Voice Over IP Using SIP

Interop Las Vegas May 2005
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InteropNet Labs

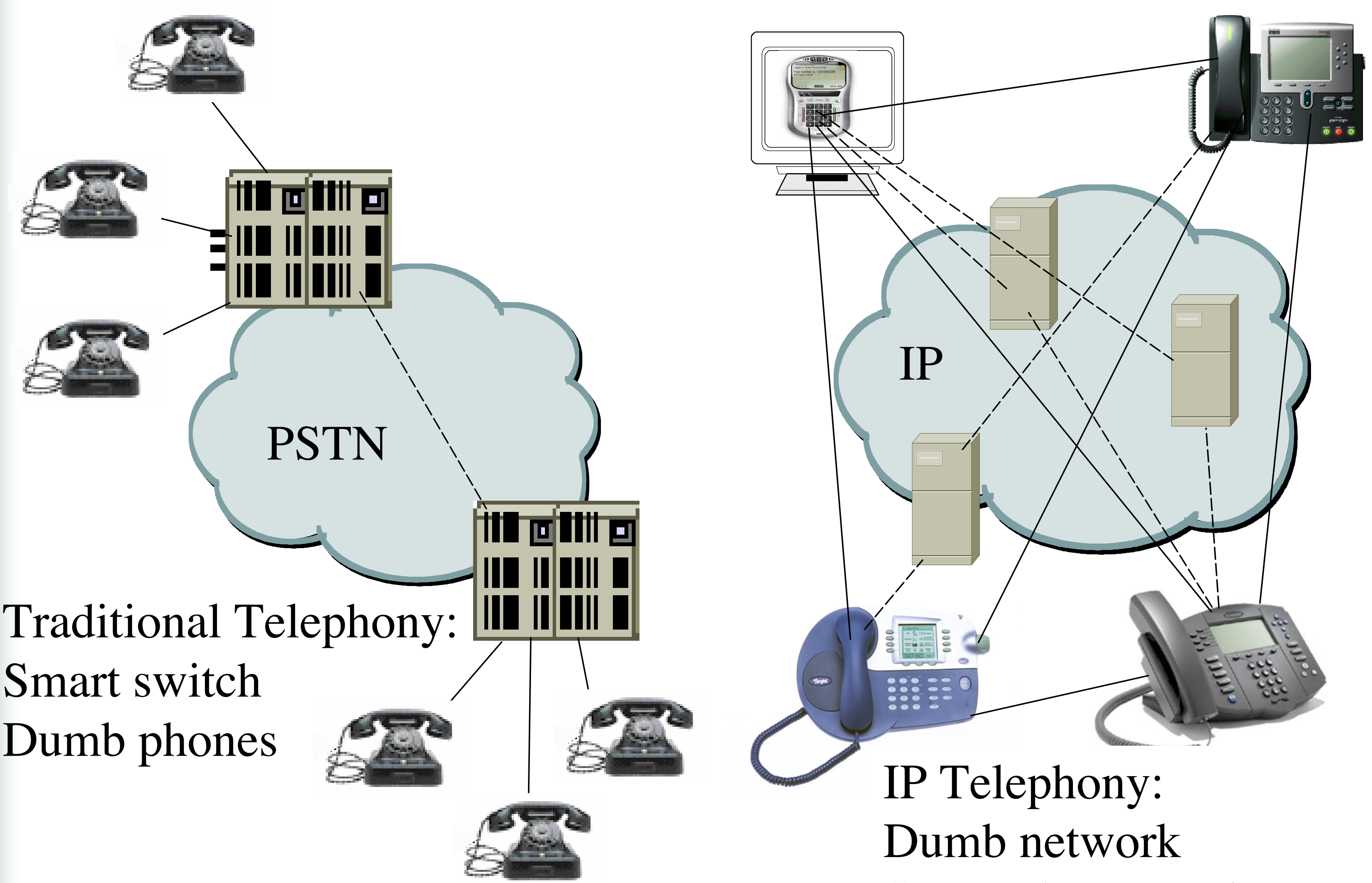
Come to the iLabs Booth

...to see, hear, touch and talk about the huge variety of SIP products we've got working.

This presentation derived in part from the excellent SIP tutorials of:

- Jiri Kuthan, iptel.org
- Dorgham Sisalem, GMD Fokus (fokus.fraunhofer.de)
- David Oran, Cisco Systems

IP Telephony is Different



Traditional Telephony:
Smart switch
Dumb phones

IP Telephony:
Dumb network
Smart phones and servers

What Protocols Are Needed for IP Telephony?

- Signaling protocol to establish presence, locate users, set up, modify and tear down sessions
- Media Transport Protocols for transmission of packetized audio/video
- Supporting Protocols for Gateway Location, QoS, interdomain AAA*, address translation, IP, etc.

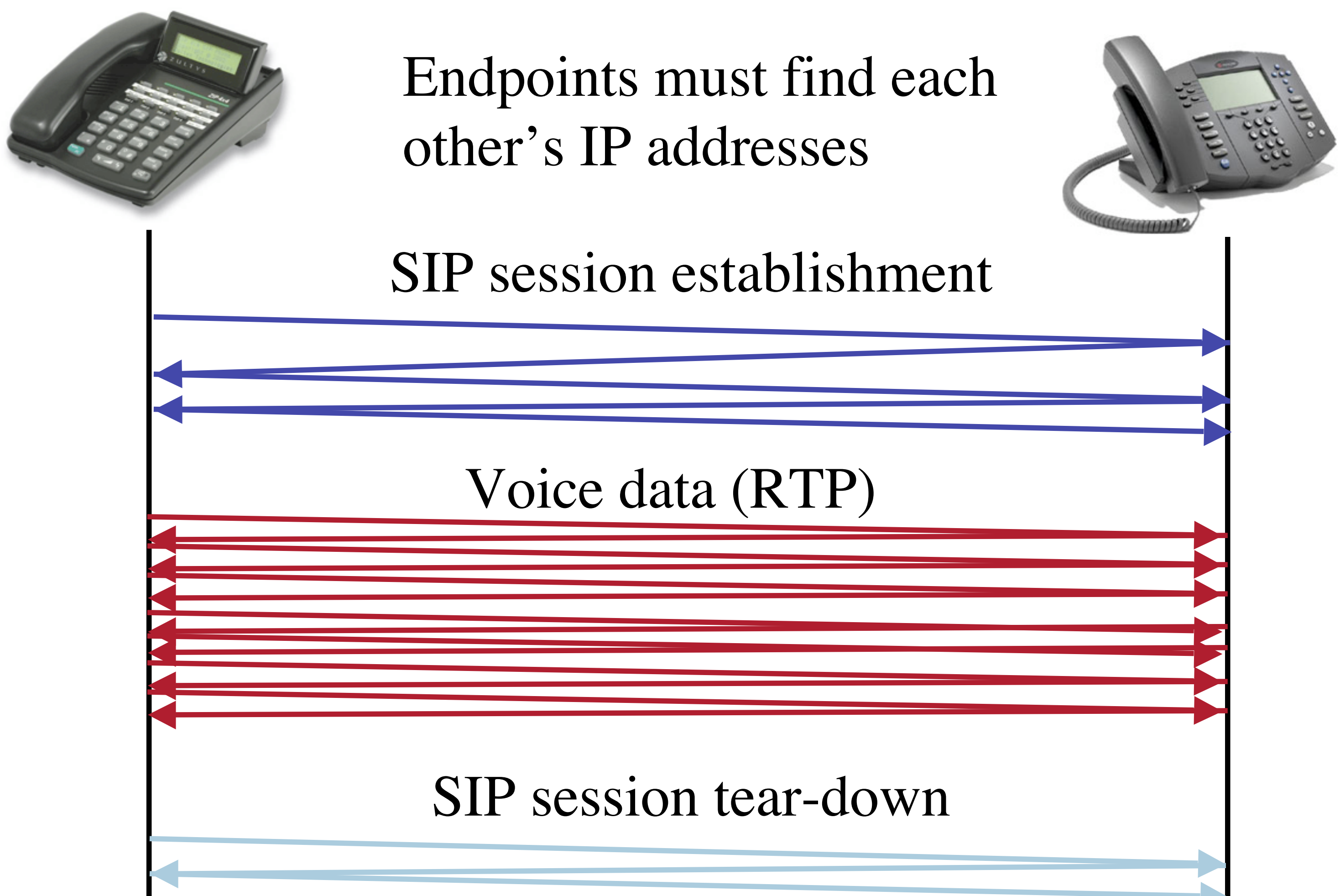
* AAA = Authentication, Authorization and Accounting

SIP is the Session Initiation Protocol

- SIP is an application layer signaling protocol
  - create, modify and terminate sessions
  - two or more participants
- Uses URL style addresses and syntax
- Flexible transport: can use **UDP**, TCP, TLS, or SCTP
- Uses SDP for describing media sessions: **Audio**, **Video**, realtime Text, **IM**, speech services, etc.
- Applications include (but not limited to): **Voice**, **video**, gaming, **instant messaging**, presence, call control, etc.
- Simple extensible protocol
  - Methods—Define transaction
  - Headers—Describe transaction
  - Body—SDP and other MIME content

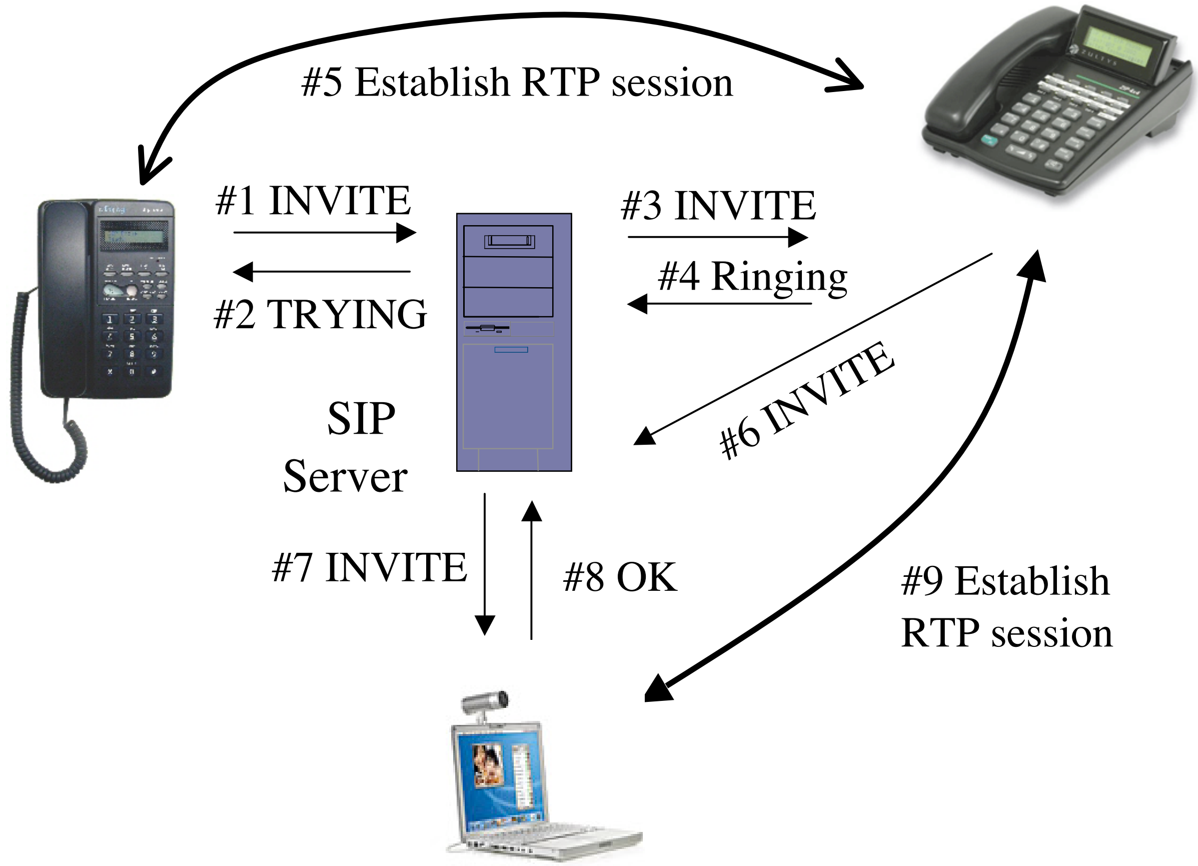
VoIP uses a subset of SIP capabilities

Basic Call With SIP

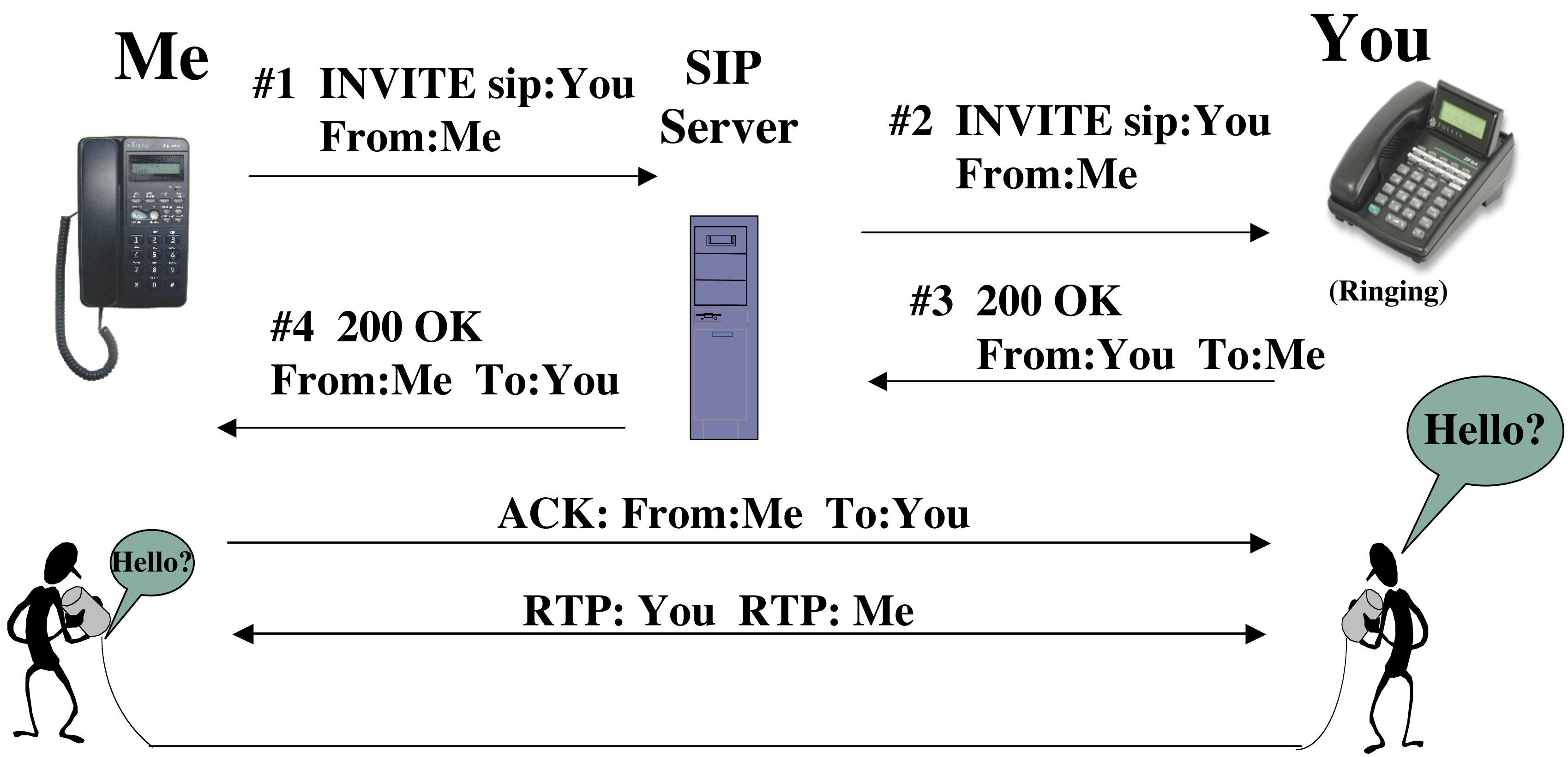


A More Complex Call With SIP

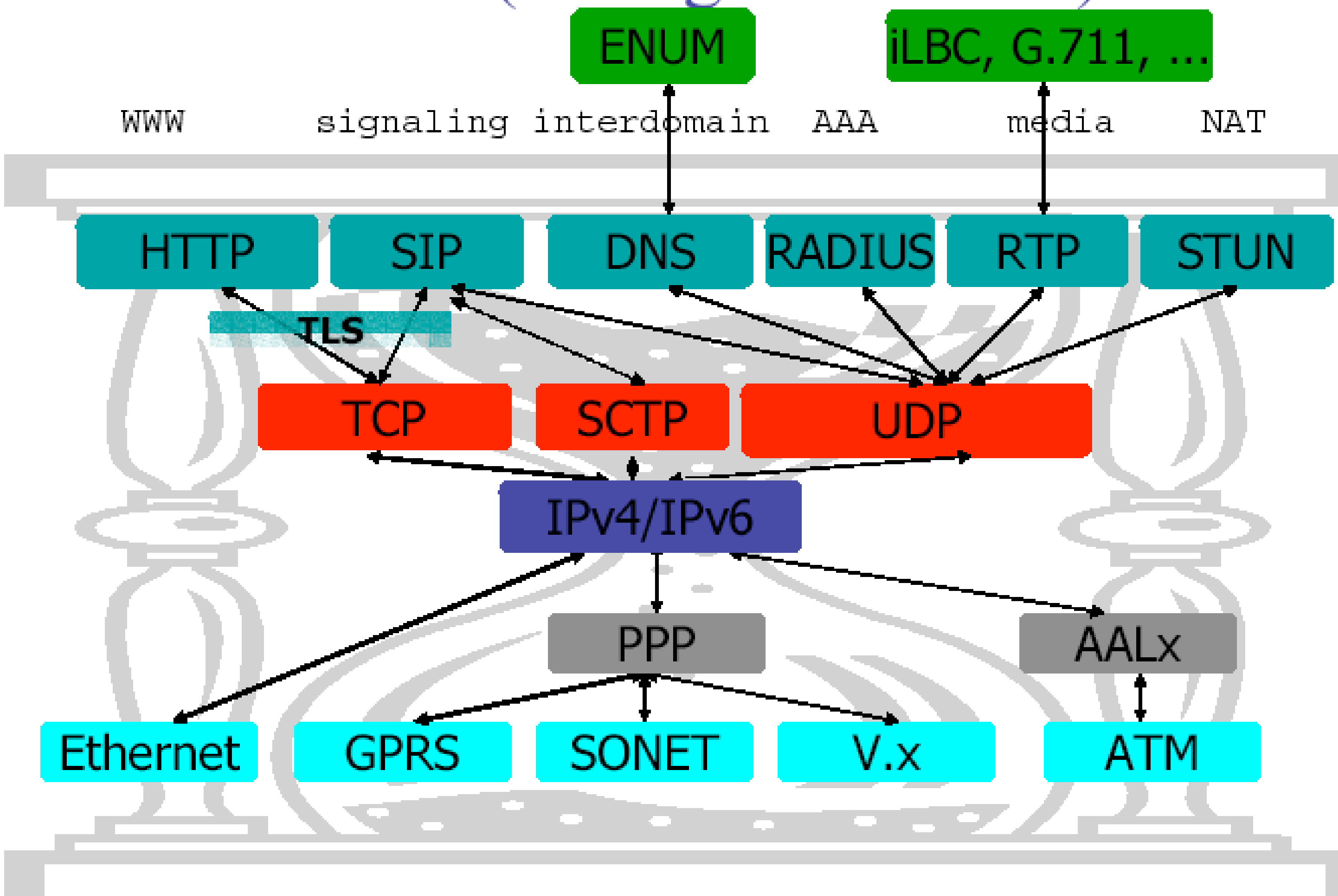
Endpoints find each other's
IP addresses using a SIP Proxy



SIP Packets Flow Through a SIP Proxy Only During Setup

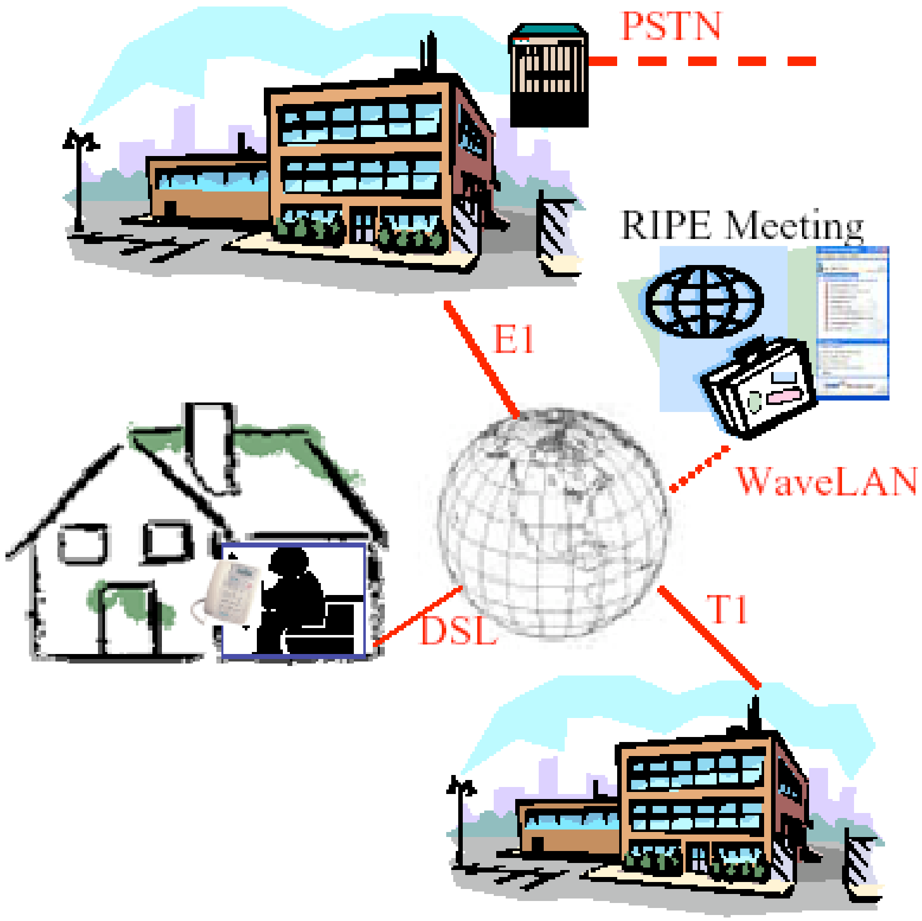


Protocol Zoo (Hourglass Model)



Jiri Kuthan, iptel.org, October 2003

VoIP in the Enterprise



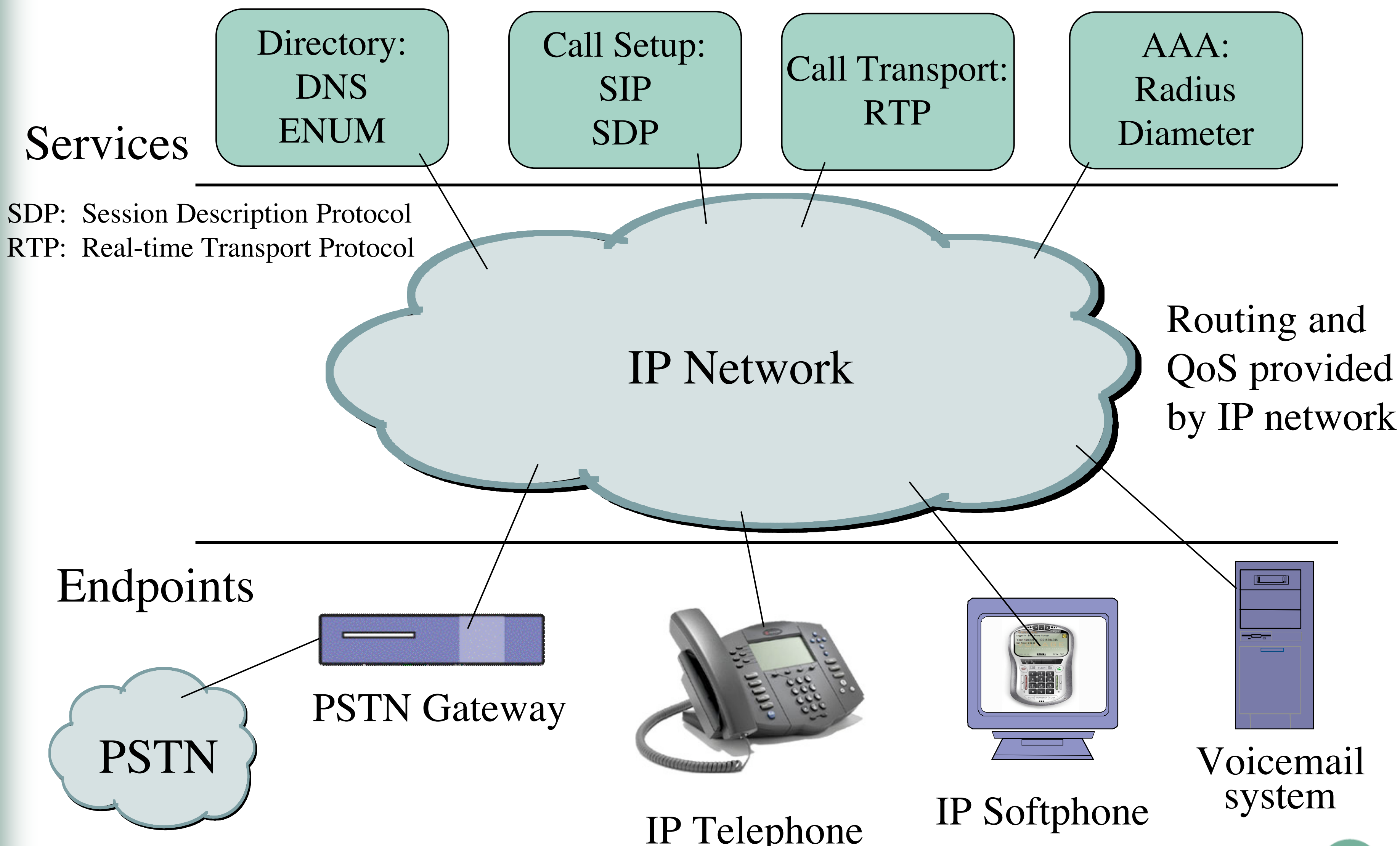
Services available to all company's users, on-site, offsite and multi-site – toll bypass.

- No telephone line required for home-workers and remote offices.
- Single infrastructure for data and voice.
- Effectiveness tools.
- Service operation can be outsourced in a Centrex-like manner (MCI Advantage). Like with web/email, single server may host multiple domains.

SIP Makes VoIP Easy and Interoperable

- IETF development, learning from HTTP experience, leads to (eventually) excellent interoperability
- Becoming an IP-Telephony operator takes complexity comparable to setting up E-mail server:
  - – Configure DNS
  - – Download and configure a SIP proxy server
  - – Configure supporting services: web provisioning, database back-end typically.
  - – Configure PSTN gateway for use with your proxy server.

SIP Architecture is Easy to Understand



SIP Addresses are Global

- SIP gives you a globally reachable address.
  - Callees bind to this address using SIP REGISTER method.
  - Callers use this address to establish real-time communication with callees.
- URLs used as address data format; examples:
  - sip:crw@transcendental.com
  - sip:voicemail@iptel.org?subject=callme
  - sip:17005553171@asterisk.sip.ilabs.interop.net
- must include host, may include user name, port number, parameters (e.g., transport), etc.
- may be embedded in Web pages, email signatures, printed on your business card, etc.
- address space unlimited
- non-SIP URLs can be used as well (mailto:, http:, ...)

SIP Leverages Internet Infrastructure Such as DNS

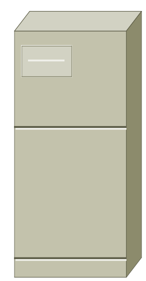
A SIP URI

sip:**crw@transcendental.com**

DNS NAPTR

DNS SRV

+1 700 555 3171



SIP Proxy Server
sipgw9.sip.transcendental.com

SIP uses DNS to Find Addresses

A SIP URI: sip:17005553171@asterisk.sip.ilabs.interop.net

SRV records in DNS are used to find SIP services:

```
_sip._udp.domain in SRV <priority> <weight> <port> server
```

Prepend server dns name with _sip and _IP transport

priority - lower numbers are chosen first

With multiple SRV records of different priority the lowest numbered server will be tried first

weight - higher numbered entries get more connects

With multiple SRV records of same priority, more connects will go to servers with higher weights

Example: this is an entry in the sip.ilabs.interop.net domain

```
_sip._udp.asterisk in SRV 10 10 5060 asterisk.sip.ilabs.interop.net.
```

A SIP endpoint will query DNS to find a SIP server for call setup to a SIP URI

ENUM Maps Telephone Numbers into DNS

A telephone number: +1 702 555 3171

NAPTR records in DNS are used to find SIP URIs for telephone numbers:

Flags and Service fields are typically fixed for SIP entries

```
domain in a NAPTR <order> <pref> "s" "SIP+D2U" <regexp> <target>
```

order in which records are read, looking for a match in capability

preference - lower numbers are chosen first

With multiple NAPTR records of the same order, the lowest number will be tried first. May be overriden to select transport.

Regular expression used to do substitution from domain into target string

Target SIP URI which may be static, or composed from domain using regex expression

The top-level domain e164.arpa is used like in-addr.arpa. A phone number like 17025553171 will have an NAPTR entry for 1.7.1.3.5.5.5.2.0.7.1.e164.arpa:

```
1.7.1.3.5.5.5.2.0.7.1.e164.arpa  in a naptr 1 10 "s" "SIP+D2U"  ""
"sip:crw@transcendental.com"
```

E.164 is the ITU standard for telephone numbering. ENUM is the the RFC standard for mapping telephone numbers into DNS.

A SIP endpoint will query DNS to find a SIP server for call setup to a telephone number

SIP Endpoints are Intelligent

Zultys ZIP4x4

Endpoints are User Agents

UA Client (originates calls)

UA Server (listens for incoming calls)

both SW and HW available

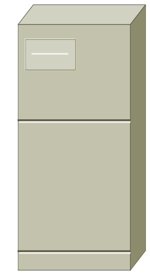
ZyXEL
Prestige 2000W
VoIP Wi-Fi Phone

Xten eyeBeam
video softphone

Grandstream analog
phone adapter
(FXS gateway)

ipDialog
Siptone II

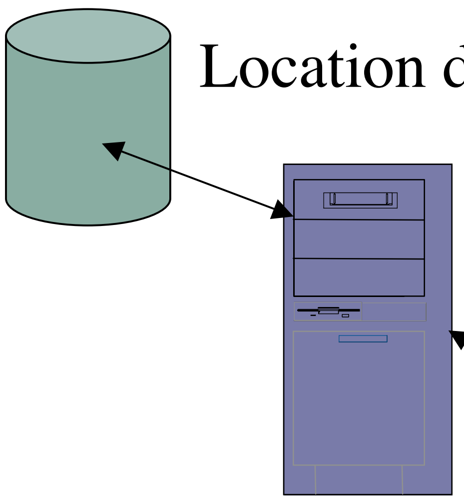
SIP Servers Perform Call Setup



- SIP Registrar
  - accept registration requests from users
  - maintains user’s whereabouts at a Location Server (like GSM HLR)
- SIP Proxy Server
  - relays call signaling, i.e. acts as both client and server
  - operates in a transactional manner, i.e., it keeps no session state
  - transparent to end-devices
  - does not generate messages on its own (except ACK and CANCEL)
  - allows for additional services (call forwarding, AAA, forking, etc.)
- SIP Redirect Server
  - redirects callers to other servers
  - Used rather rarely as operators appreciate staying in communication path. May be used to achieve very scalable load distribution.

These are logical functions and are usually on the same server

A SIP Registrar Helps Mobility



Location database

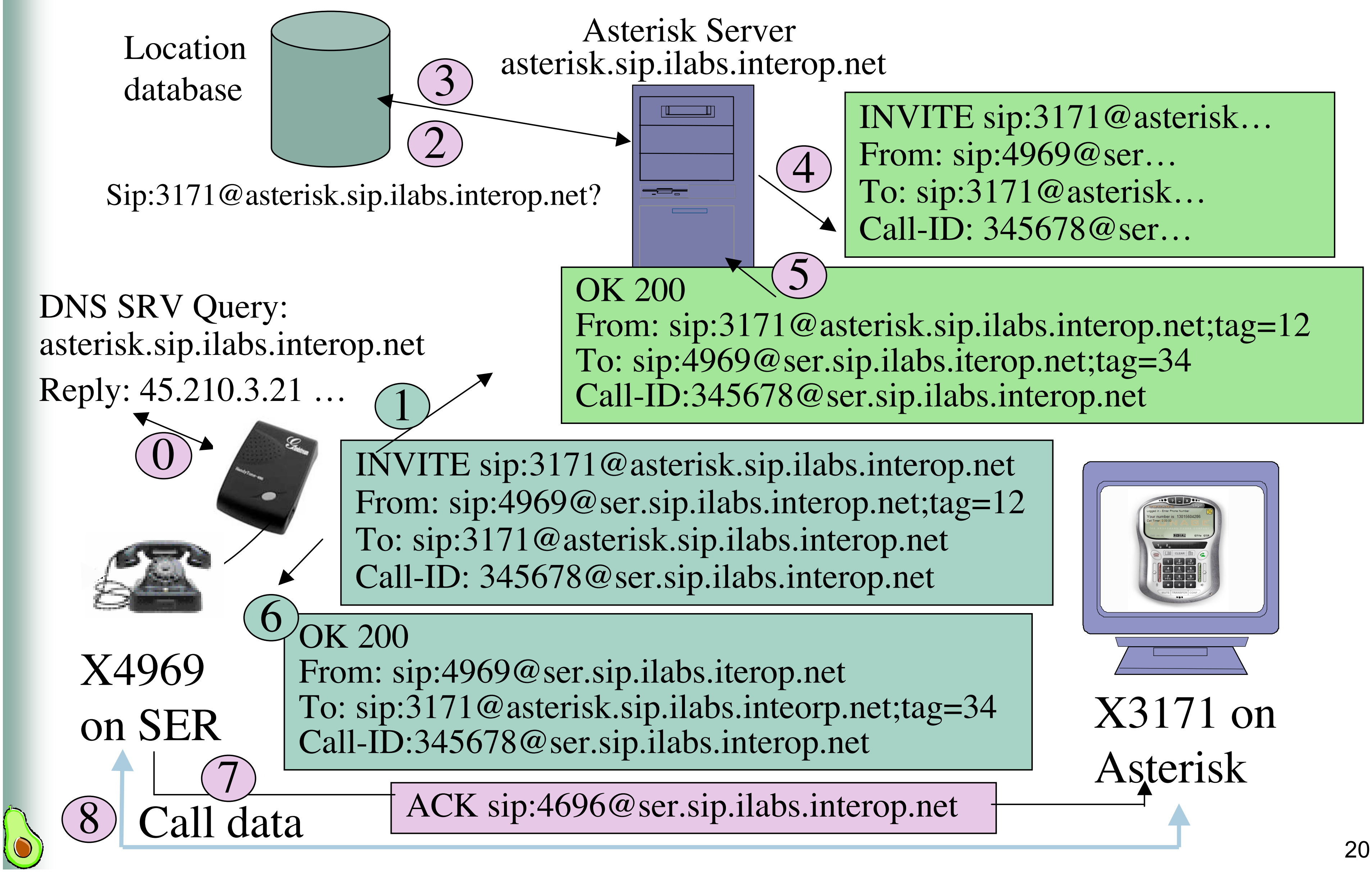
SIP Registrar

REGISTER sip:sip.ilabs.interop.net SIP/2.0
From: sip:crw@sip.ilabs.interop.net
To: sip:crw@sip.ilabs.interop.net
Contact:<sip:45.210.3.78>
Expires: 3600

A **<u>SIP registrar</u>** keeps track of users' whereabouts.
This registration example establishes presence of user with address for one hour and binds this address to user's current location

Registering is not logging in. It is optional. An unregisterd device can still make calls.

SIP Proxy Servers Negotiate Between Endpoints

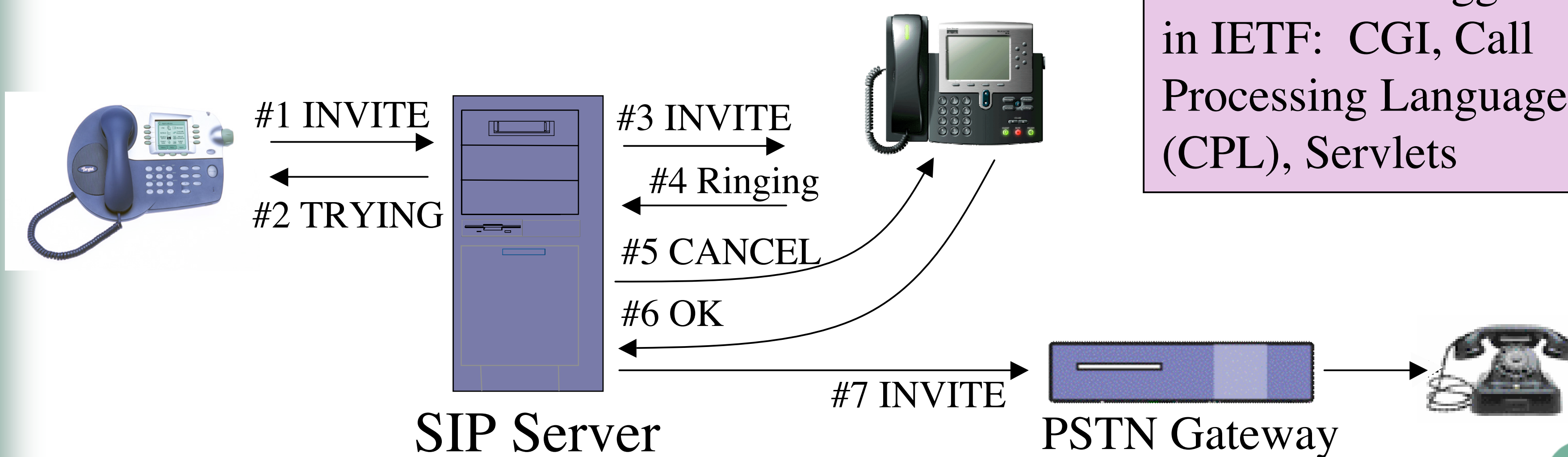


Example of SIP Programmability: Trying Multiple Destinations

A proxy may fork a request to multiple destinations either in parallel ("reach me everywhere") or serially ("forward no reply").

- A proxy can cancel pending parallel searches after a successful response is received.
- A proxy can iterate through redirection responses ("recursive forking").
- The first "OK" is taken.

Programming mechanisms suggested in IETF: CGI, Call Processing Language (CPL), Servlets



SIP Messages are like HTTP

Request

INVITE sip:UserB@there.com SIP/2.0

Via: SIP/2.0/UDP here.com:5060
From: AG <sip:UserA@here.com>;tag=123
To: BG <sip:UserB@there.com>
Call-ID: 12345600@here.com
Cseq: 1 INVITE
Contact: AG <sip:User!@here.com>
Content-Type: application/sdp
Content-Length: 147

Message Header

V=0
O=UserA 28908 28908 IN IP4 here.com
S=Session SDP
C=IN IP4 100.101.102.103
T=0 0
M=audio 49 172 RTP/AVP 0
A=rtpmap:0 PCMU/8000

Payload

Response

SIP/2.0 200 OK

Via: SIP/2.0/UDP here.com:5060
From: AG <sip:UserA@here.com>;tag=123
To: BG <sip:UserB@there.com>;tag=65a35
Call-ID: 12345600@here.com
Cseq: 1 INVITE
Contact: BG <sip:UserB@here.com>
Content-Type: application/sdp
Content-Length: 134

V=0
O=UserB 28908 28908 IN IP4 there.com
S=Session SDP
C=IN IP4 110.111.112.113
T=0 0
M=audio 3456 172 RTP/AVP 0
A=rtpmap:0 PCMU/8000

SDP (RFC2327): “receive RTP G.711-encoded audio at 100.101.102.103:49172

SIP Commands are Called Methods

SIP Method	Description
INVITE	Invites a user to a call
ACK	Used to facilitate reliable message exchange for INVITEs
BYE	Terminates a connection between users or declines a call
CANCEL	Terminates a request, or search, for a user
OPTIONS	Solicits information about a server's capabilities
REGISTER	Registers a user's current location
INFO	Used for mid-session signaling

SIP Extension	Description
SUBSCRIBE	instant messaging and presence
NOTIFY	(RFC3265, RFC3428)
MESSAGE	
REFER	call transfer (RFC3515)
PRACK	provisional reliable responses acknowledgement (RFC3262)
INFO	mid-call signaling (RFC 2976)

SIP Response Codes are Familiar

- Borrowed from HTTP: xyz explanatory text
- Receivers need to understand response class (“x”)
- x80 and higher codes avoid conflicts with future http response codes

1yz Informational
- 100 Trying
- 180 Ringing (ringing tone played locally)
- 181 Call is Being Forwarded

2yz Success
- 200 ok

3yz Redirection
- 300 Multiple Choices
- 301 Moved Permanently
- 302 Moved Temporarily

4yz Client error
- 400 Bad Request
- 401 Unauthorized
- 482 Loop Detected
- 486 Busy Here

5yz Server failure
- 500 Server Internal Error

6yz Global Failure
- 600 Busy Everywhere

Summary of SIP Properties

- Textual (HTTP-like) client-server protocol
  - – Easy to debug, extend and process with textual operating systems
- End-to-end
  - It puts most intelligence into end-devices (“user agents”) – good for scalability and extensibility
  - The network infrastructure designed to be lightweight. Network functionality (registrar, proxy) are typically logical parts of a single server.
- Internet addressing using URIs
  - – E.g., sip:crw@transcendental.com
  - – Non-SIP URIs possible to (e.g., they may be used to redirect a caller to webpage)
  - – Address space unlimited and may be used to create services (sip:sales@hotel.xy; geo.position:=48.54_-123.84_120)
- It delivers mobility: User can register from one or more locations with IP connectivity

What SIP is Not

- Transport
- QoS Reservation Protocol
- Gateway Control Protocol
- Some argue it may be used for accessing IP-enabled appliances ...
- It does NOT dictate ...
  - Product features and services (color of your phone and distinctive ringing melodies, number of simultaneous calls your phone can handle, don't disturb feature, ...)
  - Network configuration

What are Enterprise Features?

- Things you can do with a traditional PBX and a proprietary phone with lots of extra buttons on it:
  - **Call forward** (set your phone to always forward to another; many variants such as call forward on busy, etc)
  - **Call waiting** (be notified when another call comes in, be able to pick up and switch between two calls)
  - **Message Waiting Indicator** (implies voicemail)
  - **DTMF** (touch tones to drive automated menu systems)
  - **Call transfer, attended** (you initiate transfer, talk privately with new party, then connect the two and leave the call)
  - **Call transfer, blind** (you initiate transfer, hang up)
  - **Hold/resume with music**

See the Internet Draft “Session Initiation Protocol Service Examples”, at
`http://www.ietf.org/internet-drafts/draft-ietf-sipping-service-examples-08.txt`

Enterprise Feature Interoperability is Rough

Not a single **feature**... is 100% **supported**:

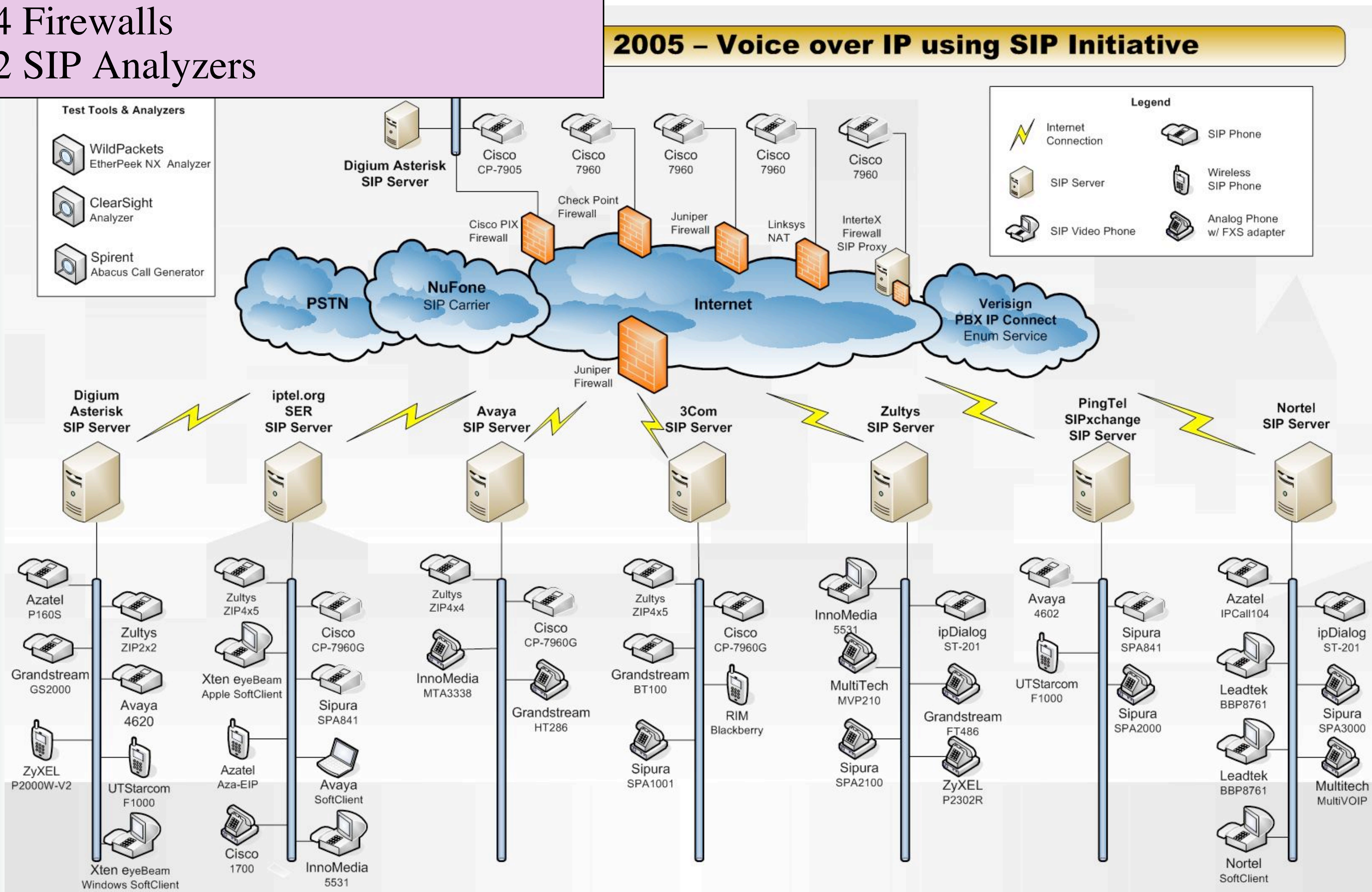
Call forward	Not all phones support it
Call waiting	Not all phones support it
Message Waiting Indicator	Old and new methods within SIP
DTMF	Not all phones support it, several methods
Call transfer, blind	Not all phones support it, AND high failure rates
Call transfer, attended	Not all phones support it, AND high failure rates
Hold/resume with music	Not all phones support it; many SIP proxies don't provide music on hold

Getting Started With SIP

- Get a Phone
- Use Someone Else's SIP Server (Free World Dialup)
- Learn to debug SIP
- Mess with NAT
- Set up your own SIP Proxy Server
- Add enterprise features (call waiting, conference, etc)
- Connect to the PSTN (Need FXS and FXO hardware)
- Write a dial plan
- Figure out how billing is going to work (CDR records)
- You're now ready to evaluate commercial products

What's In the iLabs

8 SIP Servers from 8 vendors
46 phones from 15 different vendors
1 PSTN gateway
4 Firewalls
2 SIP Analyzers



Come to the Ilabs …
See What's on the…

S
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A
B

What We Learned in the iLabs

- SIP is naturally interoperable
  - SIP is easy to debug, and basic call interoperability is good
- All phones are not created equal
  - Tremendous differences between phones in technical factors such as voice quality, performance, handling of jitter, configurability etc, as well as human factors such as how it feels and sounds
- Getting started is easy
  - We had phones working by lunch of the first day and many of them worked the first time
- Enterprise features are tough
  - What’s easy for a PBX must be reinvented with SIP ... it’s a moving target
  - For features you care about, test, test, test!

What You Can Do in the iLabs

- Call from any phone to any other phone - play with different phones! Play with enterprise features!!
- Call to the PSTN - call your office and talk to someone
- Use three different video phones
- Use three different WiFi phones
- Use a protocol analyzer to watch and listen to SIP traffic
- Make Elvis dance!
- Find out what your cell phone number spells

How to Learn More

- **Come see** the iLabs SIP demo and ask questions
- **Pick up** these white papers in the iLabs or electronically:
  - What is SIP?
  - What is ENUM?
  - Getting Started with SIP
  - Migrating to SIP
  - SIP, Firewalls and Security
  - SIP and the PSTN
  - SIP Resources
- http://www.ilabs.interop.net
  - iLabs white papers, vendor white papers, Network World articles, diagrams, team bios and vendor links, this presentation
- http://www.opus1.com/sip
  - iLabs white papers, this presentation and layout diagram (up forever)

Thanks from the Voice Over IP Using SIP Team

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