VoIP

- What's Voice-over IP?
  - Transmission of voice using IP
  - Analog speech digitized and transmitted as IP **packets**
  - Packets transmitted on top of existing networks
  - Voice connection is now packet-switched as compared to the traditional circuit-switched telephony network
    - Interesting implications w.r.t performance, security, regulatory requirements

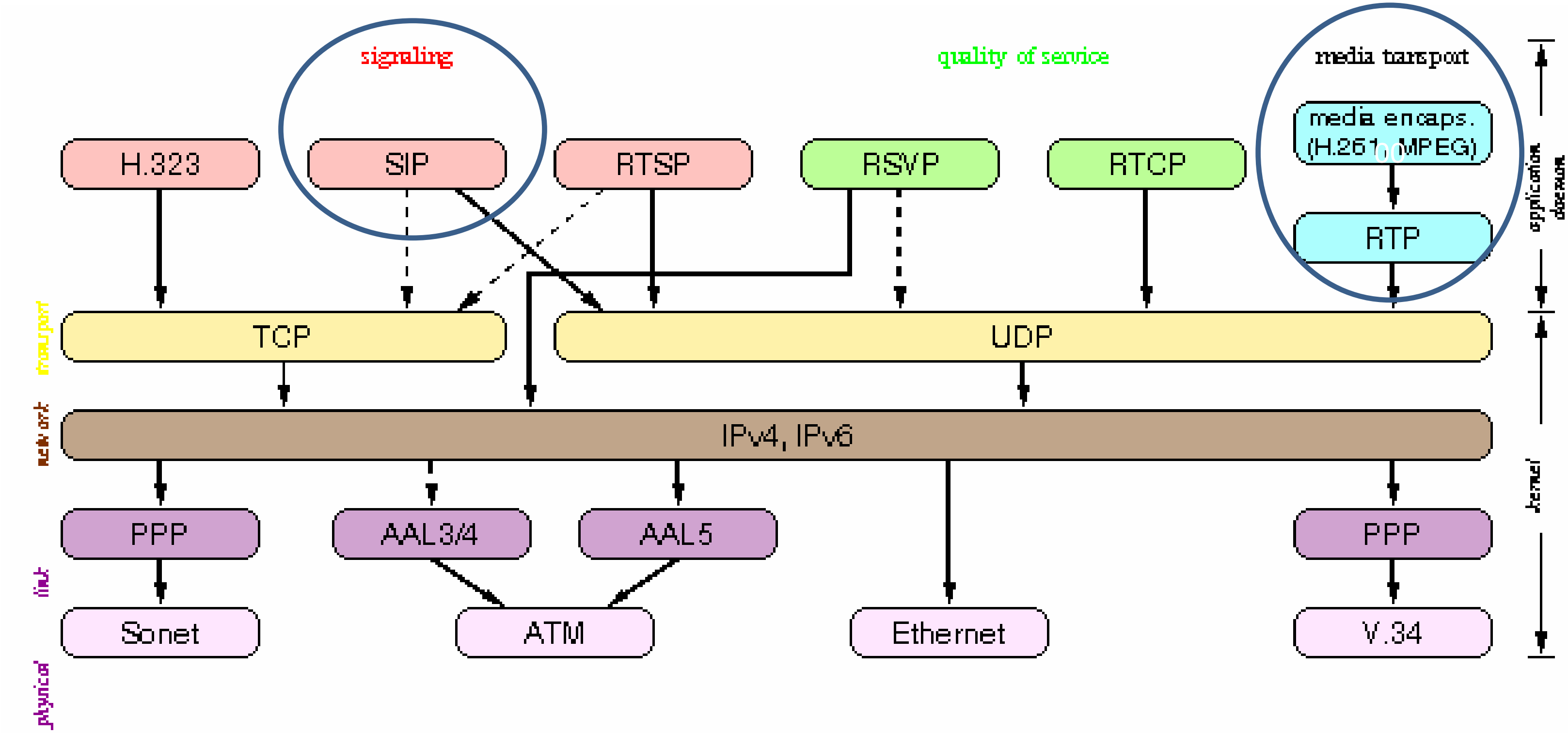
Why VoIP?

- Low cost of ownership
- Simpler maintenance of infrastructure
- Innovative applications
- Software Based
  - Easier to add new applications

VoIP Standard suites

- Internet Engineering Task Force (IETF) standards
  - Session Initiation Protocol (SIP)
  - Session Description Protocol (SDP)
  - Uses RTP, SRTP for media transmission.
  - Others for QoS
- International Telecommunication Union (ITU) standards
  - H.323 Basic Architecture
  - H.225 call control protocol
  - H.245 media control protocol
  - H.235 security
  - Other standards related to codecs.
- Proprietary (Skype)

VoIP Standards Suite



Elements of IETF Standards Suite

- Session Initiation Protocol (SIP)
  - Session establishment / teardown (HTTP-like)
- Session Description Protocol (SDP)
  - Describes the parameters of session (voice/audio)
- RTP (Real-Time Transport Protocol)
  - Actual media data packets
  - Secure variant (SRTP)
- RTP Control Protocol
  - Provides feedback on QoS provided by RTP

Session Initiation Protocol (SIP)

- SIP is the signaling protocol
  - Establishes sessions between two parties
- Major functionality
  - User Location
  - User Availability
  - Capabilities
  - Session Setup
  - Session Teardown

SIP Architectural Entities

- SIP Registrar/Location Server
  - Registers users location
- SIP User Agent Client
  - Acts on behalf of the SIP user initiating the call
- SIP User Agent Server
  - Waits on behalf of the SIP user to receive calls
- SIP Proxy
  - Acts as a proxy for the client (within a given domain)
- SIP Gateways:

  To PSTN for telephony interworking

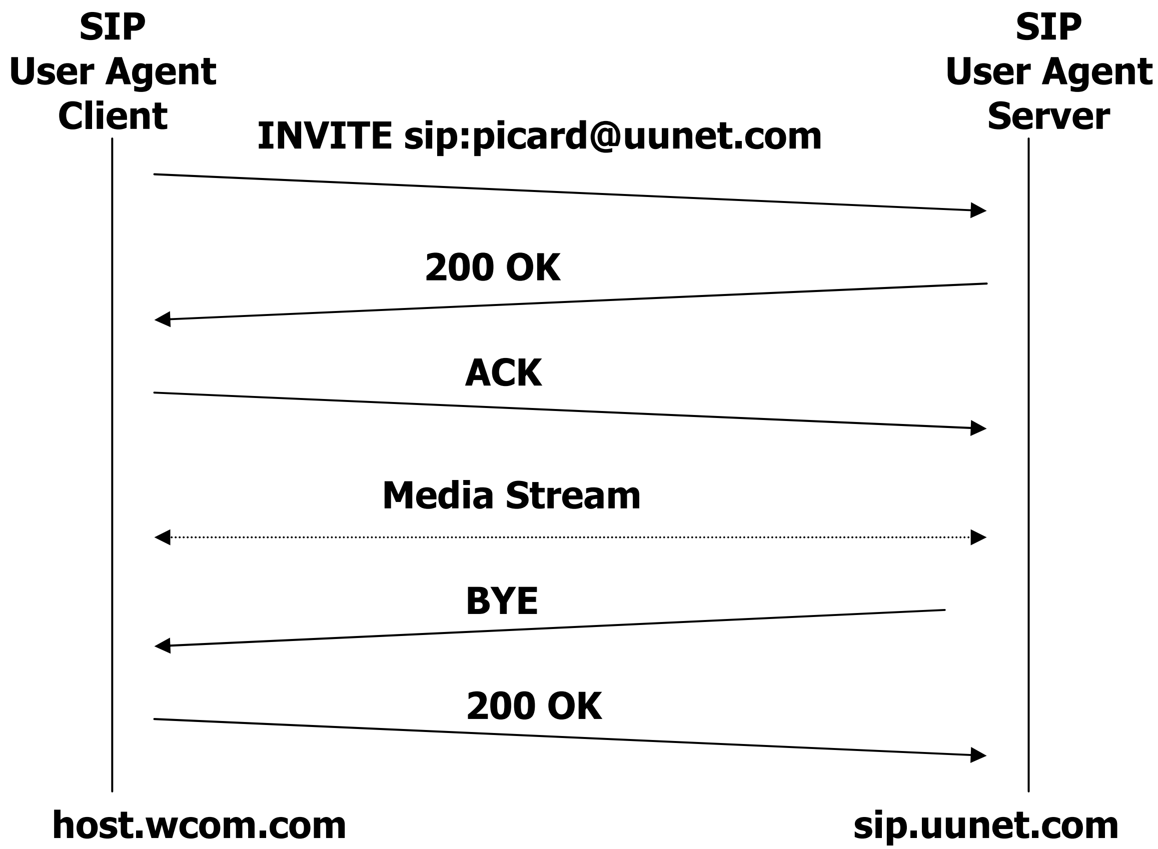
  To H.323 for IP Telephony interworking

"It is an important concept that the distinction between types of SIP servers is logical, not physical"

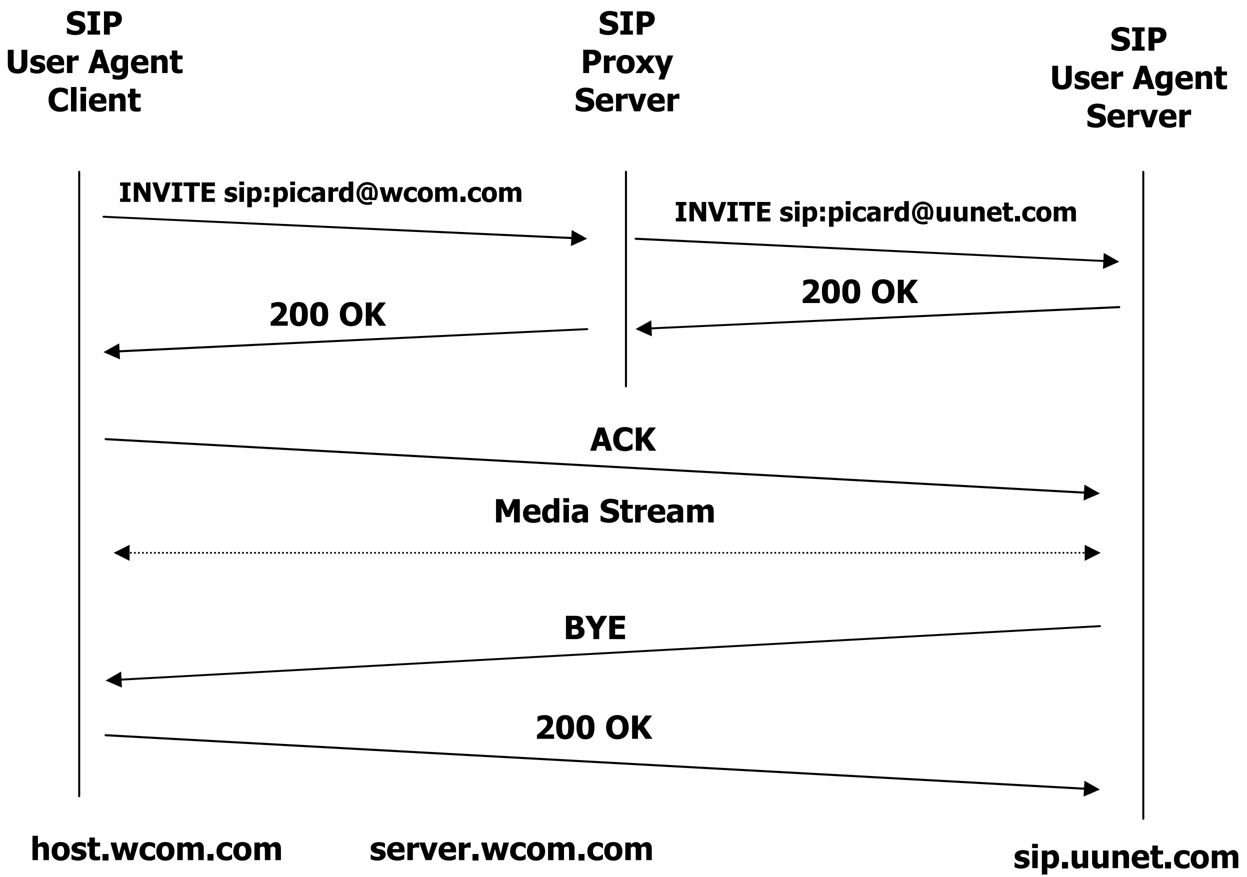
Why the need for so many elements?

- Usage of a logical SIP address and location servers allows the machine to be untied from the identity of the user
- Having a SIP proxy allows
  - Client devices to be simple (easily implemented in hardware)
  - Additional upgrades can be done in the Proxy (transparent to the SIP device)
  - More sophisticated functionality

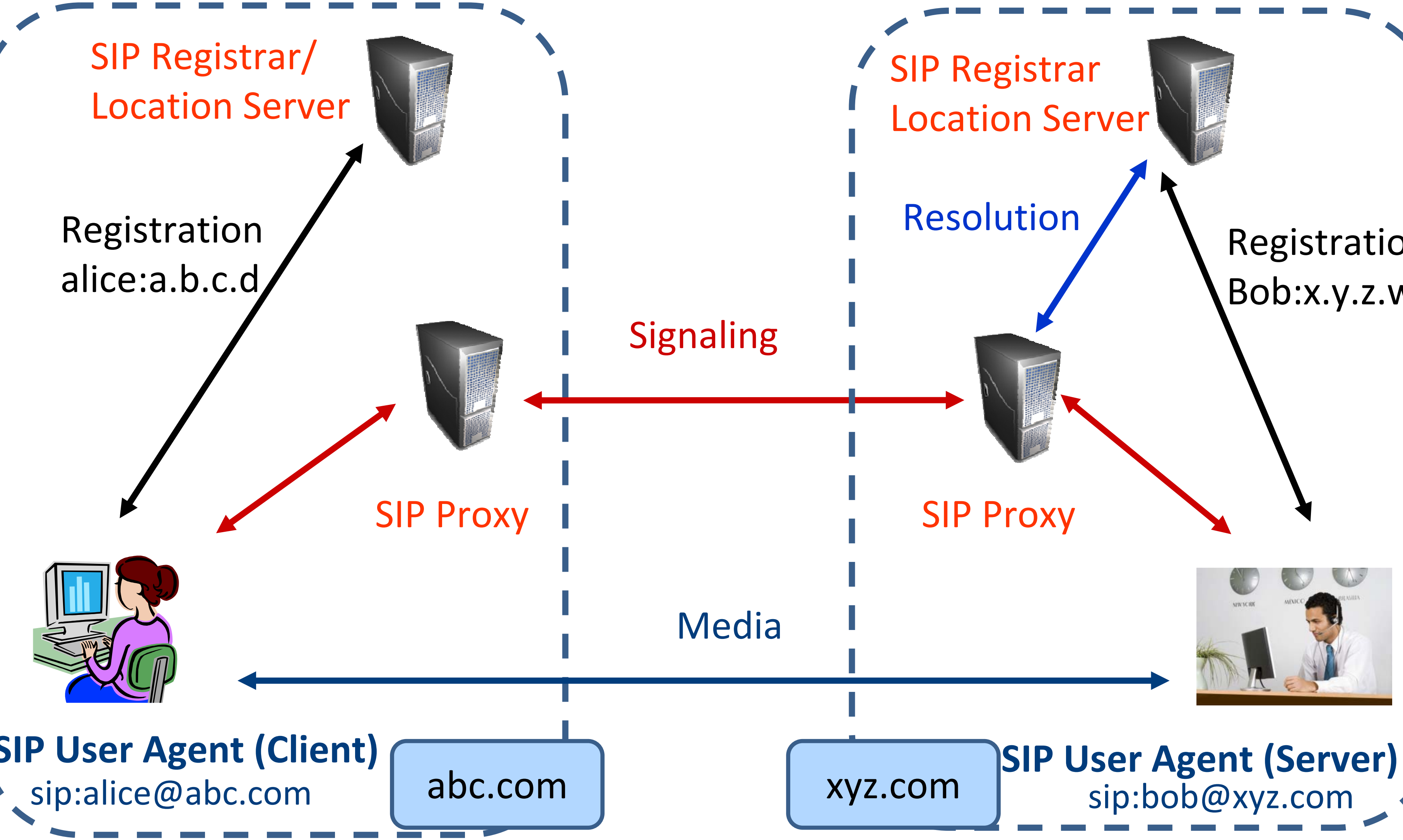
SIP Session Setup Example



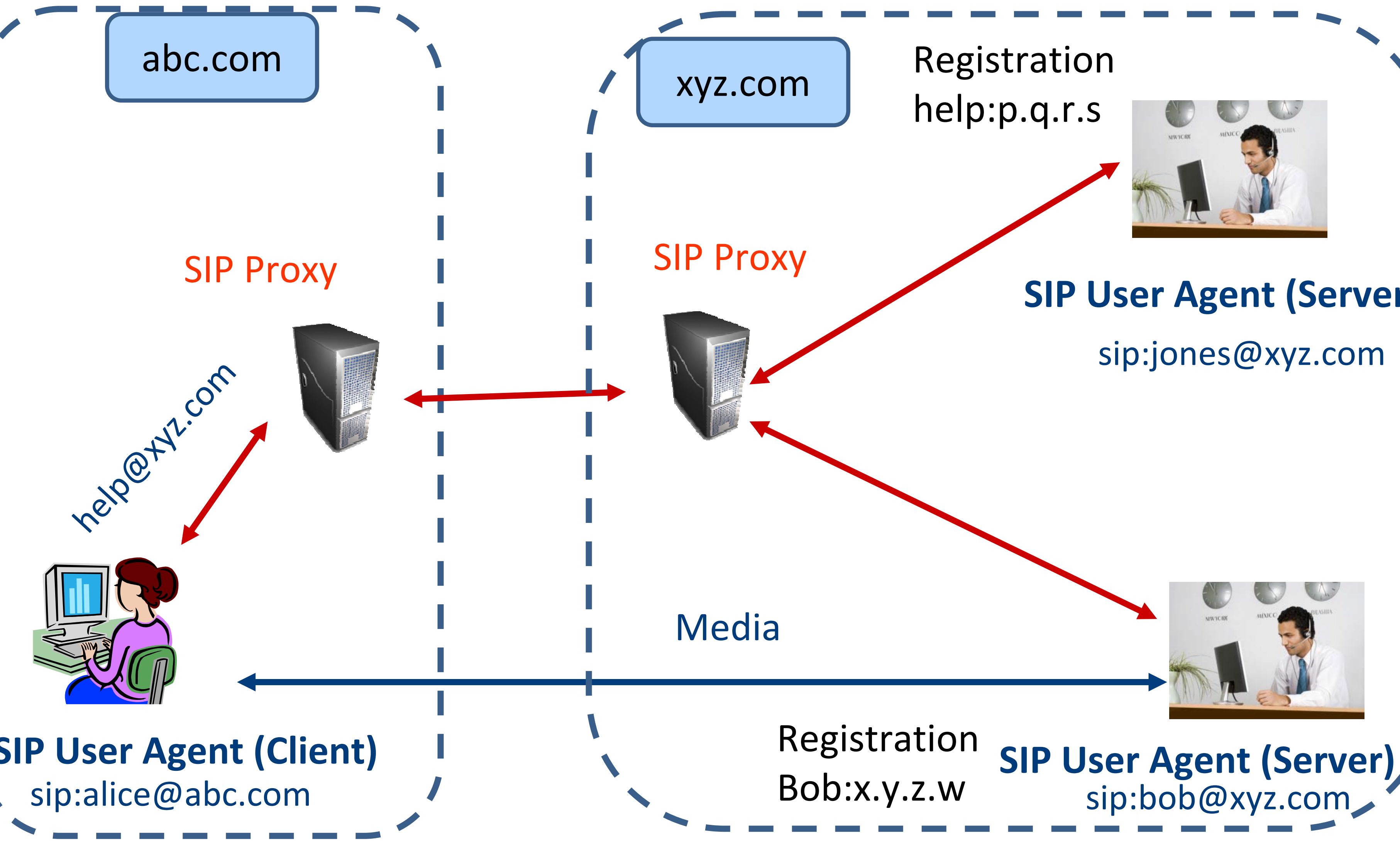
Proxy Server Example



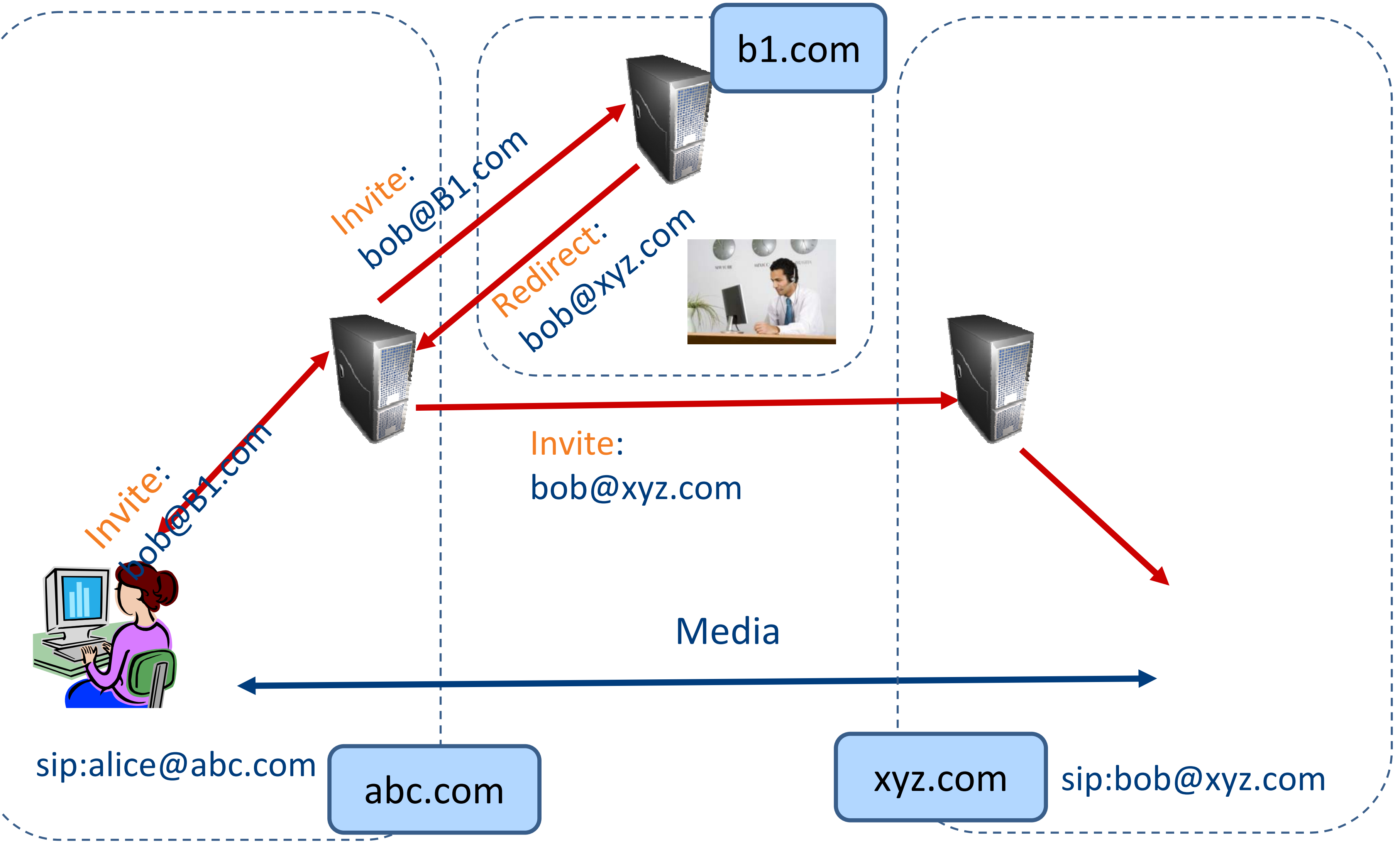
SIP: Basic Connection Setup



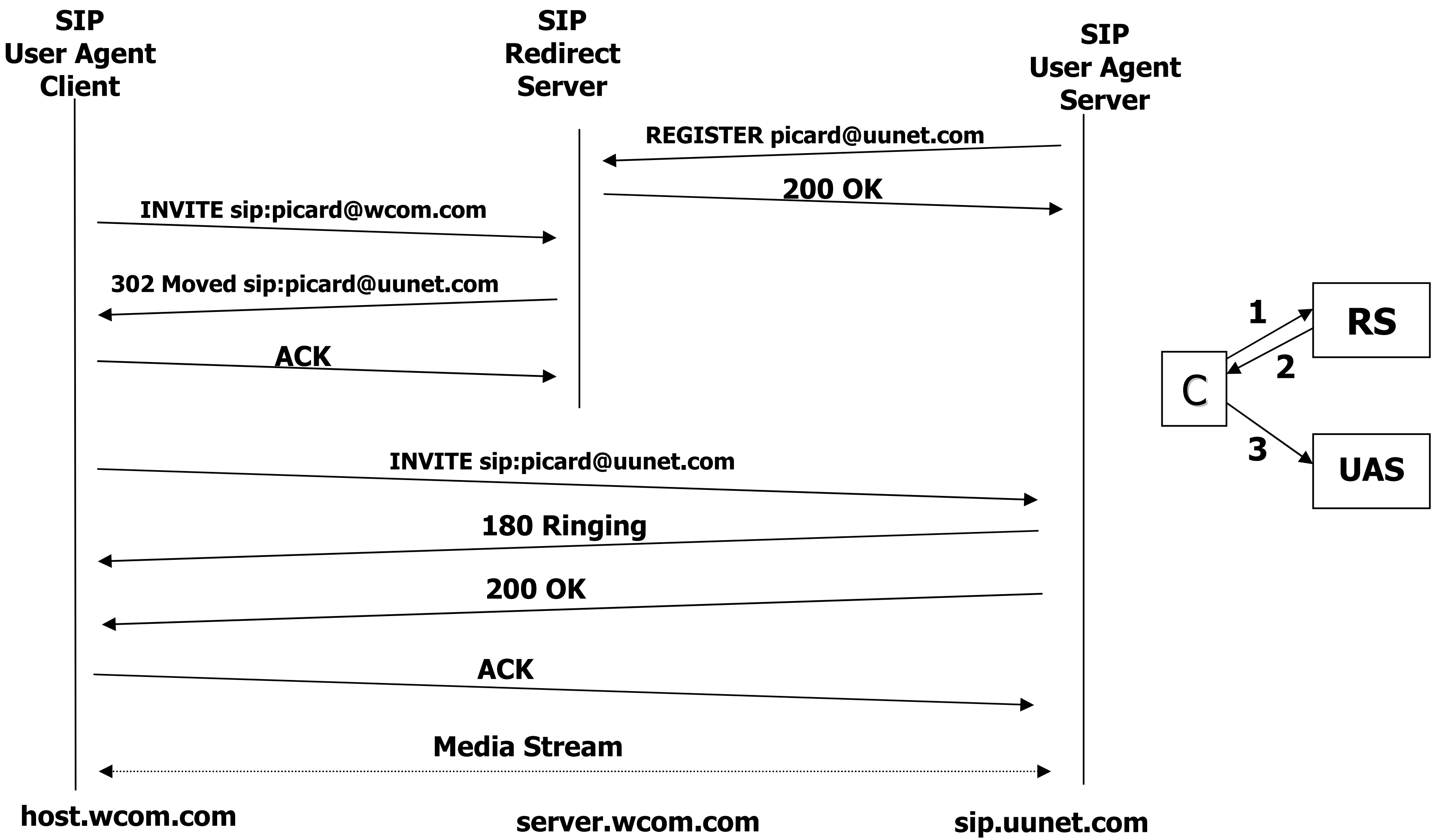
SIP Proxy Operation (Forking)



SIP Proxy Operation (Redirect)



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	Acall 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!

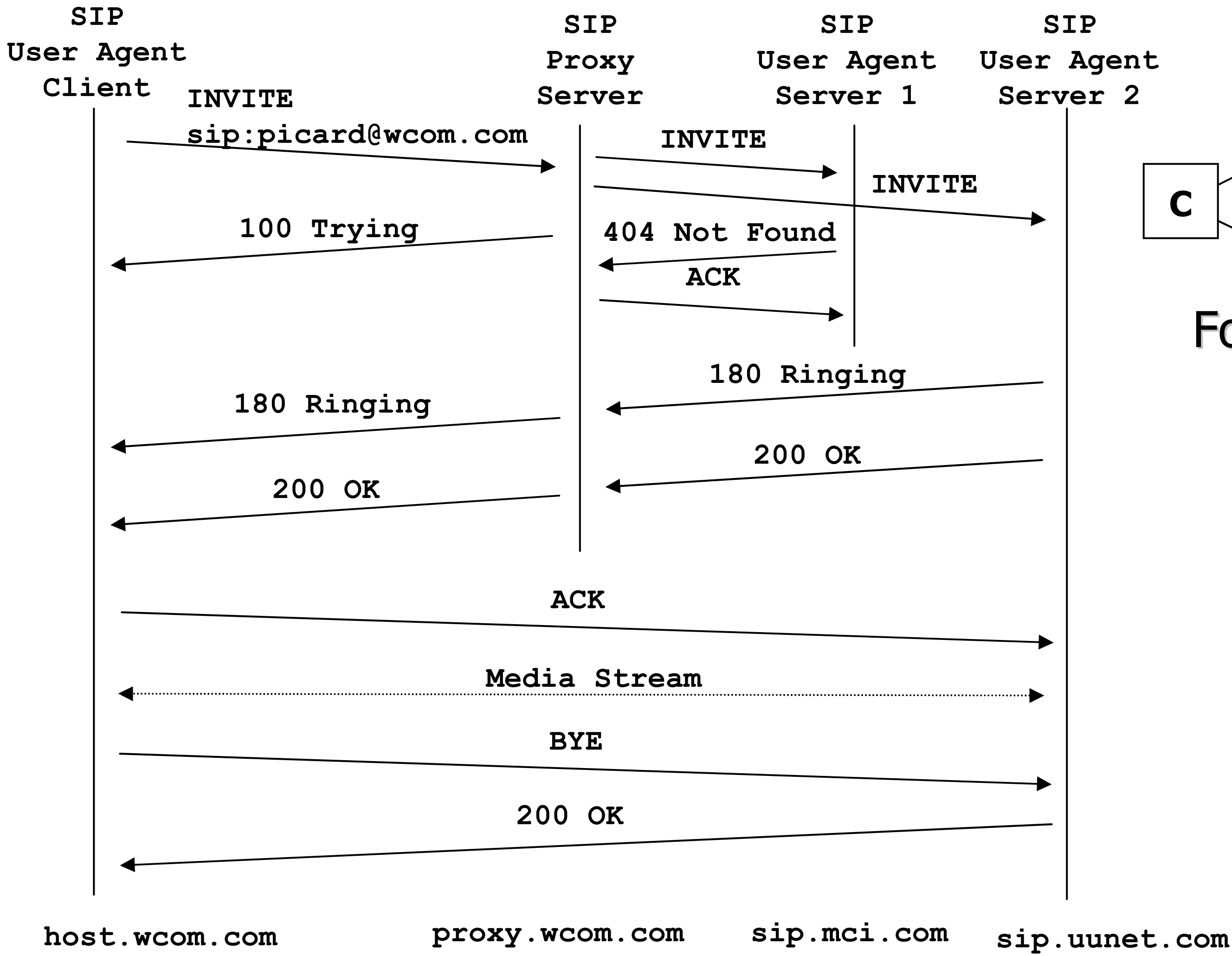
SIP Responses Example

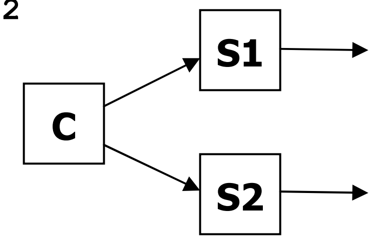
Typical SIP Response (containing SDP)

```
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





Fork

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 back to the Request initiator

Requests

- Request initiator puts address in **Via** header
- Servers (receivers) check **Via** with **received sender's IP address**, if different add "**received=…**" parameter in
- Servers (receivers) check **rport** in **Via** header, if exists add **received sender's port** in "**rport=…**" parameter (rfc 3581)
- Servers add own address into another **Via** header, and forward.

Responses

- Response initiator copies request **Via** headers.
- Servers check **Via** with own address, then forward to next **Via** address

Via Headers and Routing

Request initiator will add an rport to the via header of the SIP messages, as described in rfc3581 (see http://www.faqs.org/rfcs/rfc3581.html), this will allow a client to request that the server send the response back to the source IP address and port where the request came from.

The "rport" parameter is analogous to the "received" parameter in the VIA line, except "rport" contains a port number, not the IP 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
```