Internet Telephony based on SIP

SMU - Dallas

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Internet Multimedia

- Real Time Protocol (RTP) – media packets
- Real Time Control Protocol (RTCP) – monitor & report
- Session Announcement Protocol (SAP)
- Session Description Protocol (SDP)
- Session Initiation Protocol (SIP)
- Real Time Stream Protocol (RTSP) – play out control
- Synchronized Multimedia Integration Language (SMIL) – mixes audio/video with text and graphics

References: Search keyword at http://www.rfc-editor.org/rfc.html

For SMIL - http://www.w3.org/AudioVideo/

Telephony on the Internet

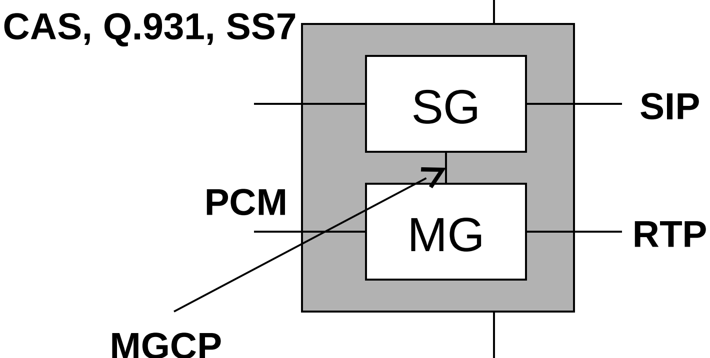
may not be a stand-alone business, but part of IP services

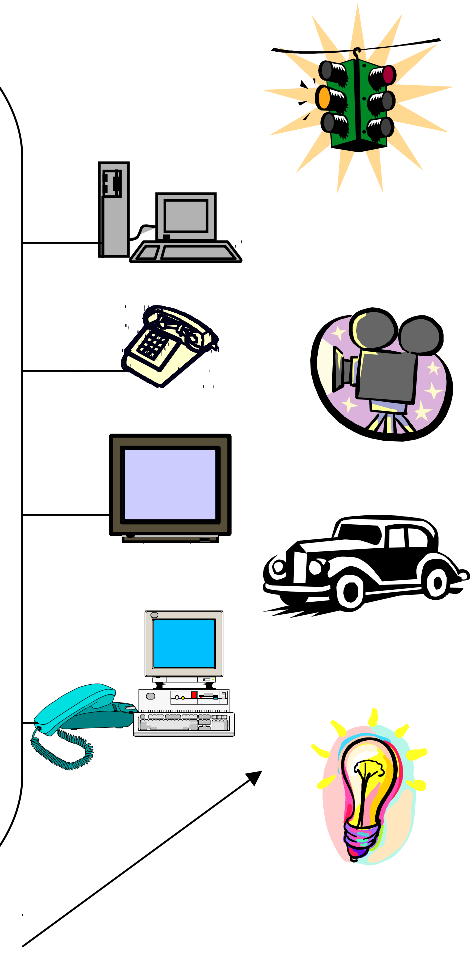
SIP/RTP Media Architecture

Public IP Backbone

- Goes everywhere
- End-to-end control
- Consistent for all services
  - DNS – mobility
  - Messaging
  - Web
  - Directory
  - Security
  - QoS
  - Media services
  - Sessions
  - Telephony
  - …………

Telephone
Gateway
SIP client





Any other sessions

Commercial Grade IP Telephony

Assure baseline PSTN features

Leverage and Commonality of telephony with the Web/Internet

New services (new revenue)

Scalability (Web-like)

Baseline PSTN&PBX features

- Client & user authentication
- Accounting assured QoS
- QoS assured signaling
- Security assured signaling
- Hiding of caller ID & location

Better than PSTN features

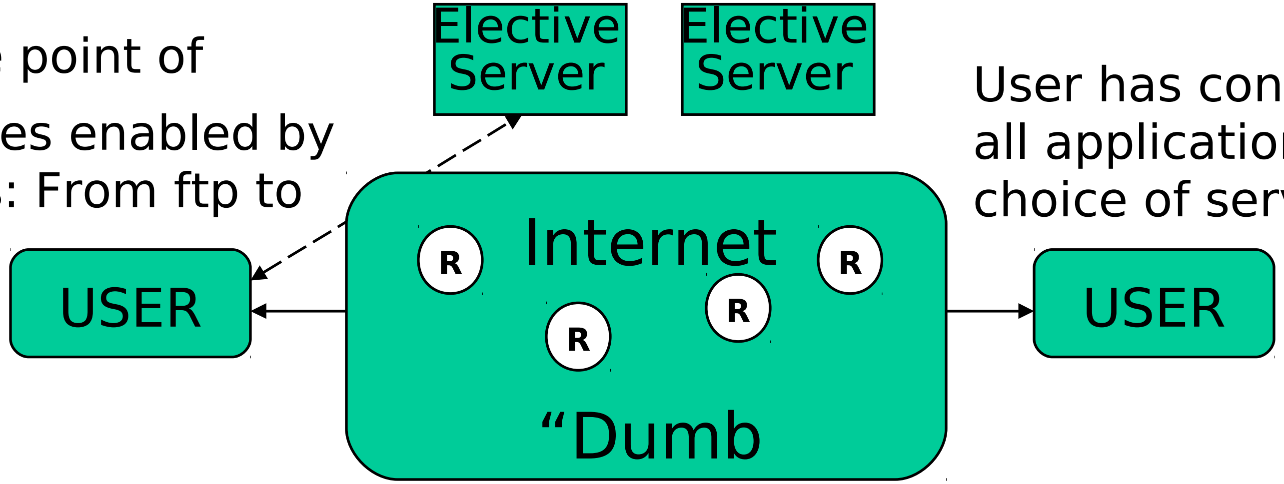
- New & fast service creation
- Internet (rapid) scalability
- Mobility
- Dynamic user preferences
- End-to-end control
  - Service selection
  - Feature control
- Mid-call control features
  - Pre-call
  - Mid-call

Internet End-to-End Control

No single point of failure

All services enabled by protocols: From ftp to web

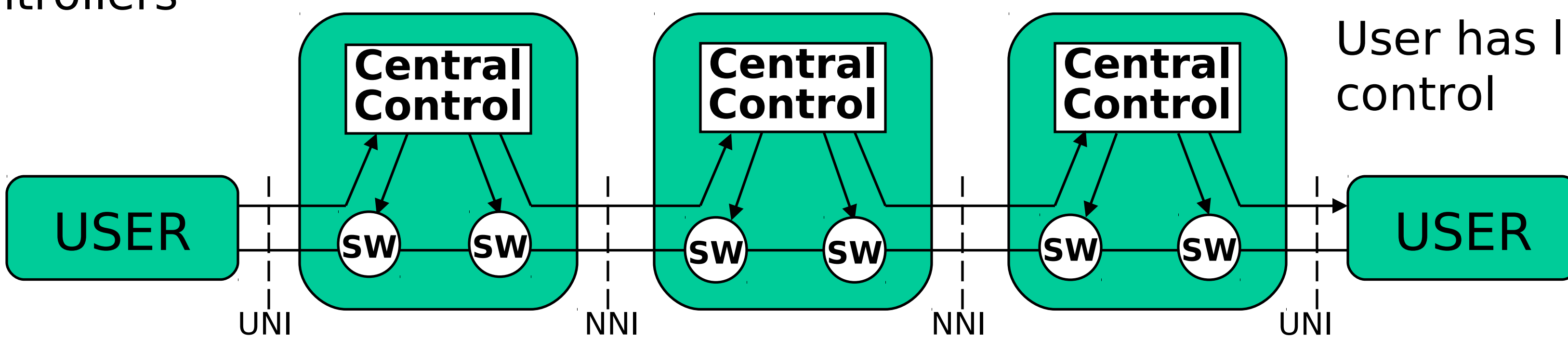
User has control of all applications and choice of servers



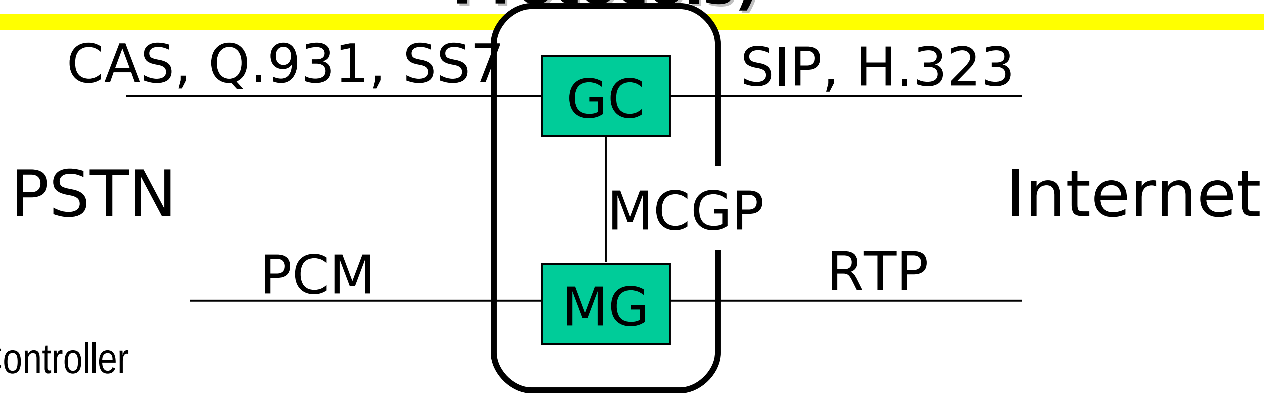
Services supported by interfaces and central controllers

ITN Intelligent Network Control:
POTS, ISDN, BISDN, FR, ATM, H.323, MEGACO/H.248, GSM

User has little control

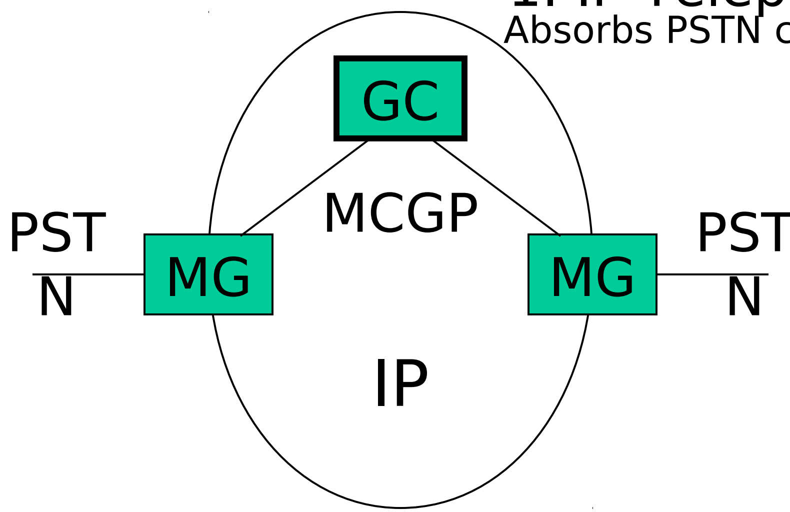


SIP vs. flavors of IPDC, SGSP, MGCP, MEGACO, H.248 (Internet Client-Server vs. Telco Master-Slave Protocols)

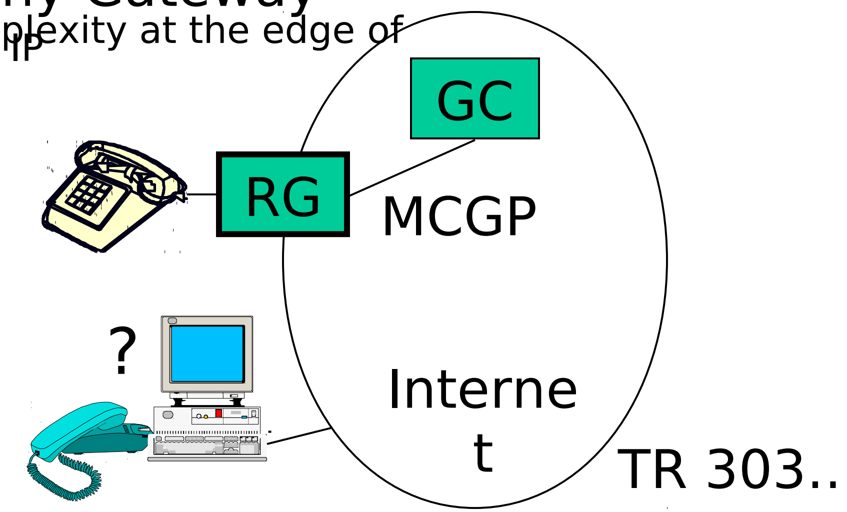


Legend
CG: gateway Controller
MG: Media Gateway

1. IP Telephony Gateway
Absorbs PSTN complexity at the edge of IP



2. "Softswitch" a la IN
- phone to phone only
- PSTN services
- single vendor solution



3. Residential GWY
- breaks e-2-e control model
- no services integration
- no choice of server and apps
- "unequal access" is reinvented

IP Communications

Complete integration of all services under full user control

PSTN/PBX-like:

- POTS
- AIN CS-1, CS-2
- PBX & Centrex

User has control of:

- All addressable devices
- Caller and called party preferences

Better quality than 3.1 kHz

Web-like:

- Presence
- Voice and text chat
- Messaging
- Voice, data, video
- Multiparty
  - Conferencing
  - Education
  - Games

Any quality

Most yet to be invented

Mixt Internet-PSTN: Click'nConnect, ICW, unified messaging

Development of SIP

- IETF - Internet Engineering Task Force
  - MMUSIC - Multiparty Multimedia Session Control Working Group
  - SIP developed by Handley, Schulzrinne, Schooler, and Rosenberg
    - Submitted as Internet-Draft 7/97
  - Assigned RFC 2543 in 3/99
  - Internet Multimedia Conferencing Architecture.
- Alternative to ITU's H.323
  - H.323 used for IP Telephony since 1994
  - Problems: No new services, addressing, features
  - Concerns: scalability, extensibility

SIP Philosophy

- Internet Standard
  - IETF - http://www.ietf.org
- Reuse Internet addressing (URLs, DNS, proxies)
  - Utilizes rich Internet feature set
- Reuse HTTP coding
  - Text based
- Makes no assumptions about underlying protocol:
  - TCP, UDP, X.25, frame, ATM, etc.
  - Support of multicast

SIP Clients and Servers - 1

- SIP uses client/server architecture
- Elements:
  - SIP User Agents (SIP Phones)
  - SIP Servers (Proxy or Redirect - used to locate SIP users or to forward messages.)
    - Can be stateless or stateful
  - SIP Gateways:
    - To PSTN for telephony interworking
    - To H.323 for IP Telephony interworking
- Client - originates message
- Server - responds to or forwards message

SIP Clients and Servers - 2

Logical SIP entities are:

- User Agents
  - User Agent Client (UAC): Initiates SIP requests
  - User Agent Server (UAS): Returns SIP responses
- Network Servers
  - Registrar: Accepts REGISTER requests from clients
  - Proxy: Decides next hop and forwards request
  - Redirect: Sends address of next hop back to client

The different network server types may be collocated

SIP Addressing

Uses Internet URLs

- Uniform Resource Locators
- Supports both Internet and PSTN addresses
- General form is **name@domain**
- To complete a call, needs to be resolved down to User@Host
- Examples:

  sip:alan@wcom.com

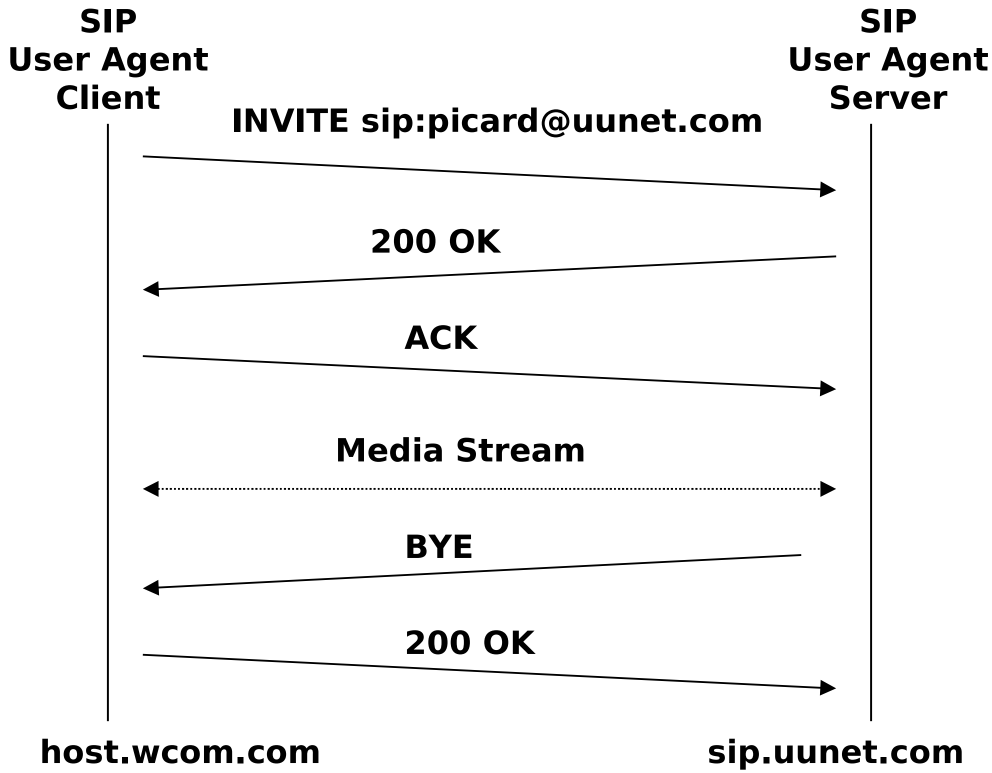
  sip:J.T. Kirk <kirk@starfleet.gov>

  sip:+1-613-555-1212@wcom.com;user=phone

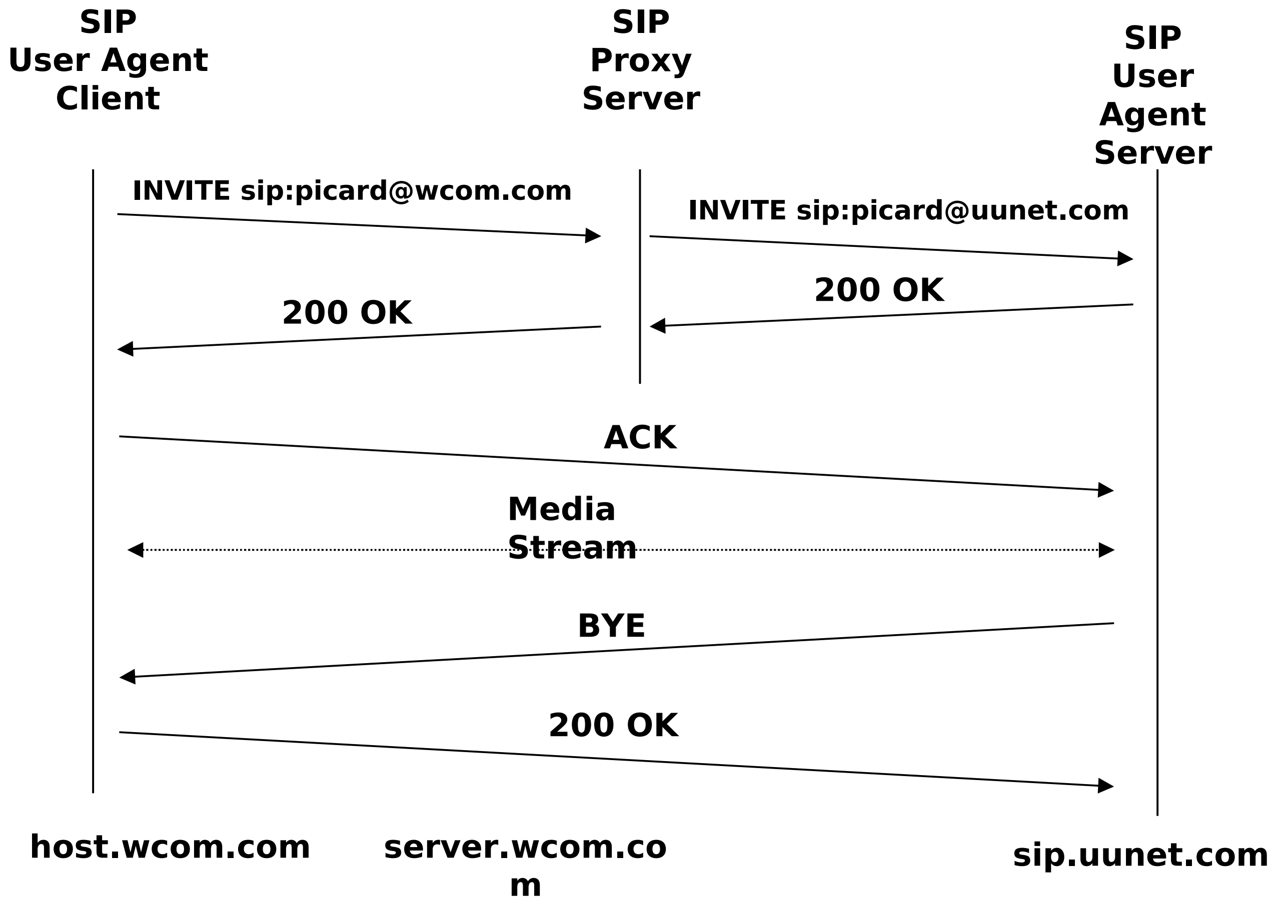
  sip:guest@10.64.1.1

  sip:790-7360@wcom.com;phone-context=VNET

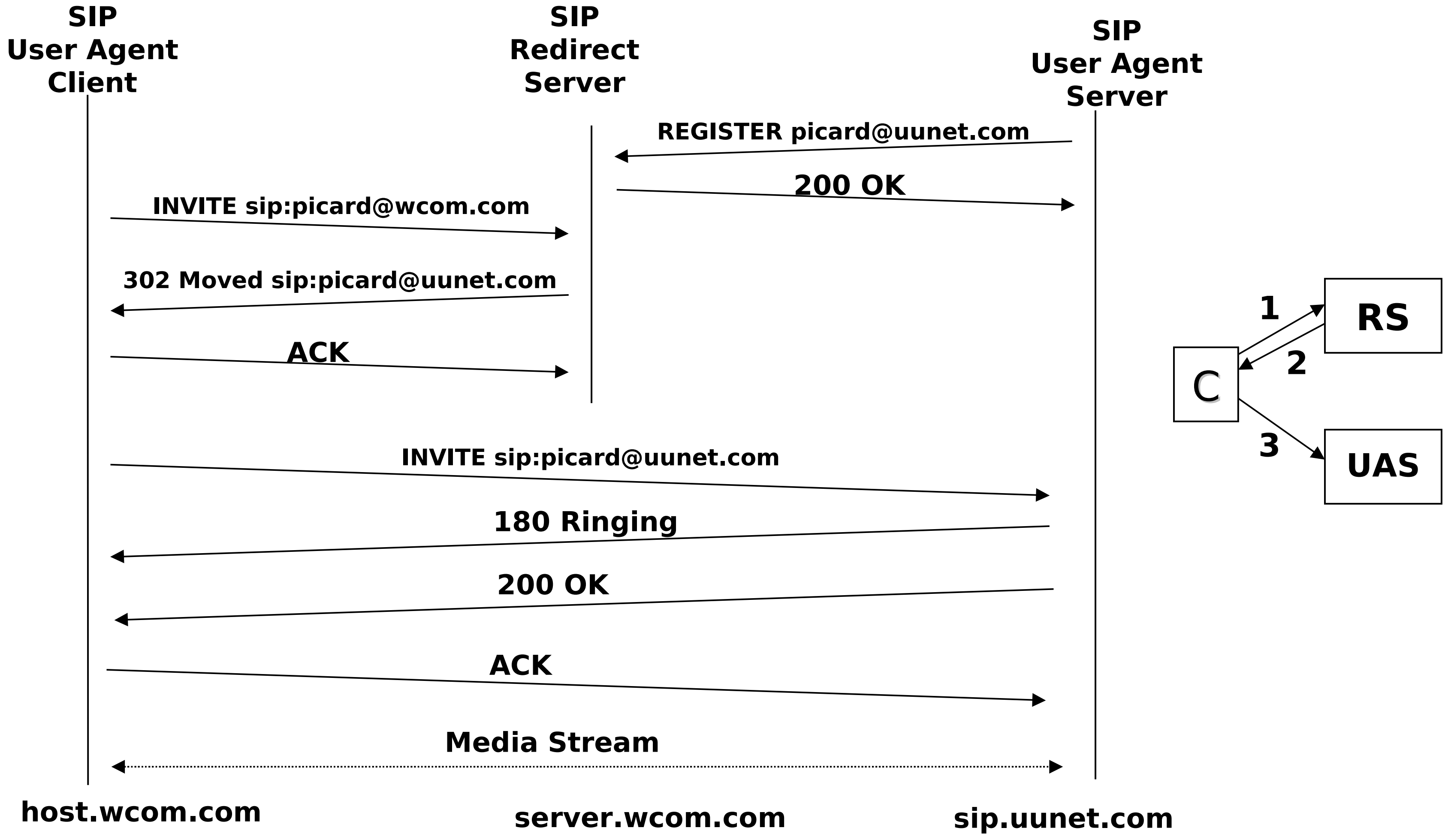
SIP Session Setup Example



Proxy Server Example



Redirect Server Example



SIP Requests

SIP Requests (Messages) defined as:

- Method SP Request-URI SP SIP-Version CRLF (SP=Space, CRLF=Carriage Return and Line Feed)
- Example: INVITE sip:picard@wcom.com SIP/2.0

Method	Description
INVITE	**A session is being requested to be setup using a specified media**
ACK	**Message from client to indicate that a successful response to an INVITE has been received**
OPTIONS	**A Query to a server about its capabilities**
BYE	**A call is being released by either party**
CANCEL	**Cancels any pending requests. Usually sent to a Proxy Server to cancel searches**
REGISTER	**Used by client to register a particular address with the SIP server**

SIP Requests Example

Required Headers (fields):

```
INVITE sip:picard@wcom.com SIP/2.0
Via: SIP/2.0/UDP host.wcom.com:5060
From: Alan Johnston <sip:alan.johnston@wcom.com>
To: Jean Luc Picard <sip:picard@wcom.com>
Call-ID: 314159@host.wcom.com
CSeq: 1 INVITE
```

Uniquely identify this session request

- **`Via`**: Shows route taken by request.
- **`Call-ID`**: unique identifier generated by client.
- **`CSeq`**: Command Sequence number
  - generated by client
  - Incremented for each successive request

SIP Requests Example

Typical SIP Request:

```
INVITE sip:picard@wcom.com SIP/2.0
Via: SIP/2.0/UDP host.wcom.com:5060
From: Alan Johnston <sip:alan.johnston@wcom.com>
To: Jean Luc Picard <sip:picard@wcom.com>
Call-ID: 314159@host.wcom.com
CSeq: 1 INVITE
Contact: sip:alan.johnston@wcom.com
Subject: Where are you these days?
Content-Type: application/sdp
Content-Length: 124

v=0
o=ajohnston 5462346 332134 IN IP4 host.wcom.com
s=Let's Talk
t=0 0
c=IN IP4 10.64.1.1
m=audio 49170 RTP/AVP 0 3
```

SIP Responses

SIP Responses defined as (HTTP-style):

- SIP-Version SP Status-Code SP Reason-Phrase CRLF
  (SP=Space, CRLF=Carriage Return and Line Feed)
- Example: **`SIP/2.0 404 Not Found`**
- First digit gives Class of response:

	Description	Examples
1xx	Informational – Request received, continuing to process request.	180 Ringing 181 Call is Being Forwarded
2xx	Success – Action was successfully received, understood and accepted.	200 OK
3xx	Redirection – Further action needs to be taken in order to complete the request.	300 Multiple Choices 302 Moved Temporarily
4xx	Client Error – Request contains bad syntax or cannot be fulfilled at this server.	401 Unauthorized 408 Request Timeout
5xx	Server Error – Server failed to fulfill an apparently valid request.	503 Service Unavailable 505 Version Not Suported
6xx	Global Failure – Request is invalid at any server.	600 Busy Everywhere 603 Decline

SIP Responses Example

Required Headers:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP host.wcom.com:5060
From: Alan Johnston <sip:alan.johnston@wcom.com>
To: Jean Luc Picard <sip:picard@wcom.com>
Call-ID: 314159@host.wcom.com
CSeq: 1 INVITE
```

- **`Via`**, **`From`**, **`To`**, **`Call-ID`**, and **`CSeq`** are copied exactly from Request.
- **`To`** and **`From`** are NOT swapped!

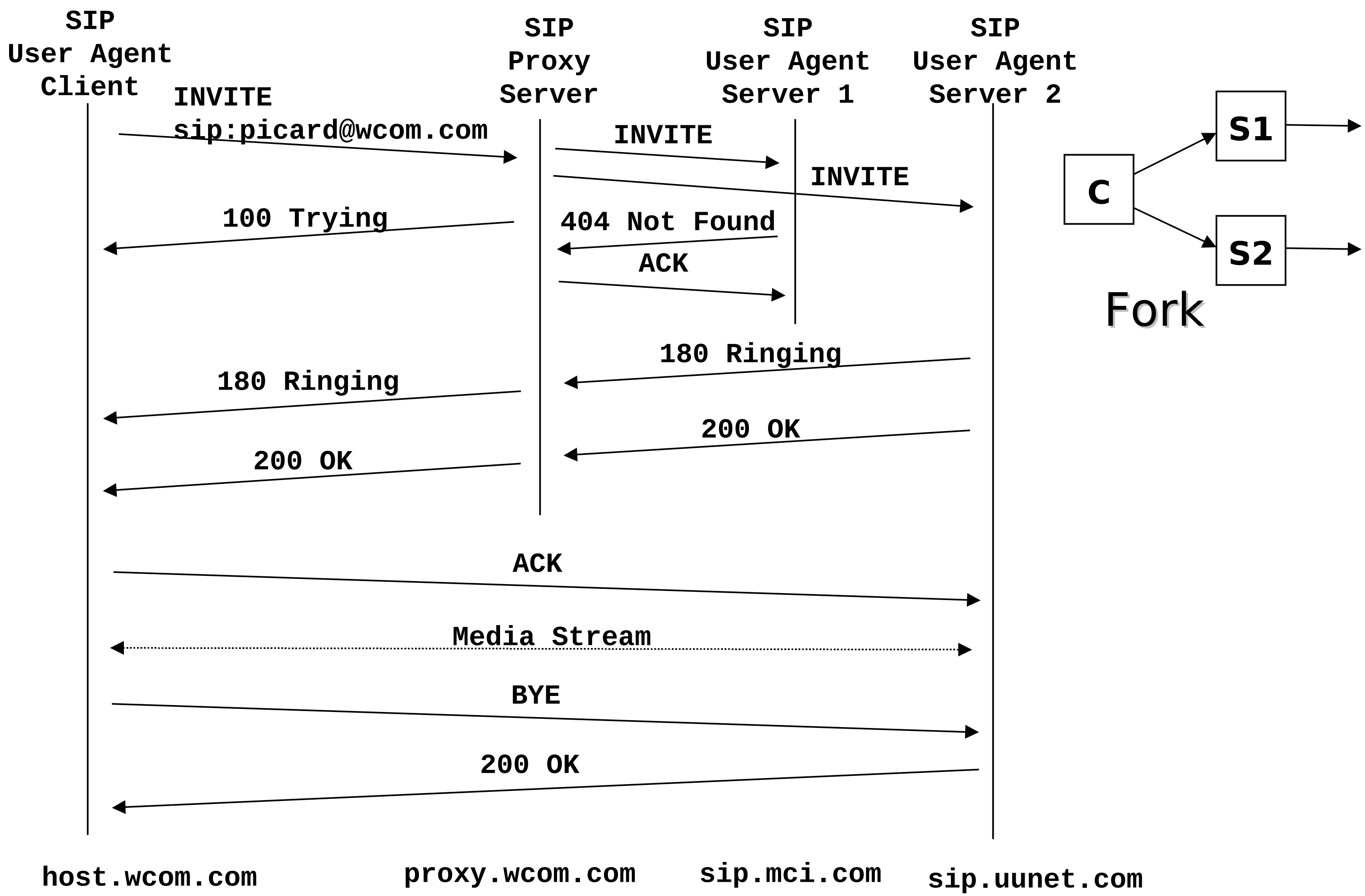
SIP Responses Example

Typical SIP Response (containing SDP)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP host.wcom.com
From: Alan Johnston <sip:alan.johnston@wcom.com>
To: Jean Luc Picard <sip:picard@wcom.com>
Call-ID: 314159@host.wcom.com
CSeq: 1 INVITE
Contact: sip:picard@wcom.com
Subject: Where are you these days?
Content-Type: application/sdp
Content-Length: 107

v=0
o=picard 124333 67895 IN IP4 uunet.com
s=Engage!
t=0 0
c=IN IP4 11.234.2.1
m=audio 3456 RTP/AVP 0
```

Forking Proxy Example



SIP Headers - Partial List

Header	Description	Examples
Accept	Indicates acceptable formats.	Accept: application/sdp Accept: currency/dollars
Authorization	Contains encryption information	Authorization: pgp info…
Call-ID	Used to uniquely identify a particular session or registration messages. Should have randomness to ensure overall global uniqueness.	Call-ID: 1@mars.brooks.net Call-ID: Jan-01-1999-1510- 1@server.mci.com i: 31415926535@uunet.com
Contact	Alternative SIP URL for more direct message routing.	Contact: W. Riker, Acting Captain <riker@starfleet.gov> Contact: room203@hotel.com; expires=3600 m: admin@mci.com
Content-Length	Octet count in message body.	Content-Length: 285
Content-Type	Content type of message body	Content-Type: application/sdp c: application/h.323
CSeq	Command Sequence number – used to distinguish different requests during the same session.	CSeq: 1 INVITE CSeq: 1000 INVITE CSeq: 4325 BYE CSeq: 1 REGISTER
Encryption	Encryption information.	Encryption: pgp info…
Expires	Used to indicate when the message content is no longer valid. Can be a number of seconds or a date and time.	Expires: 60 Expires: Thu, 07 Jan 1999 17:00 CST

SIP Headers - Continued

From	Required field containing the originating SIP URL. Can also include a display name.	From: Dana Scully <sip:dana@skeptics.org> From: sip:+1-314-342-7360@gateway.wcom.com; tag=1234567 f: sip: guest@192.168.1.1
Max-Forwards	Count decremented by each server forwarding the message. When goes to zero, server sends a 483 Too Many Hops response.	Max-Forwards: 10
Priority	Can specify message priority	Priority: normal Priority: emergency
Record-Route	Added to a request by a proxy that needs to be in the path of future messages.	Record Route: sip.mci.com
Require	Indicates options necessary for the session.	Require: local.telephony
Response-Key	Contains PGP key for encrypted response expected.	Response-Key: pgp info…
Retry-After	Indicates when the resource may be available. Can be a number of seconds or a date and time.	Retry-After: 3600 Retry-After: Sat, 01 Jan 2000 00:01 GMT

SIP Headers - Continued

Route	**Determines the route taken by a message.**	**Route: orinoco.brooks.net**
Subject	**Can be used to indicate nature of call.**	**Subject: More about SIP** **s: You'd better answer!**
To	**Required field containing the recipient SIP URL. May contain a display name.**	**To: Fox Mulder <sip:mulder@lonegunman.org>** **To: sip:10109000@operator.mci.com; tag=314** **t: sip:1800COLLECT@telecom.mci.com; tag=52**
Unsupported	**Lists features not supported by server.**	**Unsupported: tcap.telephony**
Via	**Used to show the path taken by the request.**	**Via: SIP/2.0/UDP sip.mfs.com** **Via: SIP/2.0/TCP uunet.com** **v: SIP/2.0/UDP 192.168.1.1**
Warning	**Contains a code and text to warn about a problem**	**Warning: 331 Unicast not available**

Via Headers and Routing

- `Via` headers are used for routing SIP messages
- Requests
  - Request initiator puts address in `Via` header
  - Servers check `Via` with sender’s address, then add own address, then forward. (if different, add “`received`” parameter)
- Responses
  - Response initiator copies request `Via` headers.
  - Servers check `Via` with own address, then forward to next `Via` address

SIP Firewall Considerations

- Firewall Problem
  - Can block SIP packets
  - Can change IP addresses of packets
- TCP can be used instead of UDP
- Record-Route can be used:
  - ensures Firewall proxy stays in path
- A Firewall proxy adds Record-Route header
  - Clients and Servers copy Record-Route and put in Route header for all messages

SIP Message Body

- Message body can be any protocol
- Most implementations:
  - SDP - Session Description Protocol
  - RFC 2327 4/98 by Handley and Jacobson
    - http://www.ietf.org/rfc/rfc2327.txt
  - Used to specify info about a multi-media session.
  - SDP fields have a required order
  - For RTP - Real Time Protocol Sessions:
    - RTP Audio/Video Profile (RTP/AVP) payload descriptions are often used

SDP Examples

SDP Example 1

v=0
o=ajohnston +1-613-555-1212 IN IP4 host.wcom.com
s=Let's Talk
t=0 0
c=IN IP4 101.64.4.1
m=audio 49170 RTP/AVP 0 3

SDP Example 2

v=0
o=picard 124333 67895 IN IP4 uunet.com
s=Engage!
t=0 0
c=IN IP4 101.234.2.1
m=audio 3456 RTP/AVP 0

Field	Descripton
Version	v=0
Origin	o=<username> <session id> <version> <network type> <address type> <address>
Session Name	s=<session name>
Times	t=<start time> <stop time>
Connection Data	c=<network type> <address type> <connection address>
Media	m=<media> <port> <transport> <media format list>

Another SDP Example

```
v=0
o=alan +1-613-1212 IN host.wcom.com
s=SSE University Seminar - SIP
i=Audio, Listen only
u=http://sse.mcit.com/university/
e=alan@wcom.com
p=+1-329-342-7360
c=IN IP4 10.64.5.246
b=CT:128
t=2876565 2876599
m=audio 3456 RTP/AVP 0 3
a=type:recvonly
```

Authentication & Encryption

- SIP supports a variety of approaches:
  - end to end encryption
  - hop by hop encryption
- Proxies can require authentication:
  - Responds to **`INVITE`**s with **`407 Proxy-Authentication Required`**
  - Client re-**`INVITE`**s with **`Proxy-Authorization`** header.
- SIP Users can require authentication:
  - Responds to **`INVITE`**s with **`401 Unathorized`**
  - Client re-**`INVITE`**s with **`Authorization`** header

SIP Encryption Example

```
INVITE sip:picard@wcom.com SIP/2.0
Via: SIP/2.0/UDP host.wcom.com:5060
From: Alan Johnston <sip:alan@wcom.com>
To: Jean Luc Picard <sip:picard@wcom.com>
Call-ID: 314159@host.wcom.com
CSeq: 1 INVITE
Content-Length: 224
Encryption: PGP version=2.6.2, encoding=ascii

q4aspdoCjh32a1@WoiLuaE6erIgnqD3erDg8aFs8od7idf@
hWjasGdg,ddgg+fdgf_ggEO;ALewAKFeJqAFSeDlkjhasdf
kj!aJsdfasdfKlfghgasdfasdfa|Gsdf>a!
sdasdf3w29451k45mser?
we5y;343.4kfj2ui2S8~&djGO4kP
%Hk#(Khujefjnjmbm.sd;da'l;12';123=]aw;erwAo3529
ofgk
```

PSTN Features with SIP

Features implemented by SIP Phone

- Call answering: **`200 OK`** sent
- Busy: **`483 Busy Here`** sent
- Call rejection: **`603 Declined`** sent
- Caller-ID: present in **`From`** header
- Hold: a re-INVITE is issued with IP Addr =0.0.0.0
- Selective Call Acceptance: using **`From`**, **`Priority`**, and **`Subject`** headers
- Camp On: **`181 Call Queued`** responses are monitored until **`200 OK`** is sent by the called party
- Call Waiting: Receiving alerts during a call

PSTN Features with SIP

Features implemented by SIP Server

- Call Forwarding: server issues **301 Moved Permanently** or **302 Moved Temporarily** response with **Contact** info
- Forward Don't Answer: server issues **408 Request Timeout** response
- Voicemail: server **302 Moved Temporarily** response with **Contact** of Voicemail Server
- Follow Me Service: Use forking proxy to try multiple locations at the same time
- Caller-ID blocking - Privacy: Server encrypts **From** information

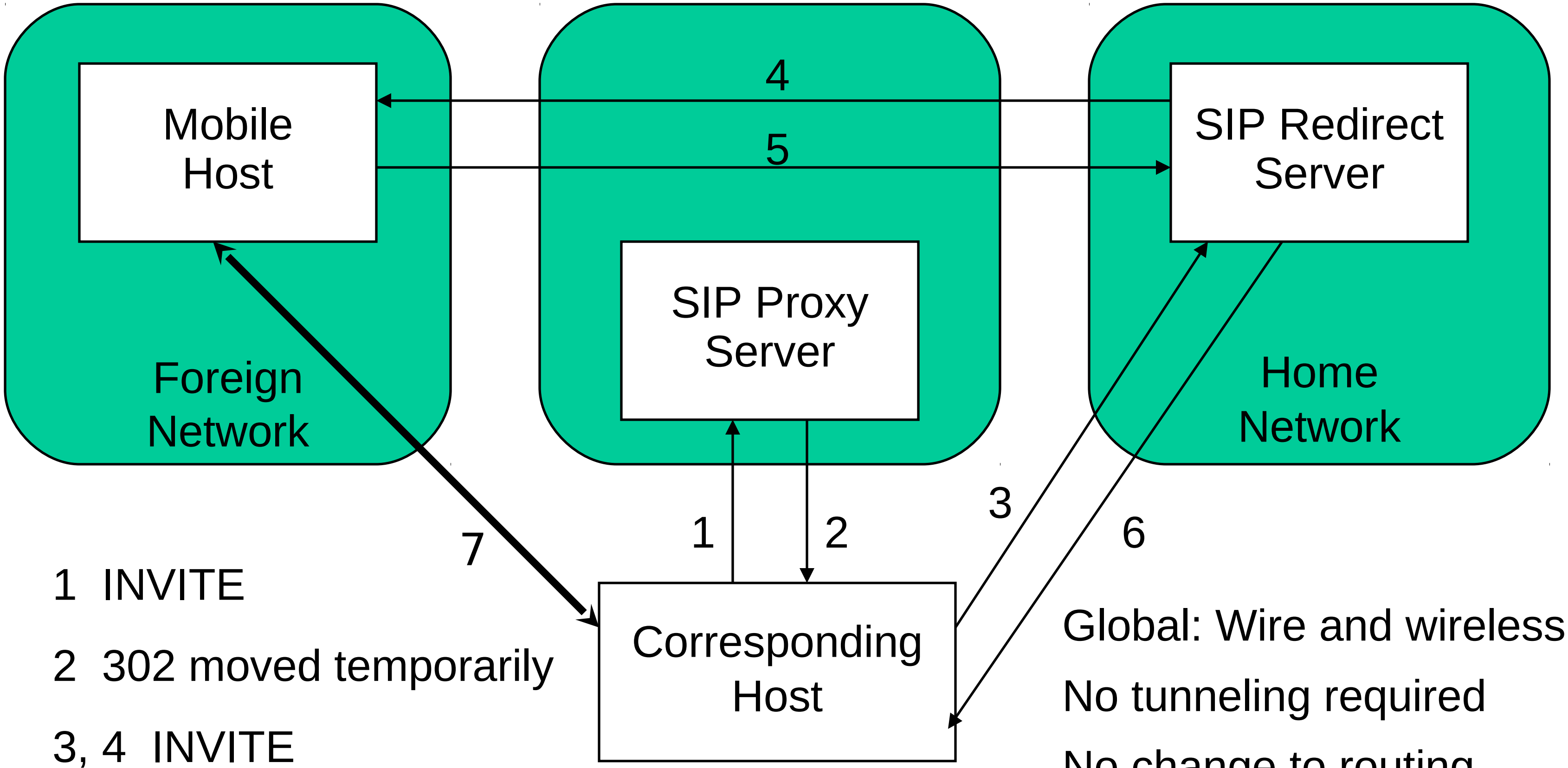
SIP User Location Example

SIP supports mobility across networks and devices

Q=*quality* gives preference

```
SIP/2.0 302 Moved temporarily
Contact: sip:henry@wcom.com
          ;service=IP,voice mail
          ;media=audio ;duplex=full ;q=0.7
Contact : phone: +1-972-555-1212; service=ISDN
          ;mobility=fixed; language=en,es, ;q=0.5
Contact : phone: +1-214-555-1212; service=pager
          ;mobility=mobile
          ;duplex=send-only ;media=text; q=0.1; priority=urgent
          ;description="For emergency only"
Contact : mailto: henry@wcom.com
```

SIP Mobility Support



1 INVITE

2 302 moved temporarily

3, 4 INVITE

5, 6 OK

7 Data

Global: Wire and wireless

No tunneling required

No change to routing

For fast hand-offs use:

- Use Cellular IP or
- Use DRCP

SIP Mobility

Pre-call mobility

- MH can find SIP server via multicast REGISTER
- MH acquires IP address via DHCP
- MH updates home SIP server

Mid-call mobility

- MH->CH: New INVITE with Contact and updated SDP
- Re-registers with home registrar

Need not bother home registrar: Use multi-stage registration

Recovery from disconnects

Mobile IP Communications

Mobile IP Requirements

- Transparency above L2:

Move but keep IP address and all sessions alive

- Mobility
  - Within subnet
  - Within domain
  - Global
- AAA and NAIs
- Location privacy
- QoS for r.t. communications

Evolution of Wireless Mobility

- Circuit Switched Mobility

based on central INs

- LAN-MAN: Cellular IP
- Wide Area: Mobile IP
- Universal (any net): SIP

Presence, Instant Messaging and Voice

http://www.ietf.org/internet-drafts/draft-ietf-impp-model-03.txt

IP SIP Phones and Adaptors

Are Internet hosts

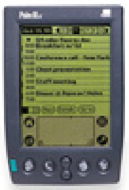
- Choice of application
- Choice of server
- IP appliance

1

Implementations

- 3Com (2)
- Cisco
- Columbia University
- Mediatrix (1)
- Nortel (3)
- Pingtel

2



3

SIP Summary

- SIP is:
  - Relatively easy to implement
  - Gaining vendor and carrier acceptance
  - Very flexible in service creation
  - Extensible and scaleable
  - Appearing in products right now
- SIP is not:
  - Going to make PSTN interworking easy
  - Going to solve all IP Telephony issues (QoS)

References

Book on “Internetworking Multimedia” by Jon Crowcroft, Mark Handley, Ian Wakeman, UCL Press, 1999 by Morgan Kaufman (USA) and Taylor Francis (UK)

RFC 2543: “SIP: Session Initiation Protocol”

ftp://ftp.isi.edu/in-notes/rfc2543.txt

The IETF SIP Working Group home page

http://www.ietf.org/html.charters/sip-charter.html

SIP Home Page

http://www.cs.columbia.edu/~hgs/sip/

Papers on IP Telephony

http://www.cs.columbia.edu/~hgs/sip/papers.html

Relevant IETF Working Groups

http://ietf.org/html.charters/wg-dir.html

- **Audio/Video Transport (avt) - RTP**
- **Differentiated Services (diffserv) – QoS in backbone**
- **IP Telephony (iptel) – CPL, GW location, TRIP**
- **Integrated Services (intserv) – end-to-end QoS**
- **Media Gateway Control (megaco) – IP telephony gateways**
- **Multiparty Multimedia Session Control (mmusic) – SIP, SDP, conferencing**
- **PSTN and Internet Internetworking (pint) – mixt services**
- **Resource Reservation Setup Protocol (rsvp)**
- **Service in the PSTN/IN Requesting InTernet Service (spirits)**
- **Session Initiation Protocol (sip) – signaling for call setup**
- **Signaling Transport (sigtran) – PSTN signaling over IP**
- **Telephone Number Mapping (enum) – surprises !**
- **Instant Messaging and Presence Protocol (impp)**

This large work effort may cause the complete re-engineering of communication systems and services