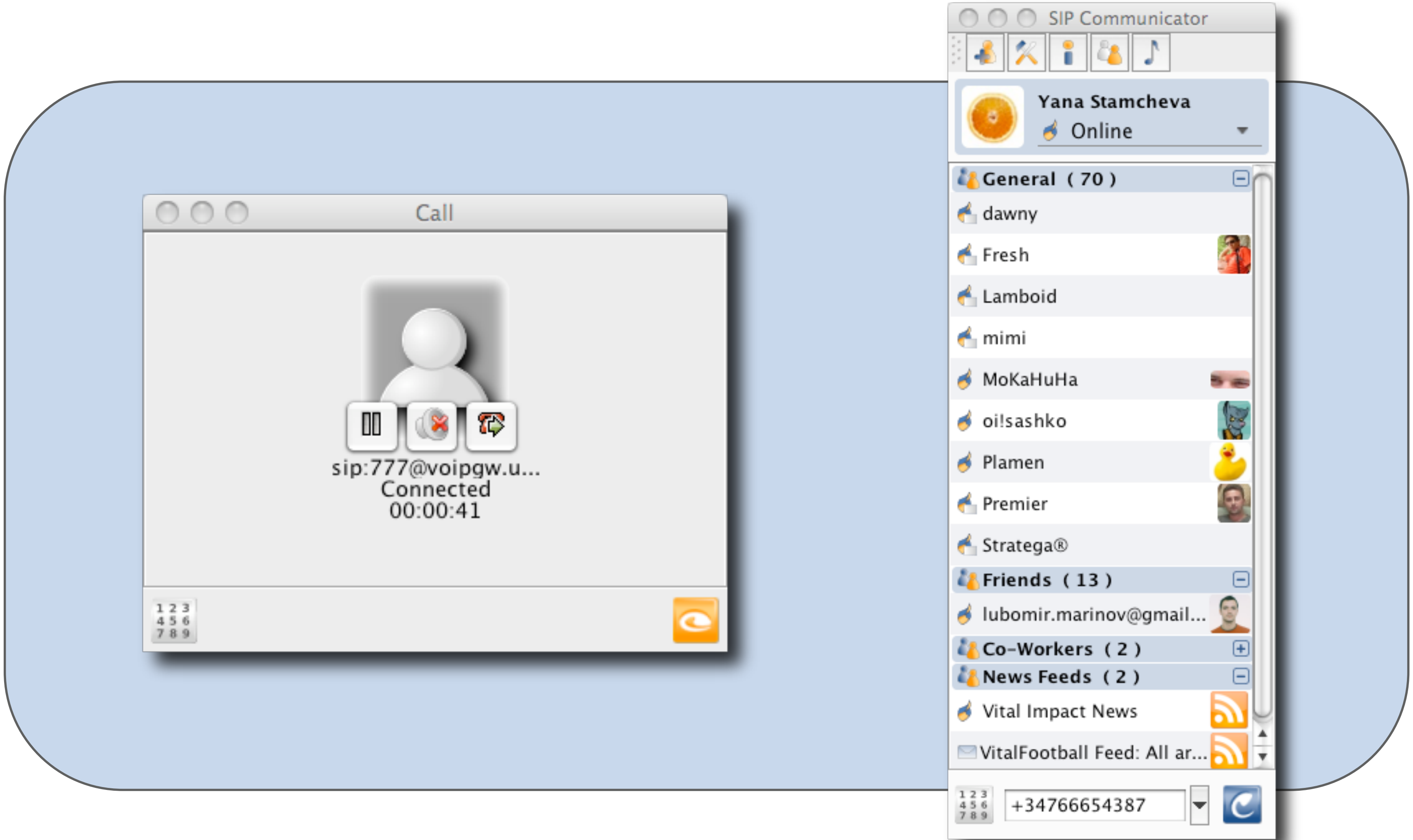
  … through their nat helper module

- Supports features such as:
  - Remote control mode
  - IPv4
  - IPv6
  - IPv4 to IPv6 relaying

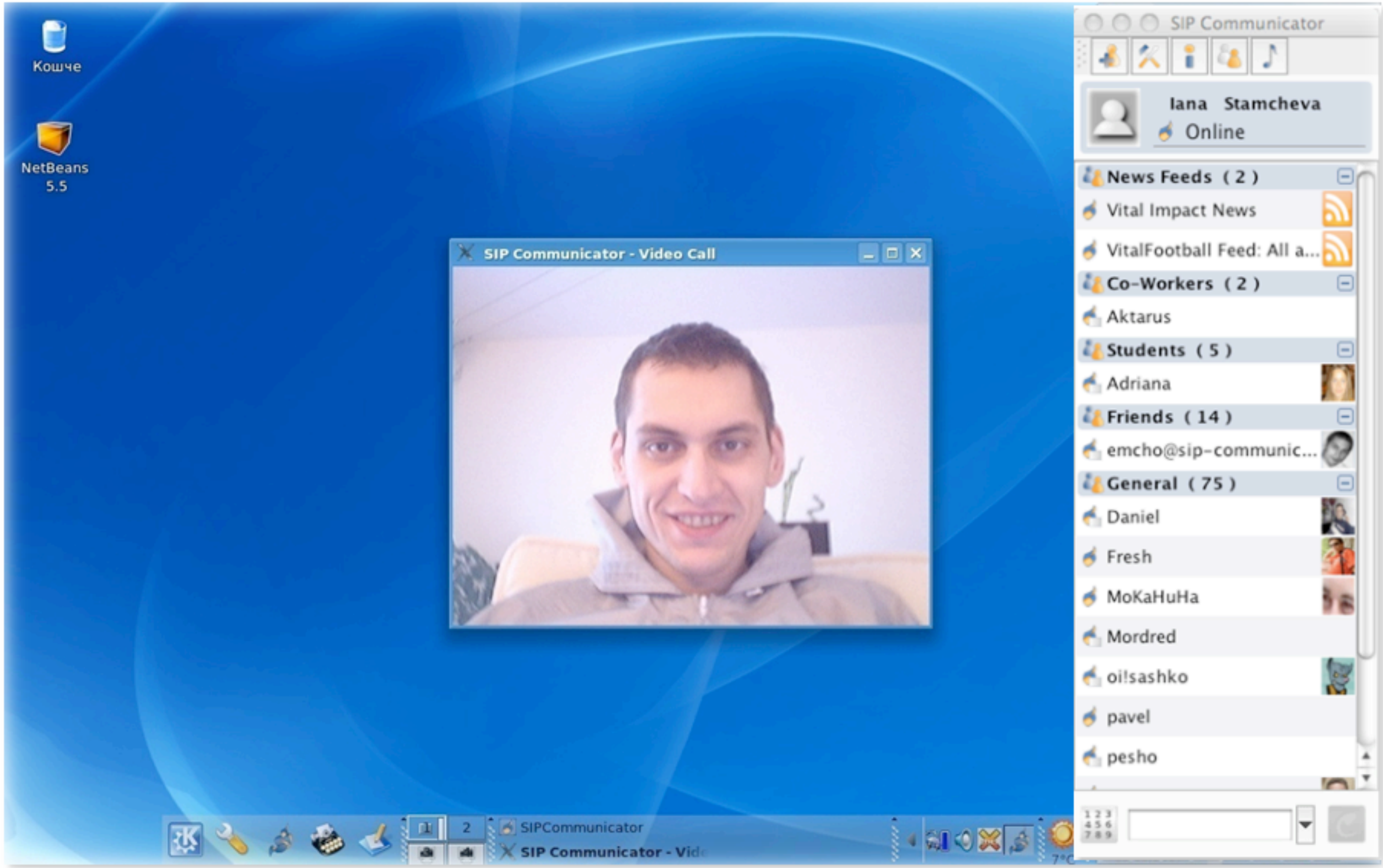
Clients

the other side

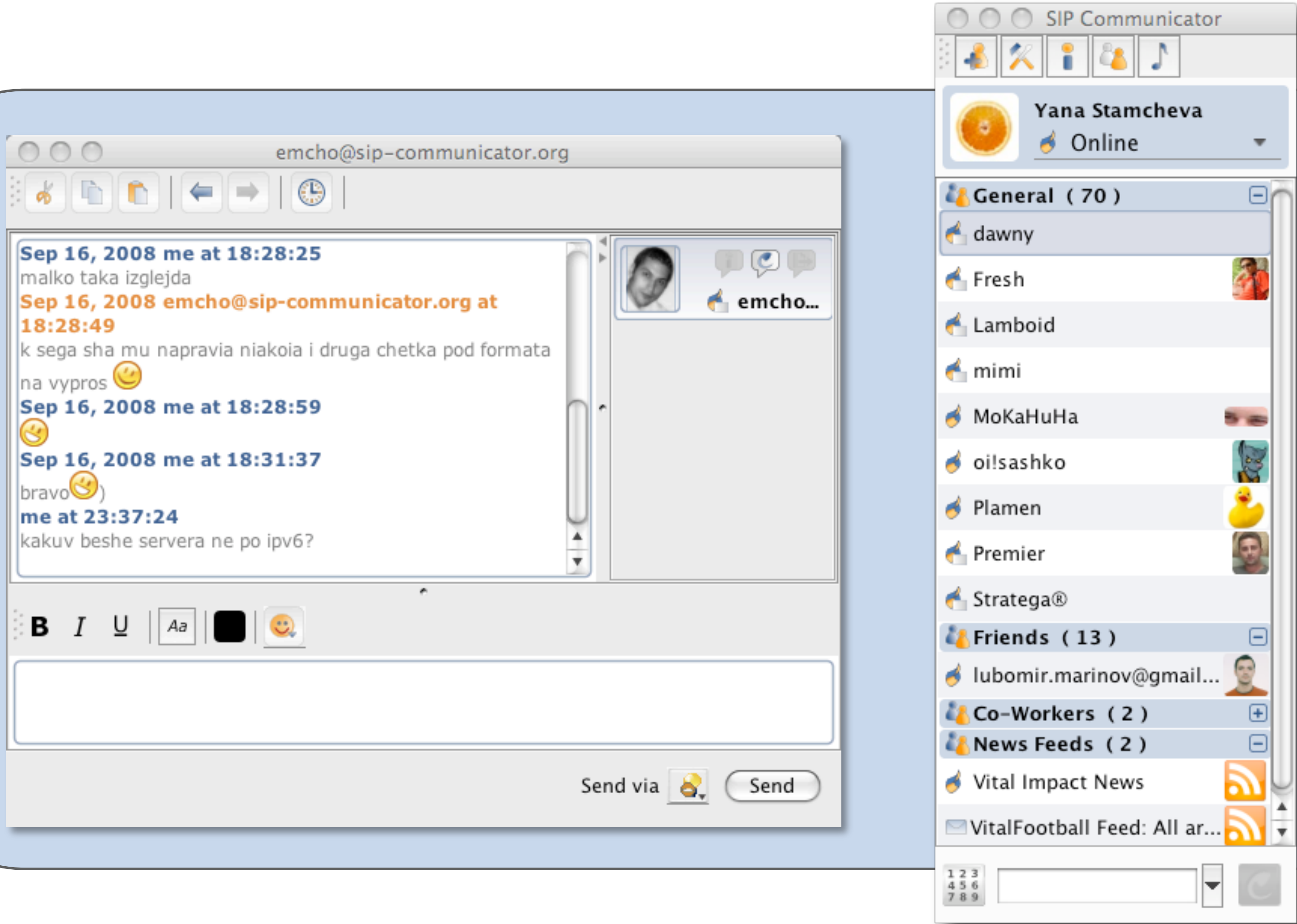
SIP Communicator Overview
Audio/Video Calls with SIP



SIP Communicator Overview
Audio/Video Calls with SIP



SIP Communicator Overview
Instant Messaging

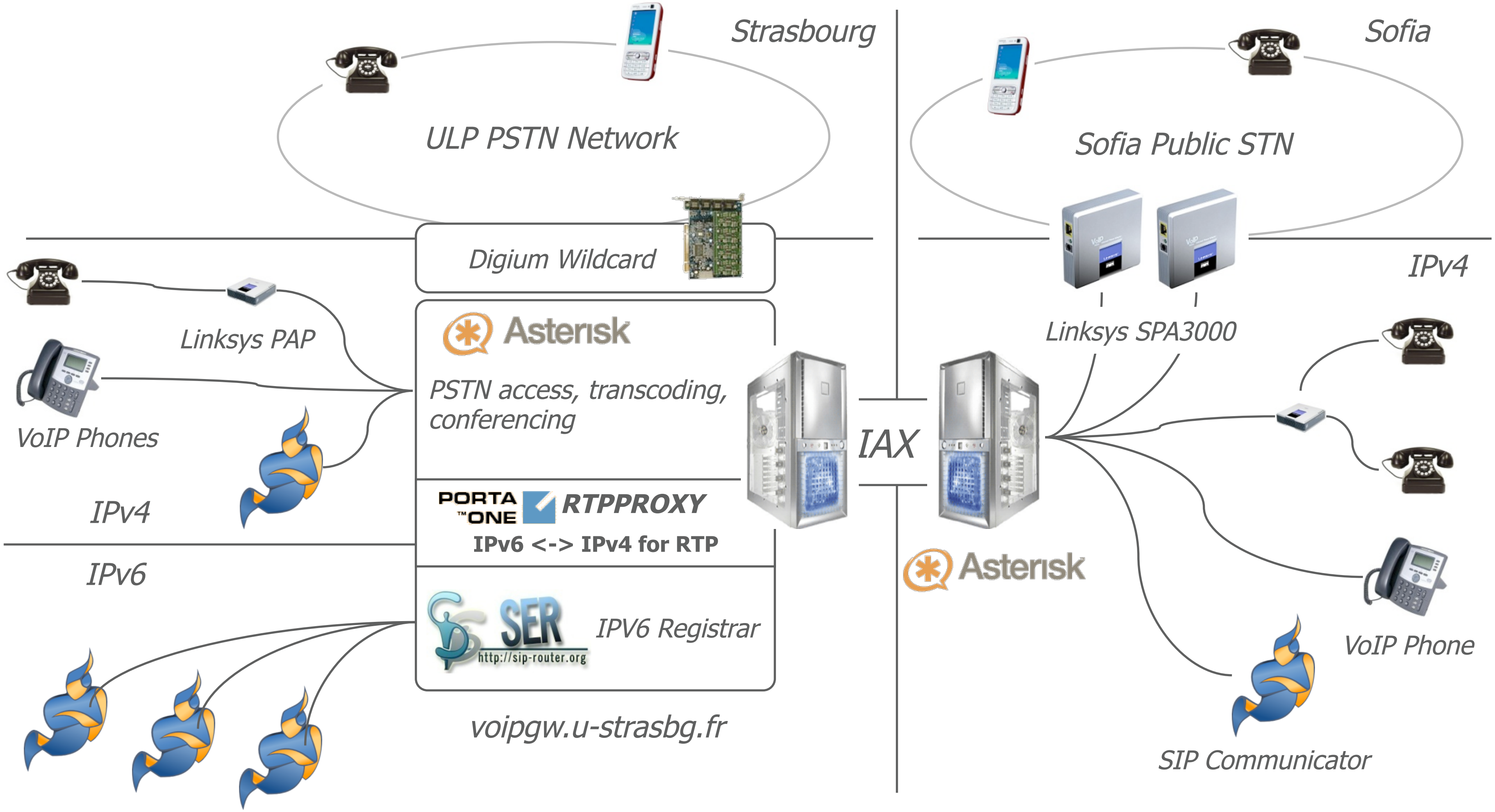


Other?

- X-Lite – http://counterpath.com
- KPhone – http://kphone.sf.net
- Linphone – http://www.linphone.org

- Hardware clients:
  - …

A Sample Deployment



VoIP Basics

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