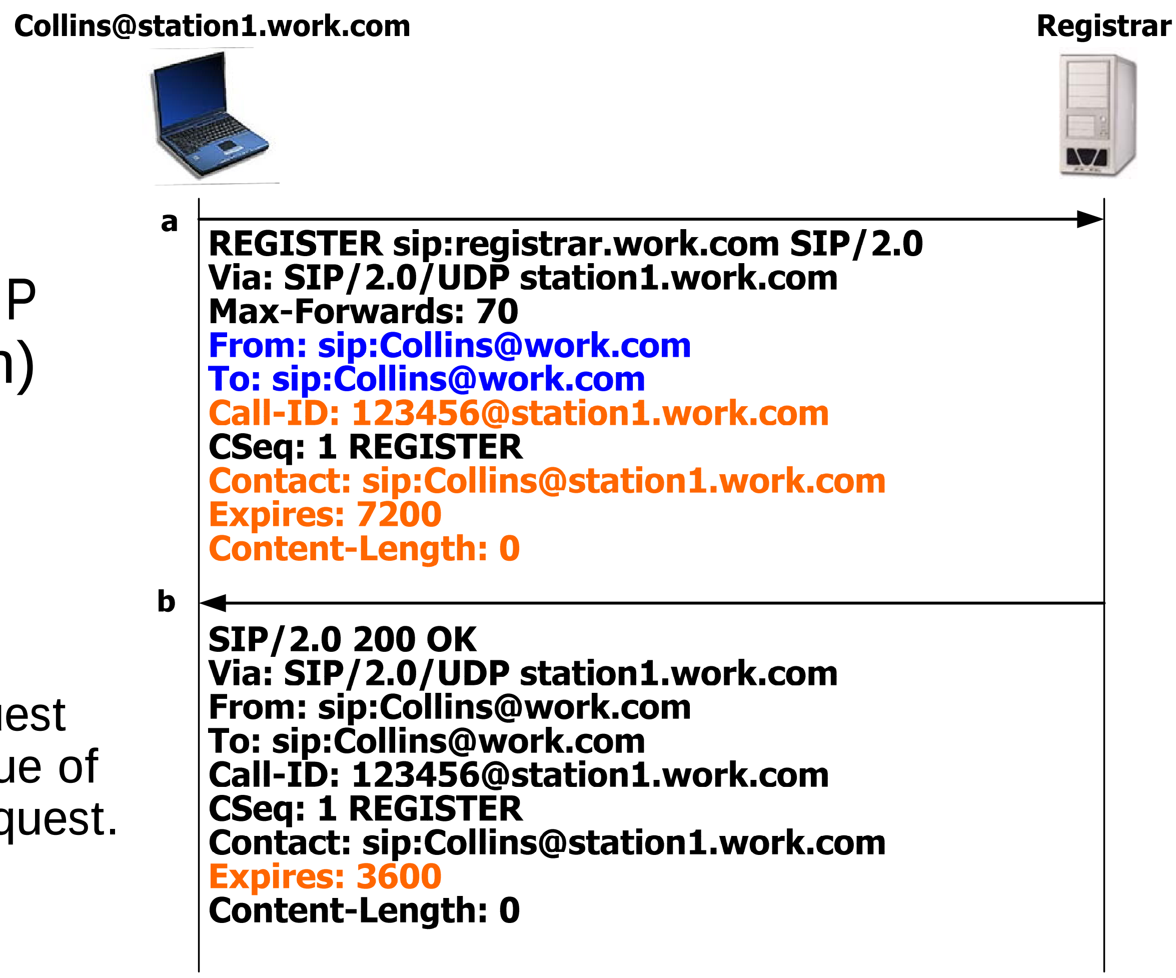
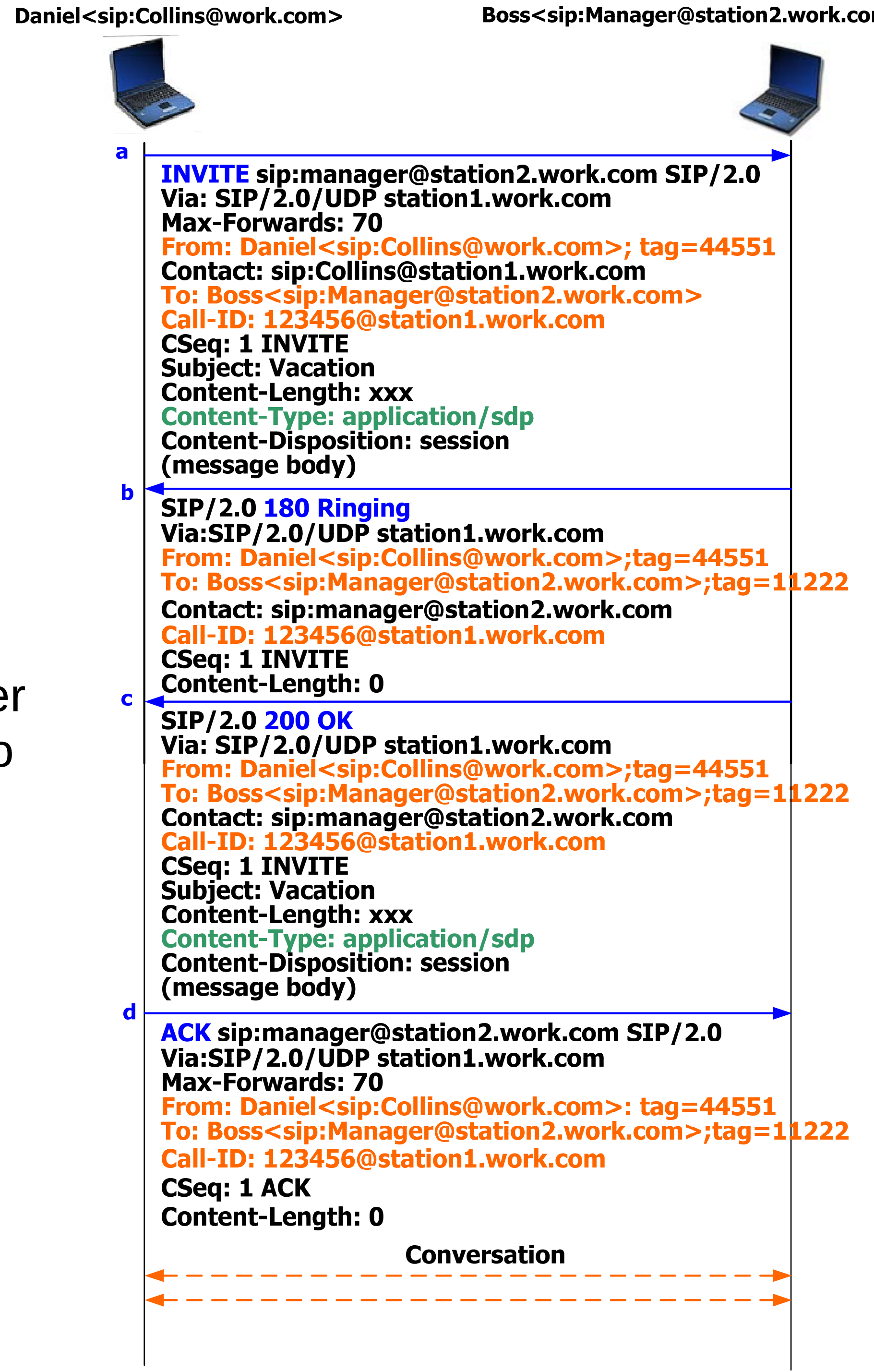
Examples of SIP Message Sequences

- Via:
- From: and To:
- Call-ID:
  - host-specific
- Contact: (for future SIP message transmission)
  - *
- Content-Length:
  - Zero, no msg body
- CSeq:
  - A response to any request must use the same value of CSeq as used in the request.
- Expires:
  - TTL
  - 0, unreg



Invitation

- A two-party call
  - Subject:
    - optional
  - Content-Type:
    - application/sdp
  - A dialog ID
    - To identify a peer-to-peer relationship between two user agents
    - Tag in From
    - Tag in To
    - Call-ID



Termination of a Call

- CSeq has changed.

Daniel<sip:Collins@work.com>

Boss<sip:Manager@station2.work.com>

a

BYE sip:manager@work.com SIP/2.0
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK123
Max-Forwards: 70
From: Daniel<sip:Collins@work.com>; tag=44551
To: Boss<sip:Manager@station2.work.com>; tag=11222
Call-ID: 123456@station1.work.com
CSeq: 2 BYE
Content-Length: 0

b

SIP/2.0 200 OK
Via: SIP/2.0/UDP station1.work.com;
branch=z9hG4bK123
From: Daniel<sip:Collins@work.com>; tag=44551
To: Boss<sip:Manager@station2.work.com>; tag=11222
Call-ID: 123456@station1.work.com
CSeq: 2 BYE
Content-Length: 0

Redirect Servers

- An alternative address
  - 302, Moved temporarily
- Another INVITE
  - Same Call-ID
  - CSeq ++

Proxy Servers [1/2]

- Sits between a user-agent client and the far-end user-agent server
- Numerous proxies can reside in a chain between the caller and callee.
  - The most common scenario will have at least two proxies: one at the caller and one at the callee end.
  - It is likely that only the last proxy in the chain changes the Request-URI.
  - The other proxies in the chain would simply use the domain part of the received Request-URI as input to a location function (e.g., DNS) to determine the next hop.

Proxy Servers [2/2]

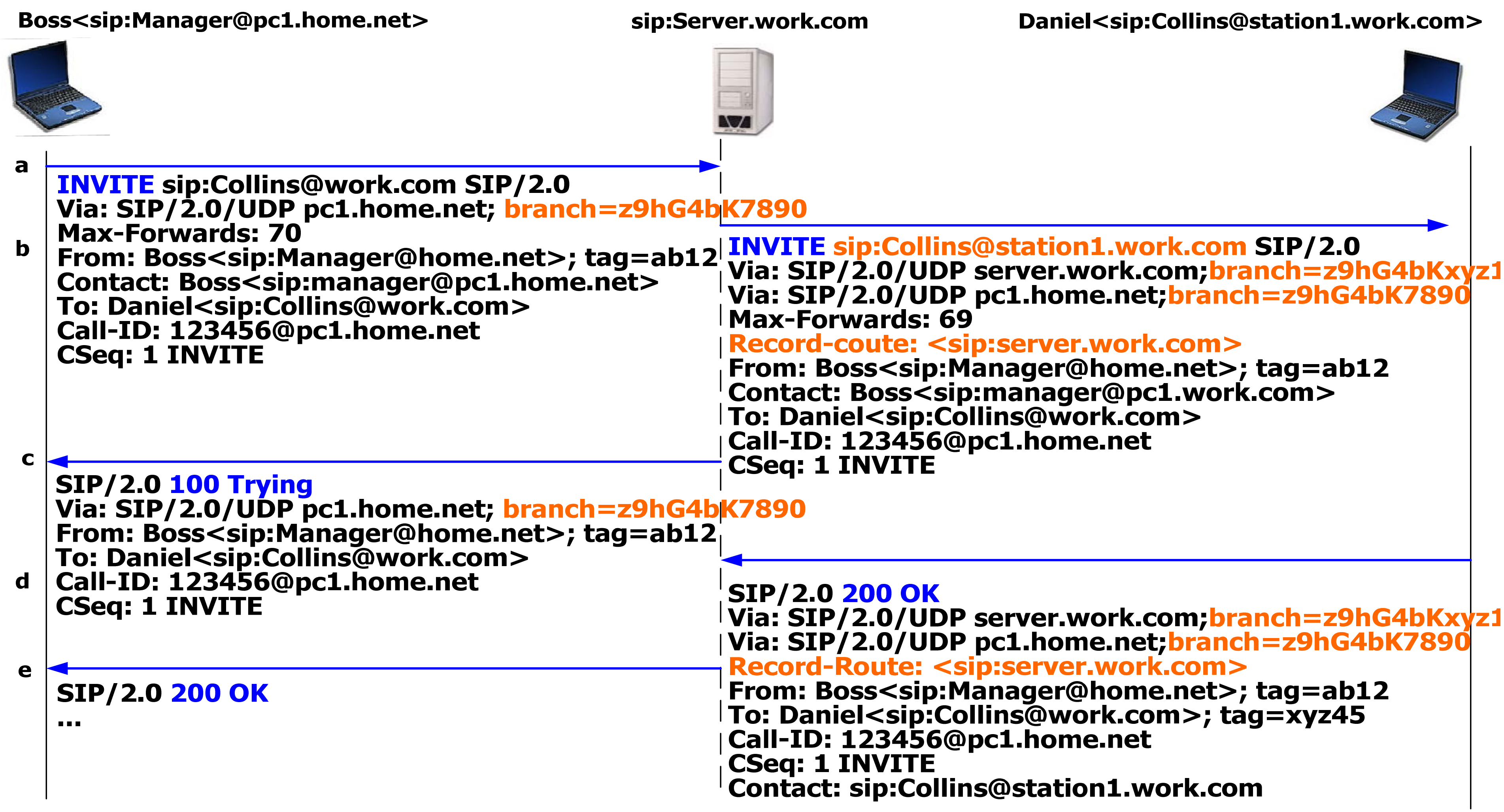
- Via:
  - The path taken by a request
  - Loop detected, 482 (status code)
  - For a response
    - The first Via: header is checked and removed.
    - The second Via: header is checked.
      - If it exists, perform forwarding.
      - If not, the response is destined to the proxy itself.
    - The response finds its way back to the originator of the request.
  - Branch: used to distinguish between multiple responses to the same request
    - Forking Proxy: Issue a single request to multiple destinations

Proxy State [1/2]

- Can be either stateless or stateful
- If stateless, the proxy takes an incoming request, performs whatever translation and forwards the corresponding outgoing request and forgets anything.
  - Retransmission takes the same path (no change on retransmission).
- If stateful, the proxy remembers incoming requests and corresponding outgoing request.
  - The proxy is able to act more intelligently on subsequent requests and responses related to the same session.

Proxy State [2/2]

- Record-Route: and Route: Headers
  - The subsequent requests may not pass through the same path as the initial request/response.
    - E.g., use Contact:
  - A Proxy might require that it remains in the signaling path for all subsequent requests to provide some advanced service.
    - In particular for a stateful proxy
  - Insert its address into the Record-Route: header
  - The response includes the Record-Route: header
  - The information contained in the Record-Route: header is used in the subsequent requests related to the same call.
  - The Route: header is used to record the path that the request is enforced to pass.


Boss<sip:Manager@pc1.home.net>
sip:Server.work.com
Daniel<sip:Collins@station1.work.com>
a
INVITE sip:Collins@work.com SIP/2.0
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
Max-Forwards: 70
From: Boss<sip:Manager@home.net>; tag=ab12
Contact: Boss<sip:manager@pc1.home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
b
INVITE sip:Collins@station1.work.com SIP/2.0
Via: SIP/2.0/UDP server.work.com;branch=z9hG4bKxyz1
Via: SIP/2.0/UDP pc1.home.net;branch=z9hG4bK7890
Max-Forwards: 69
Record-coute: <sip:server.work.com>
From: Boss<sip:Manager@home.net>; tag=ab12
Contact: Boss<sip:manager@pc1.work.com>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
c
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
From: Boss<sip:Manager@home.net>; tag=ab12
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
d
SIP/2.0 200 OK
Via: SIP/2.0/UDP server.work.com;branch=z9hG4bKxyz1
Via: SIP/2.0/UDP pc1.home.net;branch=z9hG4bK7890
Record-Route: <sip:server.work.com>
From: Boss<sip:Manager@home.net>; tag=ab12
To: Daniel<sip:Collins@work.com>; tag=xyz45
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
Contact: sip:Collins@station1.work.com
e
SIP/2.0 200 OK
...

Boss<sip:Manager@pc1.home.net> sip:Server.work.com Daniel<sip:Collins@station1.work.com>

e

SIP/2.0 200 OK
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
Record-coute: <sip:server.work.com>
From: Boss<sip:Manager@home.net>; tag=ab12
To: Daniel<sip:Collins@work.com>; tag=xyz45
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
Contact: sip:Collins@station1.work.com

f

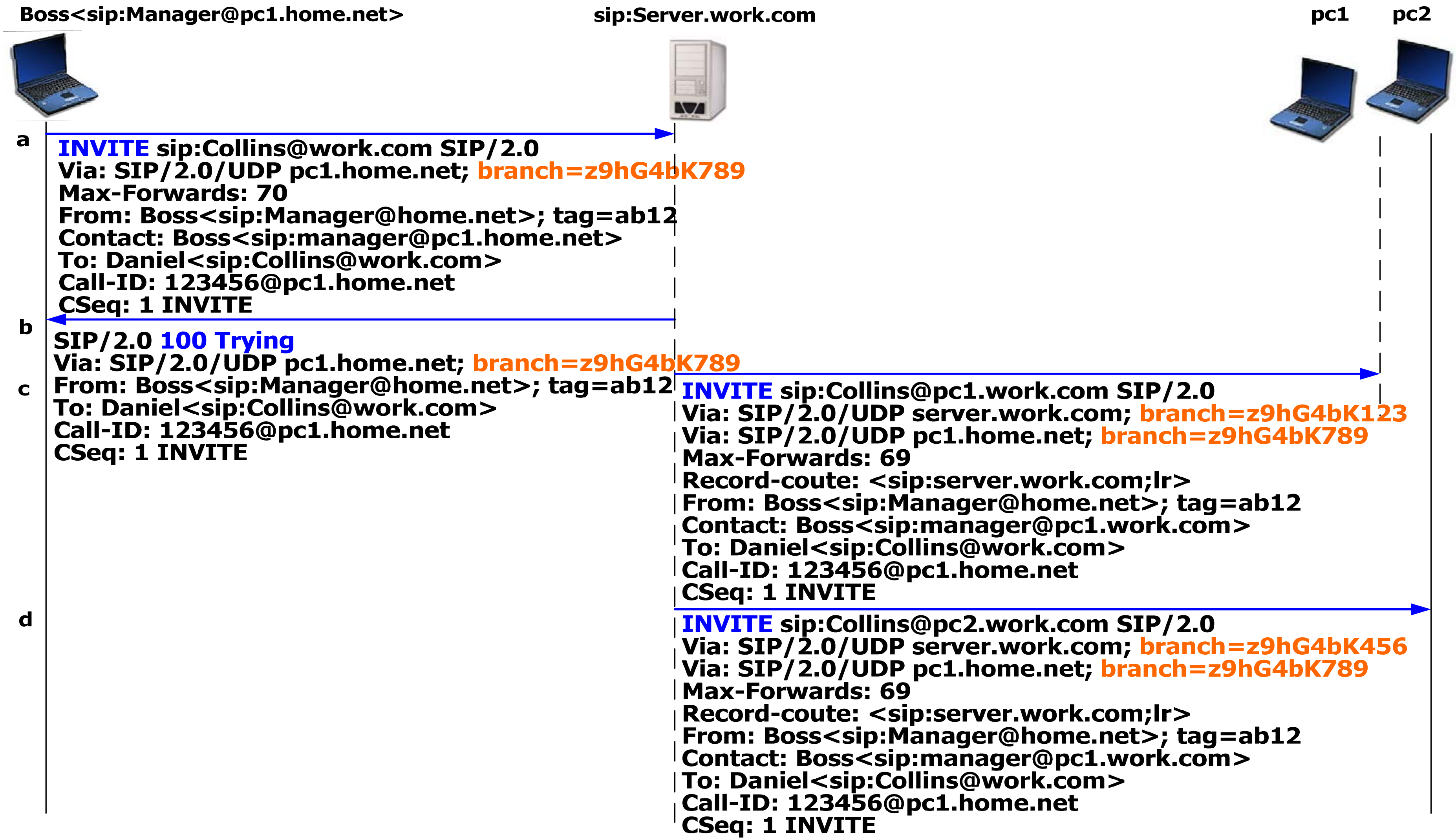
ACK sip:Collins@station1.work.com SIP/2.0
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7891
Max-Forwards: 70
Route: <sip:server.work.com>
From: Boss<sip:Manager@home.net>; tag=ab12
To: Daniel<sip:Collins@work.com>; tag=xyz45
Call-ID: 123456@pc1.home.net
CSeq: 1 ACK

g

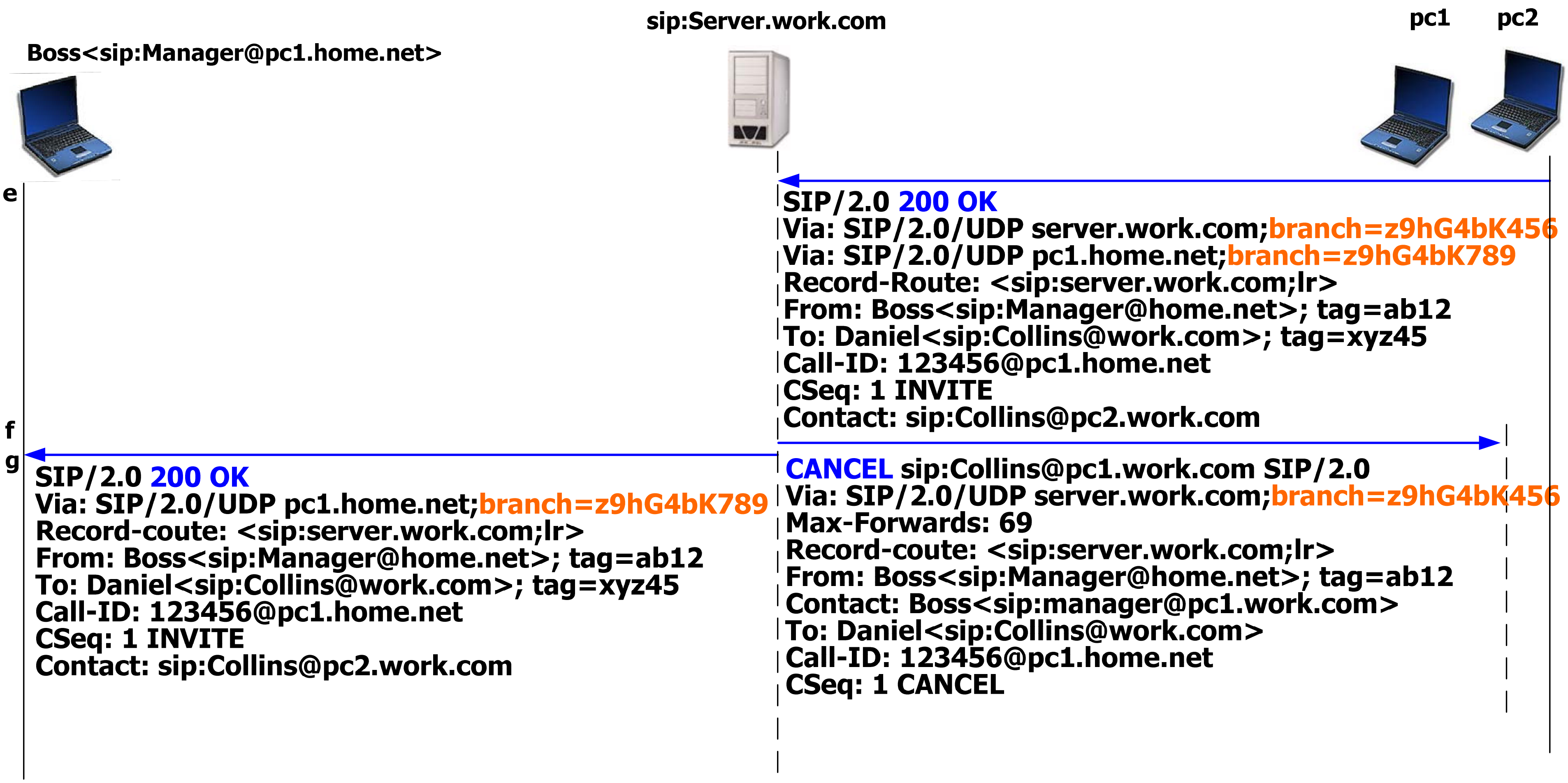
ACK sip:Collins@station1.work.com SIP/2.0
Via: SIP/2.0/UDP server.work.com;branch=z9hG4bKxyz2
Via: SIP/2.0/UDP pc1.home.net;branch=z9hG4bK7891
Max-Forwards: 69
From: Boss<sip:Manager@home.net>; tag=ab12
To: Daniel<sip:Collins@work.com>; tag=xyz45
Call-ID: 123456@pc1.home.net
CSeq: 1 ACK

Forking Proxy

- A proxy can “fork” requests
- A user is registered at several locations
  - ;branch=xxx
- In order to handle such forking, a proxy must be stateful.



Boss<sip:Manager@pc1.home.net>
sip:Server.work.com
pc1
pc2
a
INVITE sip:Collins@work.com SIP/2.0
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK789
Max-Forwards: 70
From: Boss<sip:Manager@home.net>; tag=ab12
Contact: Boss<sip:manager@pc1.home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
b
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK789
From: Boss<sip:Manager@home.net>; tag=ab12
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
c
INVITE sip:Collins@pc1.work.com SIP/2.0
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bK123
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK789
Max-Forwards: 69
Record-coute: <sip:server.work.com;lr>
From: Boss<sip:Manager@home.net>; tag=ab12
Contact: Boss<sip:manager@pc1.work.com>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
d
INVITE sip:Collins@pc2.work.com SIP/2.0
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bK456
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK789
Max-Forwards: 69
Record-coute: <sip:server.work.com;lr>
From: Boss<sip:Manager@home.net>; tag=ab12
Contact: Boss<sip:manager@pc1.work.com>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE

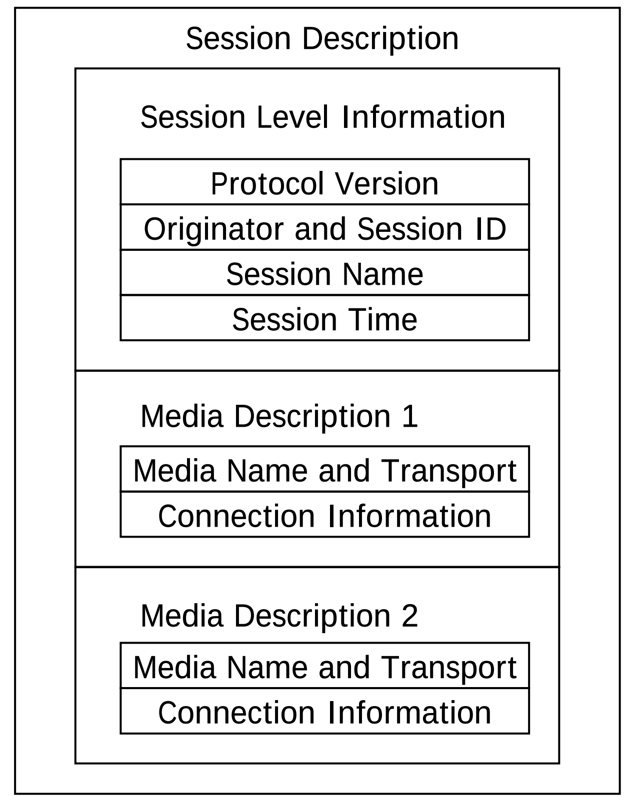

sip:Server.work.com
pc1
pc2
Boss<sip:Manager@pc1.home.net>
e
SIP/2.0 200 OK
Via: SIP/2.0/UDP server.work.com;branch=z9hG4bK456
Via: SIP/2.0/UDP pc1.home.net;branch=z9hG4bK789
Record-Route: <sip:server.work.com;lr>
From: Boss<sip:Manager@home.net>; tag=ab12
To: Daniel<sip:Collins@work.com>; tag=xyz45
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
Contact: sip:Collins@pc2.work.com
f
g
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc1.home.net;branch=z9hG4bK789
Record-coute: <sip:server.work.com;lr>
From: Boss<sip:Manager@home.net>; tag=ab12
To: Daniel<sip:Collins@work.com>; tag=xyz45
Call-ID: 123456@pc1.home.net
CSeq: 1 INVITE
Contact: sip:Collins@pc2.work.com
CANCEL sip:Collins@pc1.work.com SIP/2.0
Via: SIP/2.0/UDP server.work.com;branch=z9hG4bK456
Max-Forwards: 69
Record-coute: <sip:server.work.com;lr>
From: Boss<sip:Manager@home.net>; tag=ab12
Contact: Boss<sip:manager@pc1.work.com>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 CANCEL

The Session Description Protocol

- The Most Common Message Body
  - Session information describing the media to be exchanged between the parties
  - SDP, RFC 2327 (initial publication)
    - A number of modifications to the protocol have been suggested.
- SIP uses SDP in an answer/offer mode.
  - An agreement between the two parties as to the types of media they are willing to share
  - RFC 3264 (An Offer/Answer Model with SDP)
    - To describe how SDP and SIP should be used together

The Structure of SDP

- SDP simply provides a format for describing session information to potential session participants.
- Text-based Protocol
- The Structure of SDP
  - Session Level Info
    - Name of the session
    - Originator of the session
    - Time that the session is to be active
  - Media Level Info
    - Media type
    - Port number
    - Transport protocol
    - Media format



SDP Syntax

- A number of lines of text
- In each line
  - field=value
  - field is exactly one character (case-significant)
- Session-level fields
- Media-level fields
  - Begin with media description field (m=)

Mandatory Fields

- v=(protocol version)
- o=(session origin or creator)
- s=(session name), a text string
  - For multicast conference
- t=(time of the session), the start time and stop time
  - For pre-arranged multicast conference
- m=(media)
  - Media type
  - The transport port
  - The transport protocol
  - The media format (typically an RTP payload format)

Optional Fields [1/3]

- Some optional fields can be applied at both session and media levels.
  - The value applied at the media level overrides that at the session level
- i=(session information)
  - A text description
  - At both session and media levels
  - It would be somewhat superfluous since SIP already supports the Subject header.
- u=(URI of description)
  - Where further session information can be obtained
  - Only at session level

Optional Fields [2/3]

- e=(e-mail address)
  - Who is responsible for the session
  - Only at the session level
- p=(phone number)
  - Only at the session level
- c=(connection information)
  - Network type, address type and connection address
  - At session or media level
- b=(bandwidth information)
  - In kilobits per second
  - At session or media level

Optional Fields [3/3]

- r=(repeat times)
  - For regularly scheduled session a session is to be repeated
  - How often and how many times
- z=(timezone adjustments)
  - For regularly scheduled session
  - Standard time and daylight savings time
- k=(encryption key)
  - An encryption key or a mechanism to obtain it for the purposes of encrypting and decrypting the media
  - At session or media level
- a=(attributes)
  - Describe additional attributes

Ordering of Fields

- Session Level
  - Protocol version (v)
  - Origin (o)
  - Session name (s)
  - Session information (i)
  - URI (u)
  - E-mail address (e)
  - Phone number (p)
  - Connection info (c)
  - Bandwidth info (b)
  - Time description (t)
  - Repeat info (r)
  - Time zone adjustments (z)
  - Encryption key (k)
  - Attributes (a)
- Media level
  - Media description (m)
  - Media info (i)
  - Connection info (c)
    - Optional if specified at the session level
  - Bandwidth info (b)
  - Encryption key (k)
  - Attributes (a)

Subfields [1/3]

- Field = <value of subfield1> <value of subfield2> <value of subfield3>
- Origin
  - Username, the originator's login id or "-"
  - Session ID
    - A unique ID
    - Make use of NTP timestamp
  - Version, a version number for this particular session
  - Network type
    - A text string
    - IN refers to Internet
  - Address type
    - IP4, IP6
  - Address, a fully-qualified domain name or the IP address

Subfields [2/3]

- Connection Data
  - The network and address at which media data will be received
  - Network type
  - Address type
  - Connection address
- Media Information
  - Media type
    - Audio, video, data, or control
  - Port
  - Format
    - List the various types of media format that can be supported
    - According to the RTP audio/video profile
  - m= audio 45678 RTP/AVP 15 3 0
    - G.728, GSM, G.711

Subfields [3/3]

- Attributes
  - To enable additional information to be included
  - Property attribute
    - a=sendonly
    - a=recvonly
  - Value attribute
    - a=orient:landscape used in a shared whiteboard session
  - Rtpmap attribute
    - The use of dynamic payload type
    - a=rtpmap:<payload type> <encoding name>/<clock rate> [/<encoding parameters>].
    - m=video 54678 RTP/AVP 98
    - a=rtpmap 98 L16/16000/2
      - 16-bit linear encoded stereo (2 channels) audio sampled at 16kHz

Usage of SDP with SIP

- SIP and SDP make a wonderful partnership for the transmission of session information.
- SIP provides the messaging mechanism for the establishment of multimedia sessions.
- SDP provides a structured language for describing the sessions.
  - The entity headers identifies the message body.

SIP Inclusion in SIP Messages

- Fig 5-15
  - G.728 is selected
- INVITE with multiple media streams
  - Unsupported should also be returned with a port number of zero
- An alternative
  - INVITE

    m=audio 4444 RTP/AVP 2 4 15

    a=rtpmap 2 G726-32/8000

    a=rtpmap 4 G723/8000

    a=rtpmap 15 G728/8000
  - 200 OK

    m=audio 6666 RTP/AVP 15

    a=rtpmap 15 G728/8000

Daniel<sip:Collins@station1.work.com>

Boss<sip:Manager@station2.work.com>

a

INVITE sip:Manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>
CSeq: 1 INVITE
Content-Length: 213
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 123456 001 IN IP4 station1.work.com
s=
c=IN IP4 station1.work.com
t=0 0
m=audio 4444 RTP/AVP 2
a=rtpmap 2 G726-32/8000
m=audio 4666 RTP/AVP 4
a=rtpmap 4 G723/8000
m=audio 4888 RTP/AVP 15
a=rtpmap 15 G728/8000

b

SIP/2.0 200 OK
...

Daniel<sip:Collins@station1.work.com>

Boss<sip:Manager@station2.work.com>

b

SIP/2.0 200 OK
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>; tag = xyz789
CSeq: 1 INVITE
Content-Length: 163
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 45678 001 IN IP4 station2.work.com
s=
c=IN IP4 station2.work.com
t=0 0
m=audio 0 RTP/AVP 2
m=audio 0 RTP/AVP 4
m=audio 6666 RTP/AVP 15
a=rtpmap 15 G728/8000

c

ACK sip:manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>; tag = xyz789
CSeq: 1 ACK
Content-Length: 0

d

Conversation

SIP and SDP Offer/Answer Model

- Re-INVITE is issued when the server replies with more than one codec.
  - With the same dialog identifier (To and From headers, including tag values), Call-ID and Request-URI
  - The session version is increased by 1 in o= line of message body.
- A mismatch
  - 488 or 606
  - Not Acceptable
  - A Warning header with warning code 304 (media type not available) or 305 (incompatible media type)
  - Then the caller issues a new INVITE request.

Daniel<sip:Collins@station1.work.com>

Boss<sip:Manager@station2.work.com>

a

INVITE sip:manager@station2.work.com SIP/2.0
CSeq: 1 INVITE
Content-Length: 183
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 123456 001 IN IP4 station1.work.com
s=
c=IN IP4 station1.work.com
t=0 0
m=audio 4444 RTP/AVP 2 4 15
a=rtpmap 2 G726-32/8000
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000
a=inactive

b

SIP/2.0 200 OK
CSeq: 1 INVITE
Content-Length: 157
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 45678 001 IN IP4 station2.work.com
s=
c=IN IP4 station2.work.com
t=0 0
m=audio 6666 RTP/AVP 4 15
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000
a=inactive

Daniel<sip:Collins@station1.work.com>

Boss<sip:Manager@station2.work.com>

c

ACK sip:manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234
To: Boss<sip:Manager@station2.work.com>; tag = xyz789
CSeq: 1 ACK
Content-Length: 0

d

INVITE sip:manager@station2.work.com SIP/2.0
CSeq: 2 INVITE
Content-Length: 126
Content-Type: application/sdp
Content-Disposition: session

v=0
o=collins 123456 002 IN IP4 station1.work.com
s=
c=IN IP4 station1.work.com
t=0 0
m=audio 4444 RTP/AVP 15
a=rtpmap 15 G728/8000

OPTIONS Method

- Determine the capabilities of a potential called party
- Accept Header
  - Indicate the type of information that the sender hopes to receive
- Allow Header
  - Indicate the SIP methods that servers/clients can handle
- Supported Header
  - Indicate the SIP extensions that can be supported

Daniel<sip:Collins@station1.work.com>

Boss<sip:Manager@station2.work.com>

a

```
OPTIONS sip:manager@station2.work.com SIP/2.0
Via: SIP/2.0/UDP Station1.work.com; branch=z9hG4bK7890123
From: Daniel<sip:Collins@work.com>; tag=lmnop123
To: Boss<sip:Manager@station2.work.com>
Call-ID: 123456@station1.work.com
Contact: Daniel<sip:Collins@station1.work.com>
CSeq: 1 OPTIONS
Accept: application/sdp
Content-Length: 0
```

b

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP Station1.work.com; branch=z9hG4bK7890123
From: Daniel<sip:Collins@work.com>; tag=lmnop123
To: Boss<sip:Manager@station2.work.com>; tag=xyz5678
Call-ID: 123456@station1.work.com
CSeq: 1 OPTIONS
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE
Supported: newfield
Content-Length: 146
Content-Type: application/sdp

v=0
o=manager 45678 001 IN IP4 station2.work.com
s=
c=IN IP4 station2.work.com
t=0 0
m=audio 0 RTP/AVP 4 15
a=rtpmap 4 G723/8000
a=rtpmap 15 G728/8000
```

SIP Extensions and Enhancements

- RFC 2543, March 1999
- RFC 3261, June 2002
  - SIP has attracted enormous interest.
  - Traditional telecommunications companies, cable TV providers and ISP
- A large number of extensions to SIP have been proposed.
  - SIP will be enhanced considerably before it becomes an Internet standard.

183 Session Progress

- It has been included within the revised SIP spec.
  - To open one-way audio path from called end to calling end
    - Enable in-band call progress information to be transmitted
      - Tones or announcements
  - Interworking with SS7 network
    - ACM (Address Complete Message)
    - For SIP-PSTN-SIP connections

The Supported Header

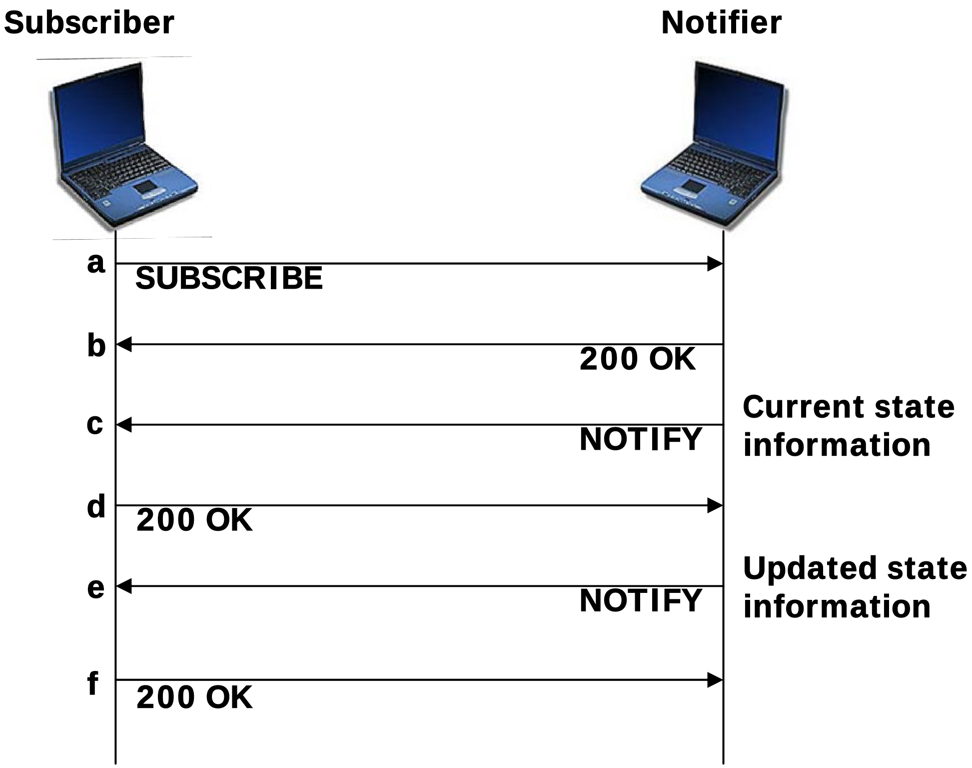
- Base RFC 2543
  - Require: Header
    - In request (client ->server)
      - A client indicates that a server must support certain extension.
  - Unsupported Header
    - In response (server -> client)
      - 420 (bad extension)
  - A cumbersome way of determining what extensions a server does or does not support
- Supported: Header (RFC 3261)
  - May be included in the response corresponding to OPTIONS request
  - Can also be included in response of 421 (extension required) to indicate that the server requires a particular extension of the client.

SIP INFO Method

- Be specified in RFC 2976
- For transferring information during an ongoing session
  - DTMF digits, account-balance information, mid-call signaling information (from PSTN)
  - Application-layer information could be transferred in the middle of a call.
- A powerful, flexible tool to support new services

SIP Event Notification

- Several SIP-based applications have been devised based on the concept of a user being informed of some event.
  - E.g., Instant messaging
- RFC 3265 has addressed the issue of event notification.
  - SUBSCRIBE and NOTIFY
  - The Event header



SIP for Instant Messaging

- The IETF working group – SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE)
- A new SIP method – MESSAGE
  - This request carries the actual message in a message body.

Boss<sip:Manager@pc1.home.com>

sip:Server.work.com

Daniel<sip:Collins@station1.work.com>

a

MESSAGE sip:Collins@work.com SIP/2.0
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
Max-Forwards: 70
From: Boss<sip:Manager@home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 19
Content-Disposition: render

Hello. How are you?

b

MESSAGE sip:Collins@station1.work.com SIP/2.0
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bKxyz1
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
Max-Forwards: 69
From: Boss<sip:Manager@home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 19
Content-Disposition: render

Hello. How are you?

c

SIP/2.0 200 OK
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bKxyz1
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
From: Boss<sip:Manager@home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 MESSAGE
Content-Length: 0

d

SIP/2.0 200 OK
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
From: Boss<sip:Manager@home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 MESSAGE
Content-Length: 0

Boss<sip:Manager@pc1.home.com> sip:Server.work.com Daniel<sip:Collins@station1.work.com>

MESSAGE sip:Manager@home.net SIP/2.0
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK123
Max-Forwards: 70
From: Daniel<sip:Collins@work.com>
To: Boss<sip:Manager@home.net>
Call-ID: 456789@station1.work.com
CSeq: 1101 MESSAGE
Content-Type: text/plain
Content-Length: 22
Content-Disposition: render

I'm fine. How are you?

MESSAGE sip:Manager@pc1.home.net SIP/2.0
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bKabcd
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK123
Max-Forwards: 69
From: Daniel<sip:Collins@work.com>
To: Boss<sip:Manager@home.net>
Call-ID: 456789@station1.work.com
CSeq: 1101 MESSAGE
Content-Type: text/plain
Content-Length: 22
Content-Disposition: render

I'm fine. How are you?

SIP/2.0 200 OK
Via: SIP/2.0/UDP server.work.com; branch=z9hG4bKabcd
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK123
From: Daniel<sip:Collins@work.com>
To: Boss<sip:Manager@home.net>
Call-ID: 456789@station1.work.com
CSeq: 1101 MESSAGE
Content-Length: 0

SIP/2.0 200 OK
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK123
From: Daniel<sip:Collins@work.com>
To: Boss<sip:Manager@home.net>
Call-ID: 456789@station1.work.com
CSeq: 1101 MESSAGE
Content-Length: 0

SIP REFER Method

- To enable the sender of the request to instruct the receiver to contact a third party
  - With the contact details for the third party included within the REFER request
  - For Call Transfer applications
- The Refer-to: and Refer-by: Headers
- The dialog between Mary and Joe remains established.
  - Joe could return to the dialog after consultation with Susan.

sip:Mary@station1.work.com

sip:Joe@station2.work.com

sip:Susan@station3.work.com

a

REFER sip:Joe@station2.work.com SIP/2.0
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK789
Max-Forwards: 70
From: Mary<sip:Mary@work.com>; tag=123456
To: Joe<sip:Joe@work.com>; tag=67890
Contact: Mary<Mary@station1.work.com>
Refer-To: Sussan<sip:Sussan@station3.work.com>
Call-ID: 123456@station1.work.com
CSeq: 123 REFER
Content-Length: 0

b

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP station1.work.com; branch=z9hG4bK789
From: Mary<sip:Mary@work.com>; tag=123456
To: Joe<sip:Joe@work.com>; tag=67890
Contact: Joe<Joe@station2.work.com>
Call-ID: 123456@station1.work.com
CSeq: 123 REFER
Content-Length: 0

c

INVITE sip:Susan@station3.work.com SIP/2.0
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bKxyz1
Max-Forwards: 70
From: Joe<sip:Joe@work.com>; tag=abcxyz
To: Susan<sip:Susan@station3.work.com>
Contact: Joe<Joe@station2.work.com>
Call-ID: 67890@station2.work.com
CSeq: 567 INVITE
Content-Type: application/sdp
Content-Length: xx
Content-Disposition: session
{message body}

sip:Mary@station1.work.com sip:Joe@station2.work.com sip:Susan@station3.work.com



e

SIP/2.0 200 OK
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bKxyz1
From: Joe<sip:Joe@work.com>; tag=abcxyz
To: Susan<sip:Susan@station3.work.com>; tag=123xyz
Call-ID: 67890@station2.work.com
CSeq: 567 INVITE
Content-Type: application/sdp
Content-Length: xx
Content-Disposition: session
{message body}

f

ACK sip:Susan@station3.work.com SIP/2.0
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bKxyz1
Max-Forwards: 70
From: Joe<sip:Joe@work.com>; tag=abcxyz
To: Susan<sip:Susan@station3.work.com>; tag=123xyz
Call-ID: 67890@station2.work.com
CSeq: 567 ACK
Content-Length: 0

g

NOTIFY sip:Mary@station1.work.com SIP/2.0
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bK123
Max-Forwards: 70
From: Joe<sip:Joe@work.com>
To: Mary<sip:Mary@work.com>
Contact: Joe<Joe@station2.work.com>
Call-ID: 123456@station1.work.com
CSeq: 124 NOTIFY
Content-Type: message/sipfrag;version=2.0
Content-Length: 15

h

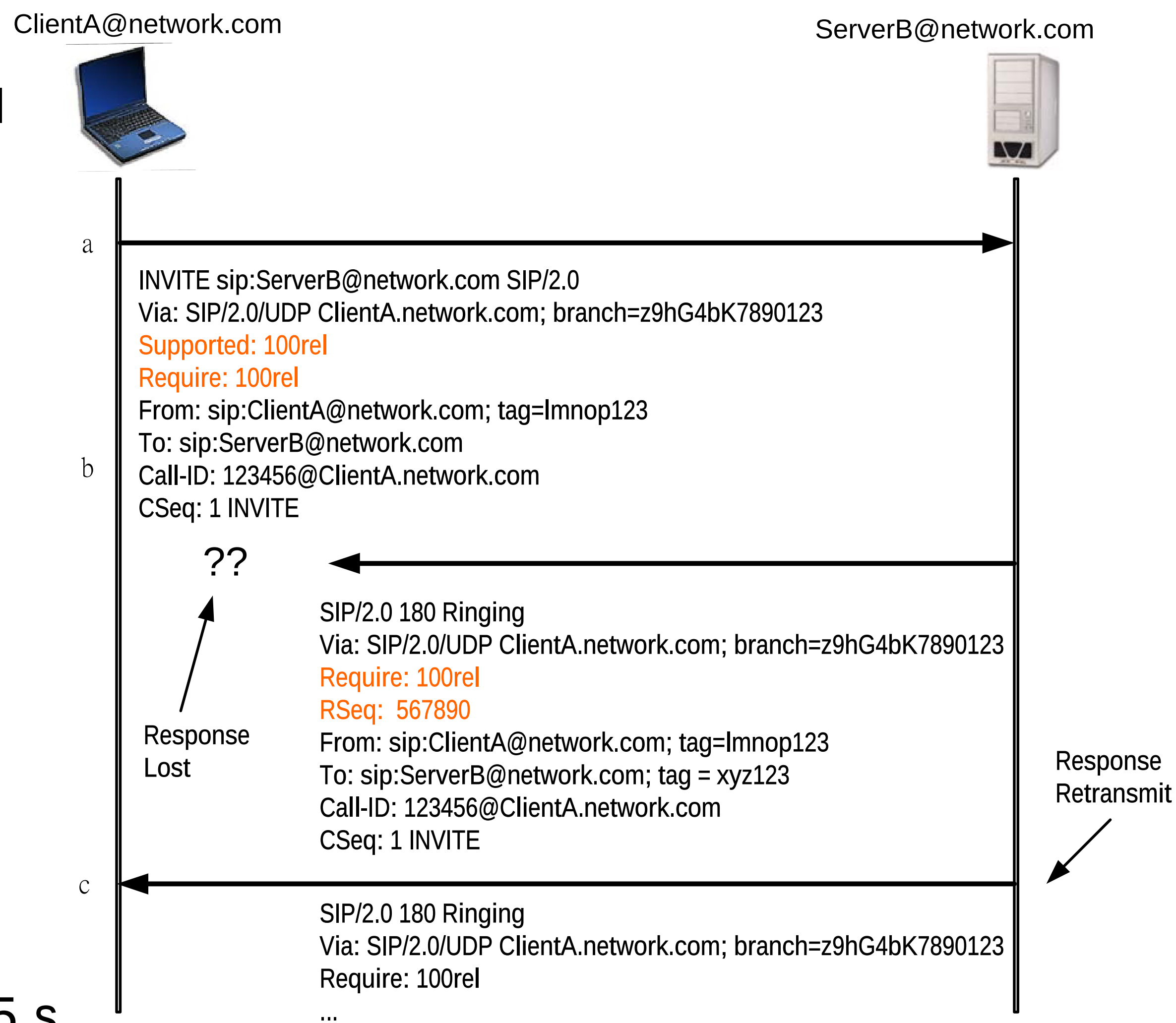
SIP/2.0 200 OK
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bK123
From: Joe<sip:Joe@work.com>
To: Mary<sip:Mary@work.com>
Call-ID: 123456@station1.work.com
CSeq: 124 NOTIFY
Content-Length: 0

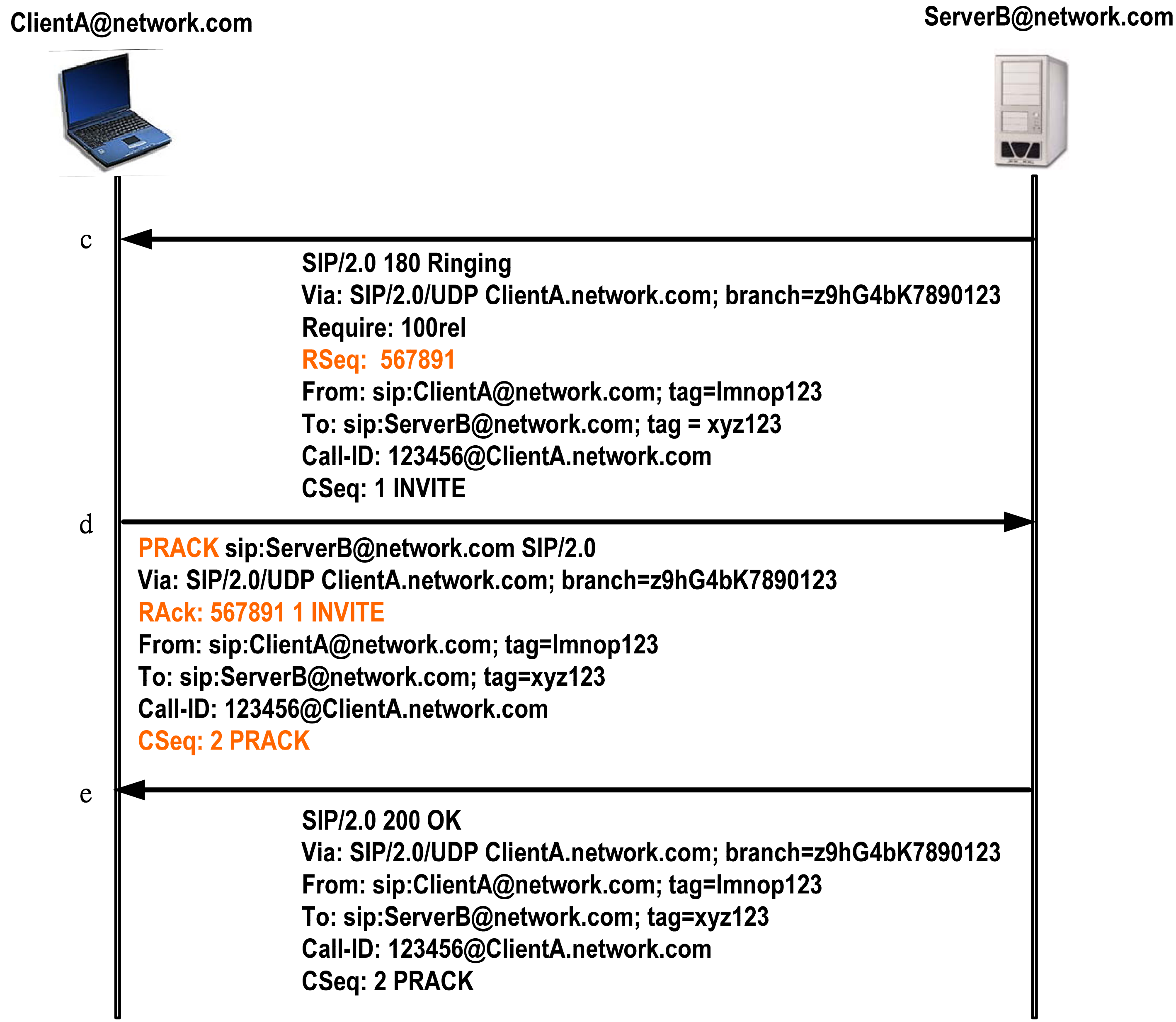
Reliability of Provisional Responses [1/2]

- Provisional Responses
  - 100 (trying), 180 (ringing), 183 (session in progress)
  - Are not answered with an ACK
- If the messages is sent over UDP
  - Unreliable
- Lost provisional response may cause problems when interoperating with other network
  - 180, 183 → Q.931 alerting or ISUP ACM
  - To drive a state machine
  - E.g., a call to an unassigned number
    - ACM to create a one-way path to relay an announcement such as "The number you have called has been changed"
    - If the provisional response is lost, the called might left in the dark and not understand why the call did not connect.

Reliability of Provisional Responses [2/2]

- RFC 3262
  - Reliability of Provisional Responses in SIP
- Supported: 100rel
- RSeq Header
  - Response Seq
  - +1, when retxm
- RAck Header
  - Response ACK
  - In PRACK
  - RSeq+CSeq
- PRACK
  - Prov. Resp. ACK
- Should not
  - Apply to 100
- Default timer value = 0.5 s




ClientA@network.com
ServerB@network.com
c
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
Require: 100rel
RSeq: 567891
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com; tag = xyz123
Call-ID: 123456@ClientA.network.com
CSeq: 1 INVITE
d
PRACK sip:ServerB@network.com SIP/2.0
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
RAck: 567891 1 INVITE
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com; tag=xyz123
Call-ID: 123456@ClientA.network.com
CSeq: 2 PRACK
e
SIP/2.0 200 OK
Via: SIP/2.0/UDP ClientA.network.com; branch=z9hG4bK7890123
From: sip:ClientA@network.com; tag=lmnop123
To: sip:ServerB@network.com; tag=xyz123
Call-ID: 123456@ClientA.network.com
CSeq: 2 PRACK

The SIP UPDATE Method

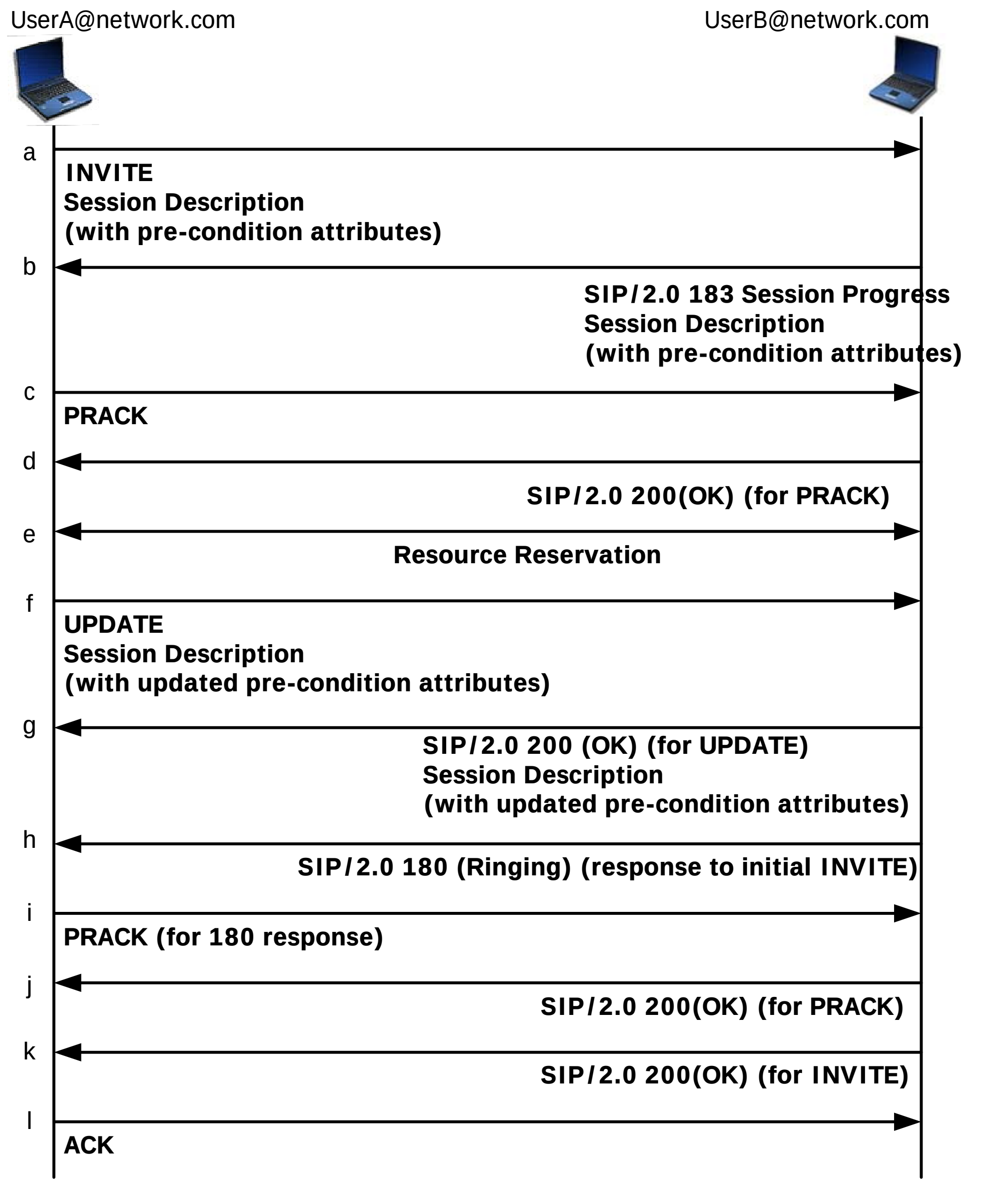
- To enable the modification of session information before a final response to an INVITE is received
  - The dialog is in the early state (An INVITE that receives a 183 response that includes a message body)
    - The message body might establish a media stream from callee to caller for sending a ring tone or music while the called party is alerted.
  - The UPDATE method can be used to change the codec
- Another important usage is when reserving network resources as part of a SIP session establishment.

Integration of SIP Signaling and Resource Management [1/2]

- Ensuring that sufficient resources are available to handle a media stream is very important.
  - To provide a high-quality service for a carrier-grade network
- The signaling might take a different path from the media.
  - The successful transfer of signaling messages does not imply to a successful transfer of media.
- Assume resource-reservation mechanisms are available (Chapter 8)
  - On a per-session basis
    - End-to-end network resources are reserved as part of session establishment.
  - On an aggregate basis
    - A certain amount of network resources are reserved in advance for a certain type of usage.
    - Policing functions at the edge of the network

Integration of SIP Signaling and Resource Management [2/2]

- Reserving network resources in advance of altering the called user
- A new draft – "Integration of Resource Management and SIP"
  - By using the provisional responses and UPDATE method
  - By involving extensions to SDP



Example of e2e Resource Reservation [1/2]

- SDP for initial INVITE

v=0
o=userA 45678 001 IN IP4 stationA.network.com
s=
c=IN IP4 stationA.nework.com
t=0 0
m=audio 4444 RTP/AVP 0
a=curr: qos e2e none
a=des: qos mandatory e2e sendrecv

- SDP for 183 response

v=0
o=userB 12345 001 IN IP4 stationB.network.com
s=
c=IN IP4 stationB.nework.com
t=0 0
m=audio 6666 RTP/AVP 0
a=curr: qos e2e none
a=des: qos mandatory e2e sendrecv
a=conf: qos e2e recv

Example of e2e Resource Reservation [2/2]

- SDP for UPDATE

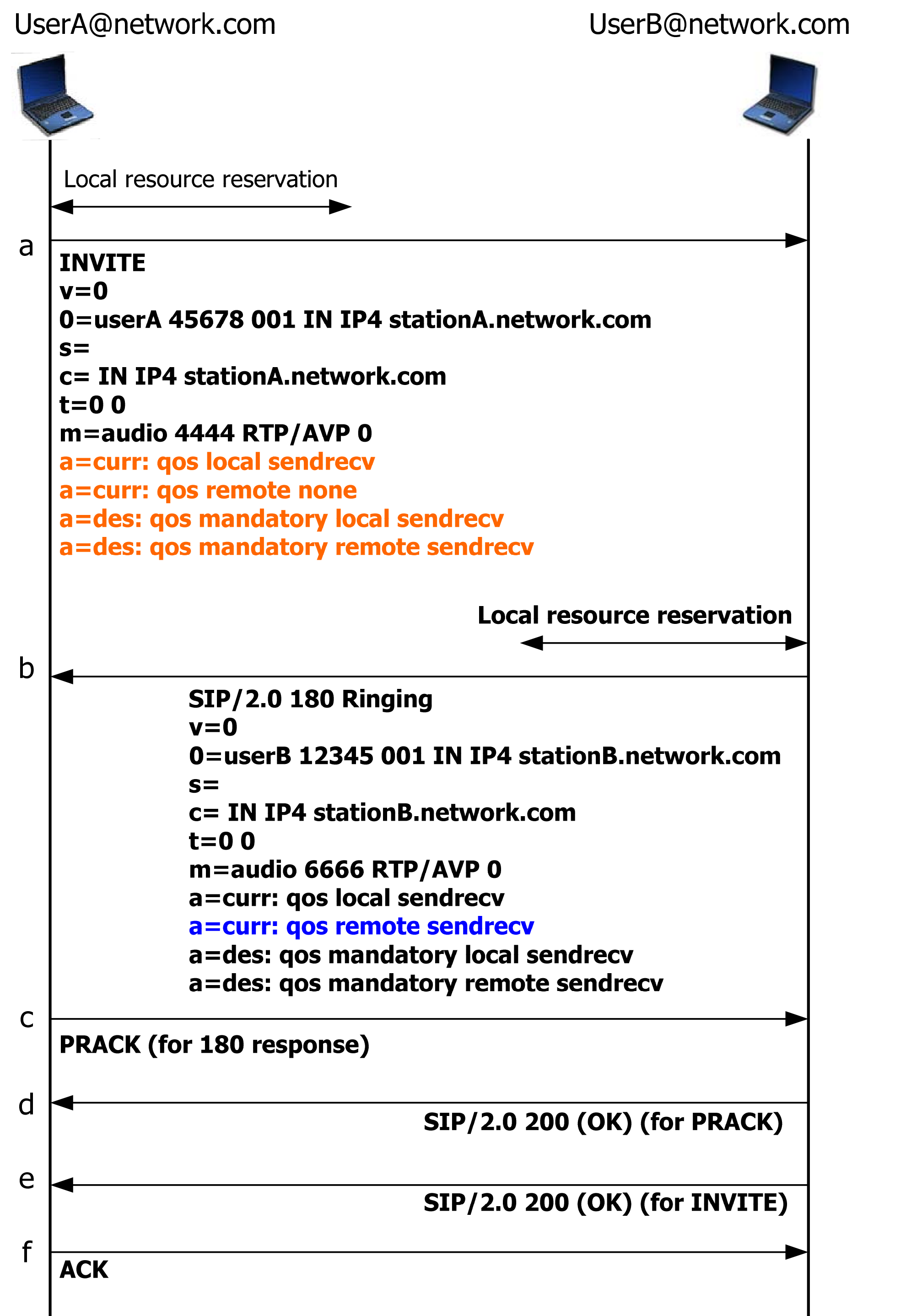
  v=0
  o=userA 45678 001 IN IP4 stationA.network.com
  s=
  c=IN IP4 stationA.nework.com
  t=0 0
  m=audio 4444 RTP/AVP 0
  a=curr: qos e2e send
  a=des: qos mandatory e2e sendrecv

- SDP for 200 response

  v=0
  o=userB 12345 001 IN IP4 stationB.network.com
  s=
  c=IN IP4 stationB.nework.com
  t=0 0
  m=audio 6666 RTP/AVP 0
  a=curr: qos e2e sendrecv
  a=des: qos mandatory e2e sendrecv

Example of Aggregate-based Reservation

- **Each participant deals with network access permission at its own end.**
- **Mandatory means that the session can not continue unless the required resources are definitely available.**
- **None is the initial situation and indicates that no effort to reserve resources has yet taken place.**
- **Response 580 (precondition failure)**

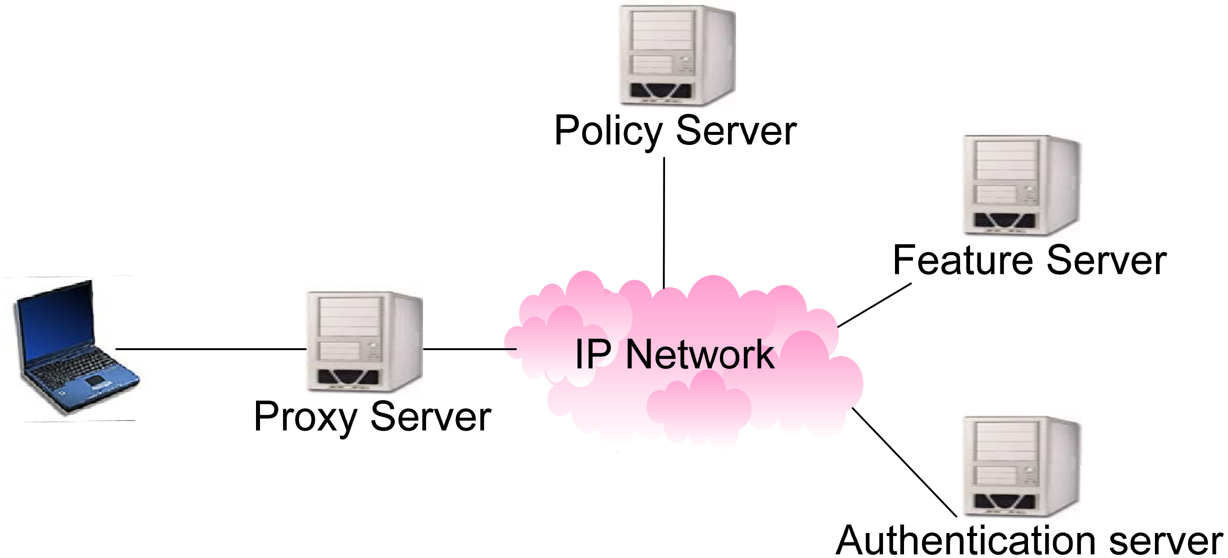


Usage of SIP for Features/Services [1/2]

- Call-transfer application (with REFER method)
- Personal Mobility through the use of registration
- One number service through forking proxy
- Call-completion services by using Retry-After: header
- To carry MIME content as well as an SDP description
  - To include a piece of text, an HTML document, an image and so on
- SIP address is a URL
  - Click-to-call applications
- The existing supplementary services in traditional telephony
  - Call waiting, call forwarding, multi-party calling, call screening

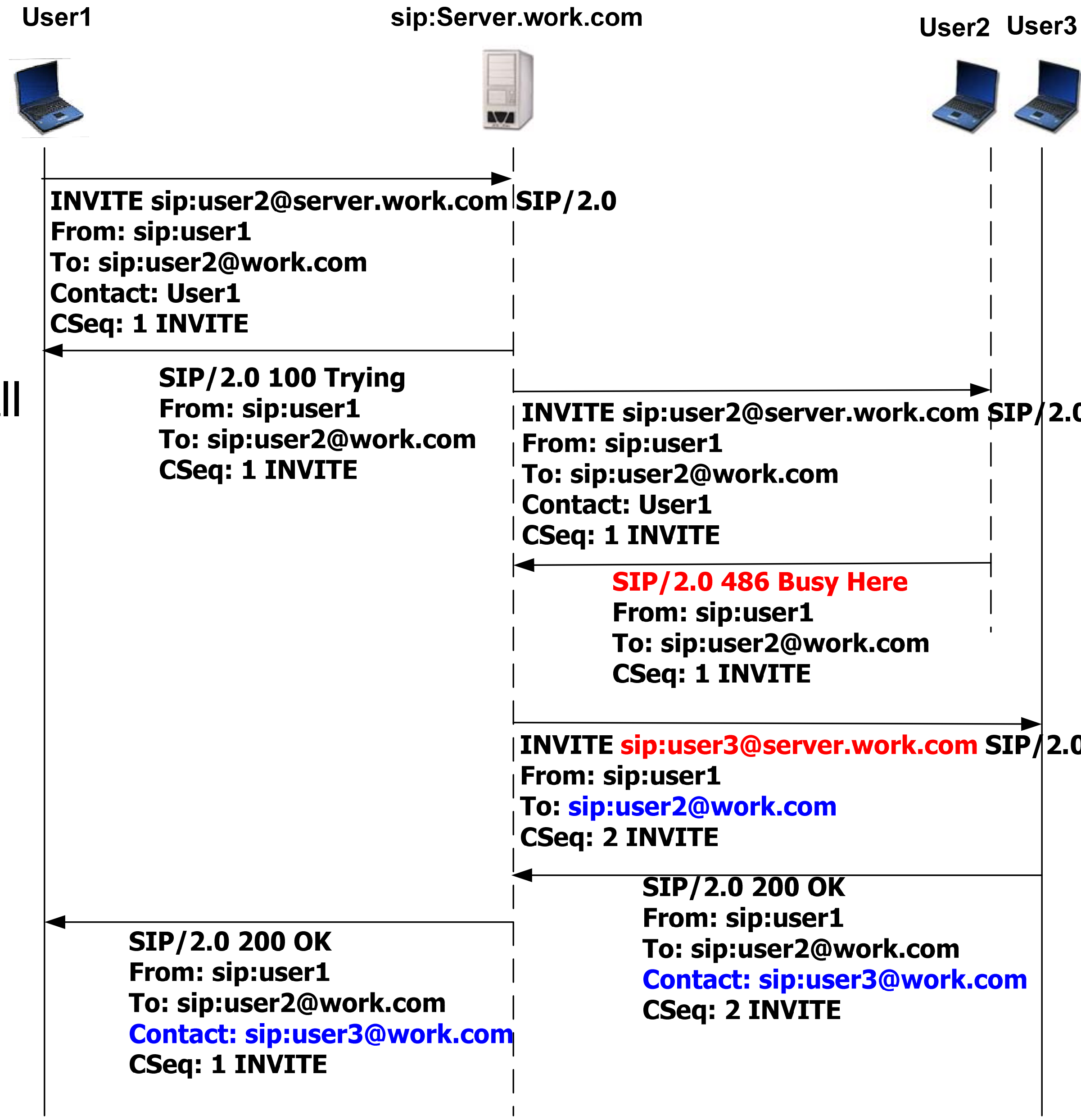
Usage of SIP for Features/Services [2/2]

- Proxy invokes various types of advanced feature logic.
  - Policy server (call-routing, QoS)
  - Authentication server
  - Use the services of an IN SCP over INAP
- The network might use the Parley Open Service Access (OSA) approach, utilizing application programming interfaces (APIs) between the nodes.



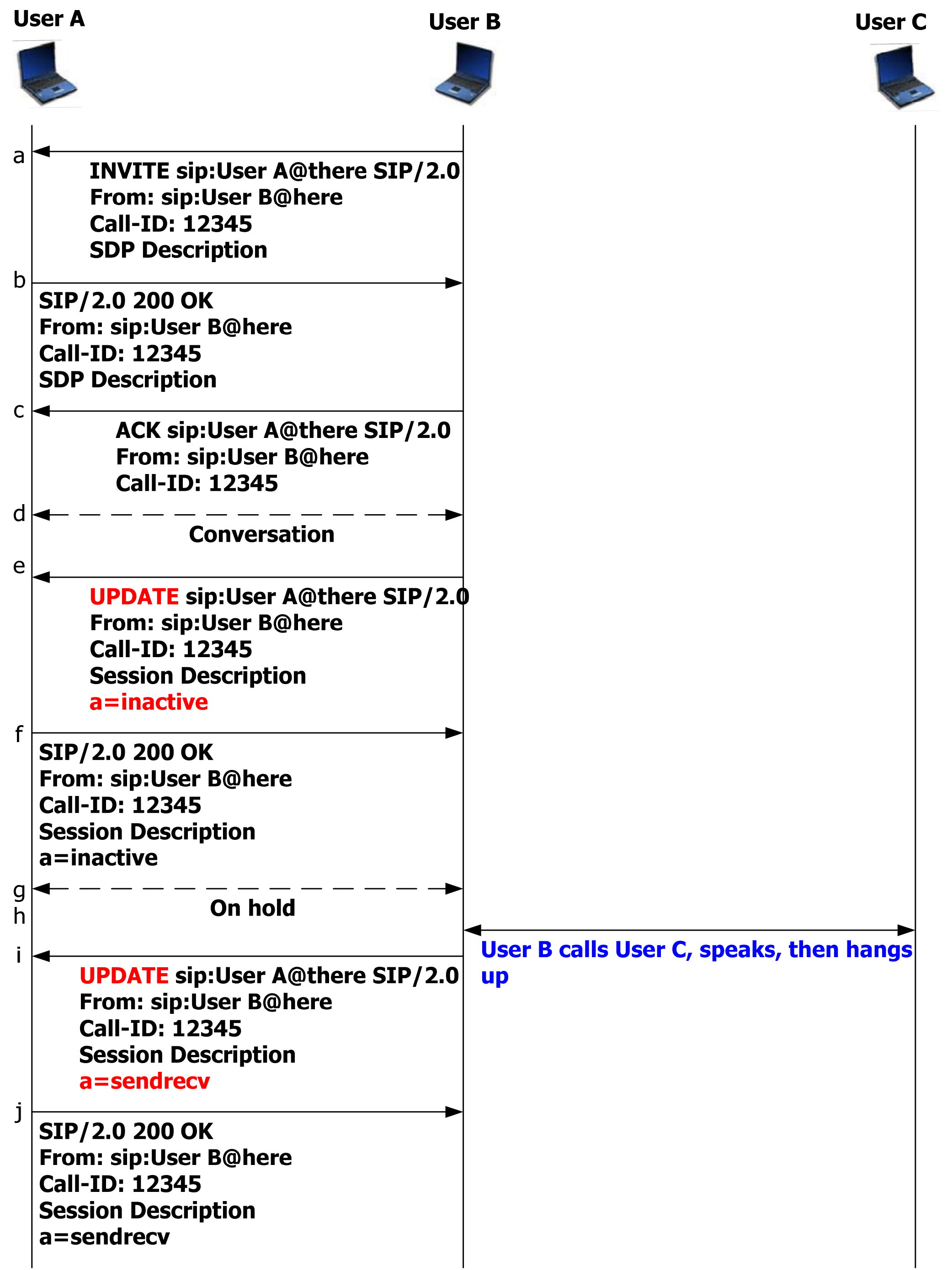
Call Forwarding

- On busy
- 486, busy here
- With the same To, User 3 can recognize that this call is a forwarded call, originally sent to User 2.
- Contact: header in 200 response
- Call-forwarding-on-no-answer
  - Timeout
  - CANCEL method



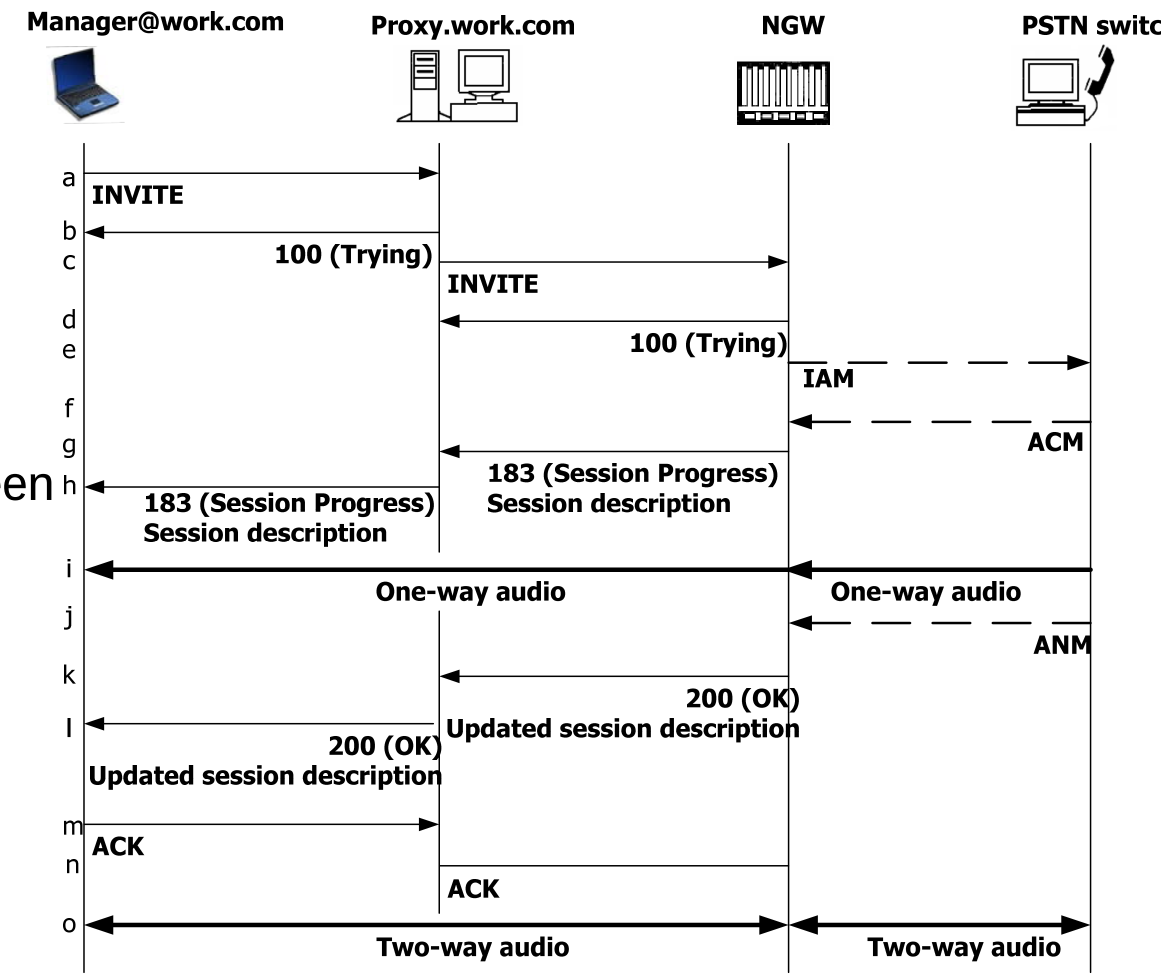
Consultation Hold

- A SIP UPDATE
- User A asks User B a question, and User B need to check with User C for the correct answer.
- If User C needs to talk to User A directly, User B could use the REFER method to transfer the call to User C.



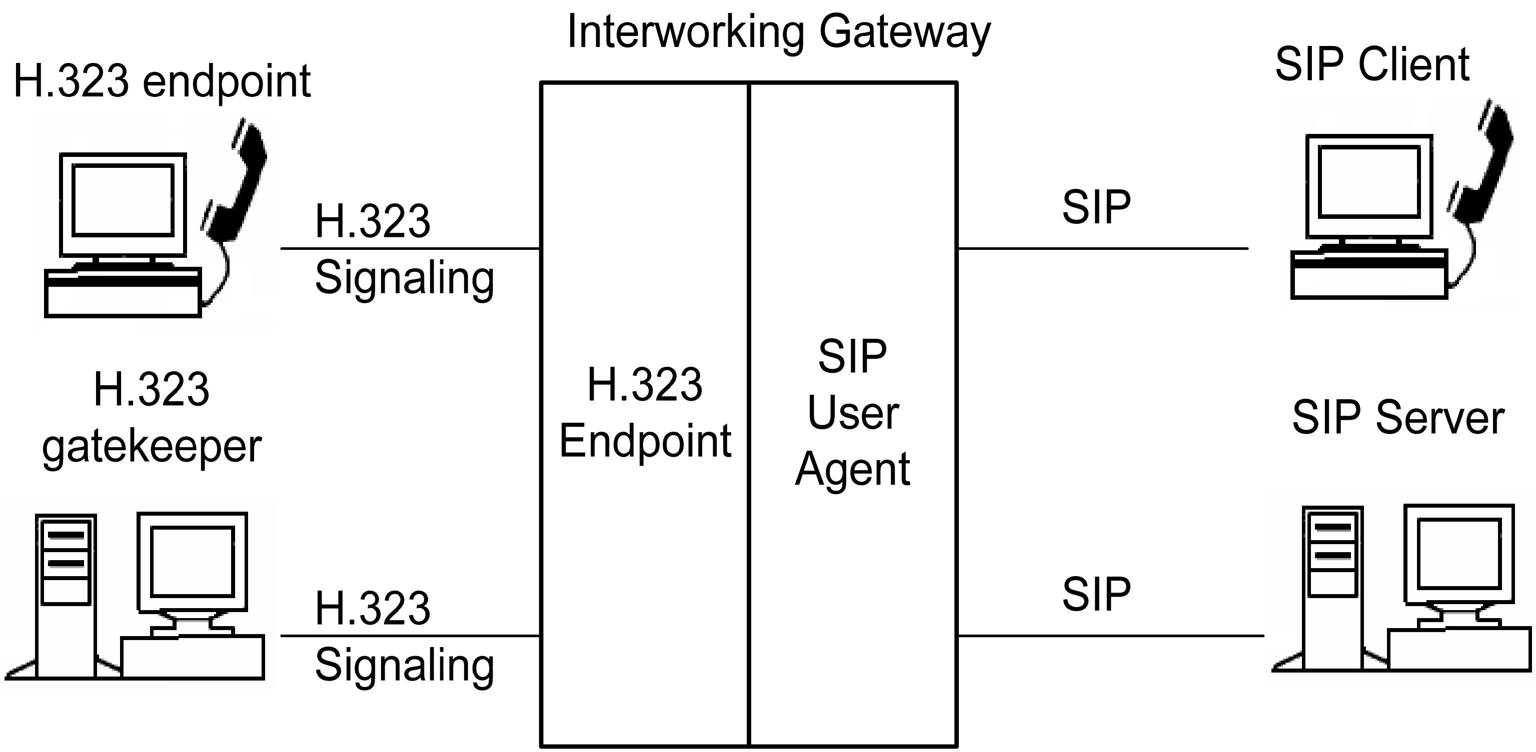
PSTN Interworking

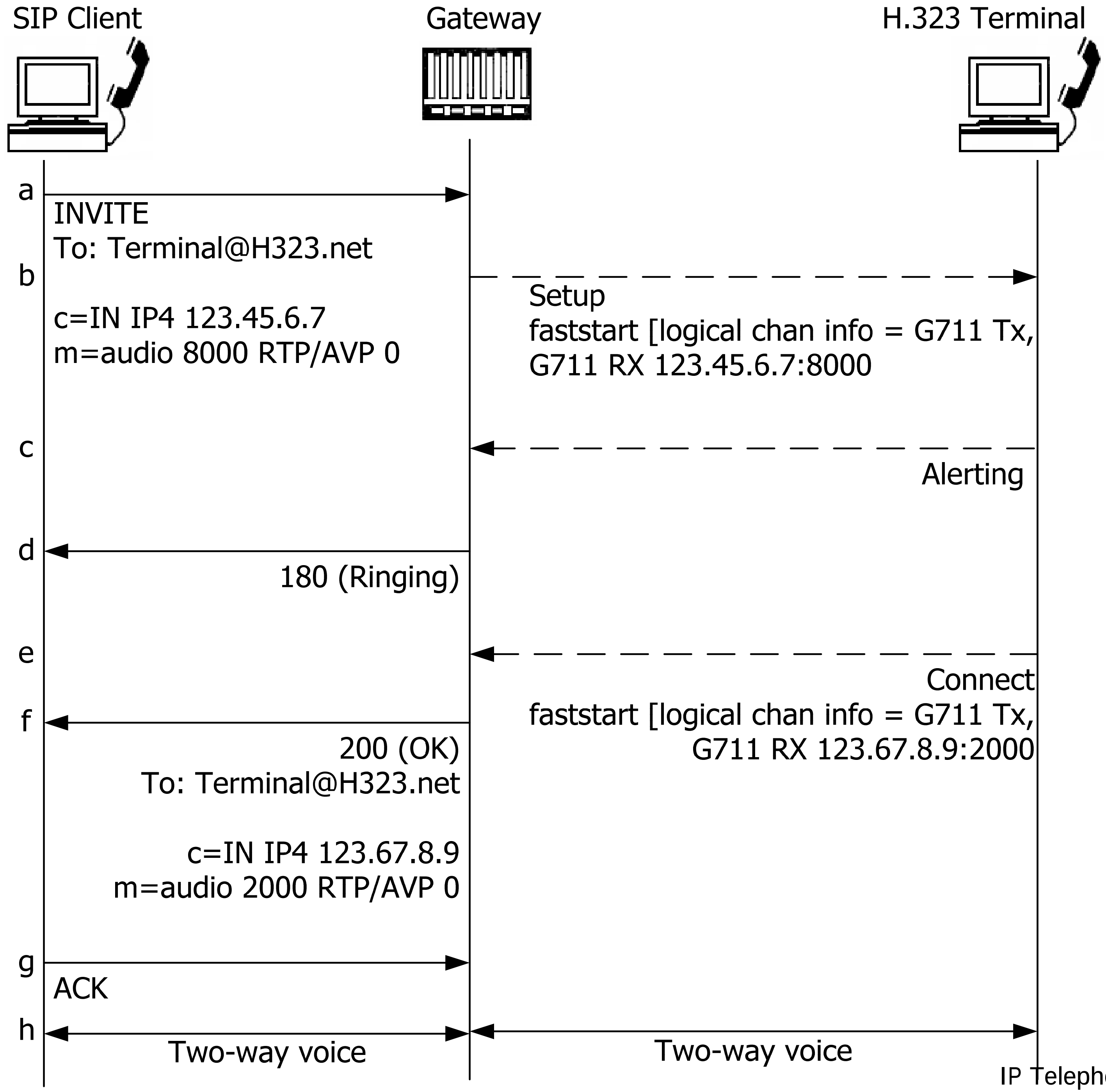
- PSTN Interworking
  - A SIP URL to a telephone number
  - A network gateway
- Seamless interworking between two different protocols is not quite easy.
  - One-to-one mapping between these protocols
- PSTN – SIP – PSTN
  - MIME media types
  - For ISUP
  - SIP for Telephony (SIP-T)
- The whole issue of interworking with SS7 is fundamental to the success of VoIP in the real world.

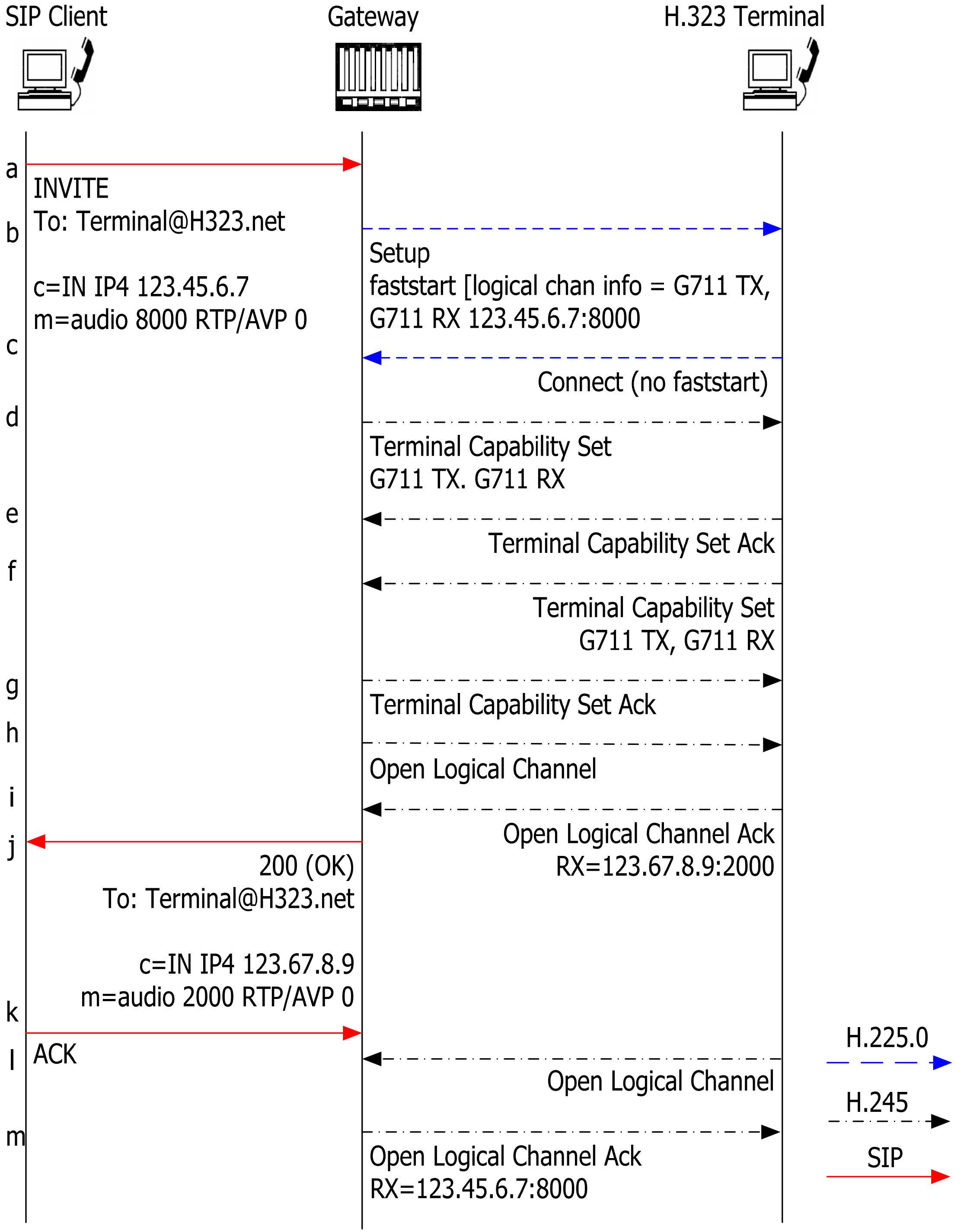


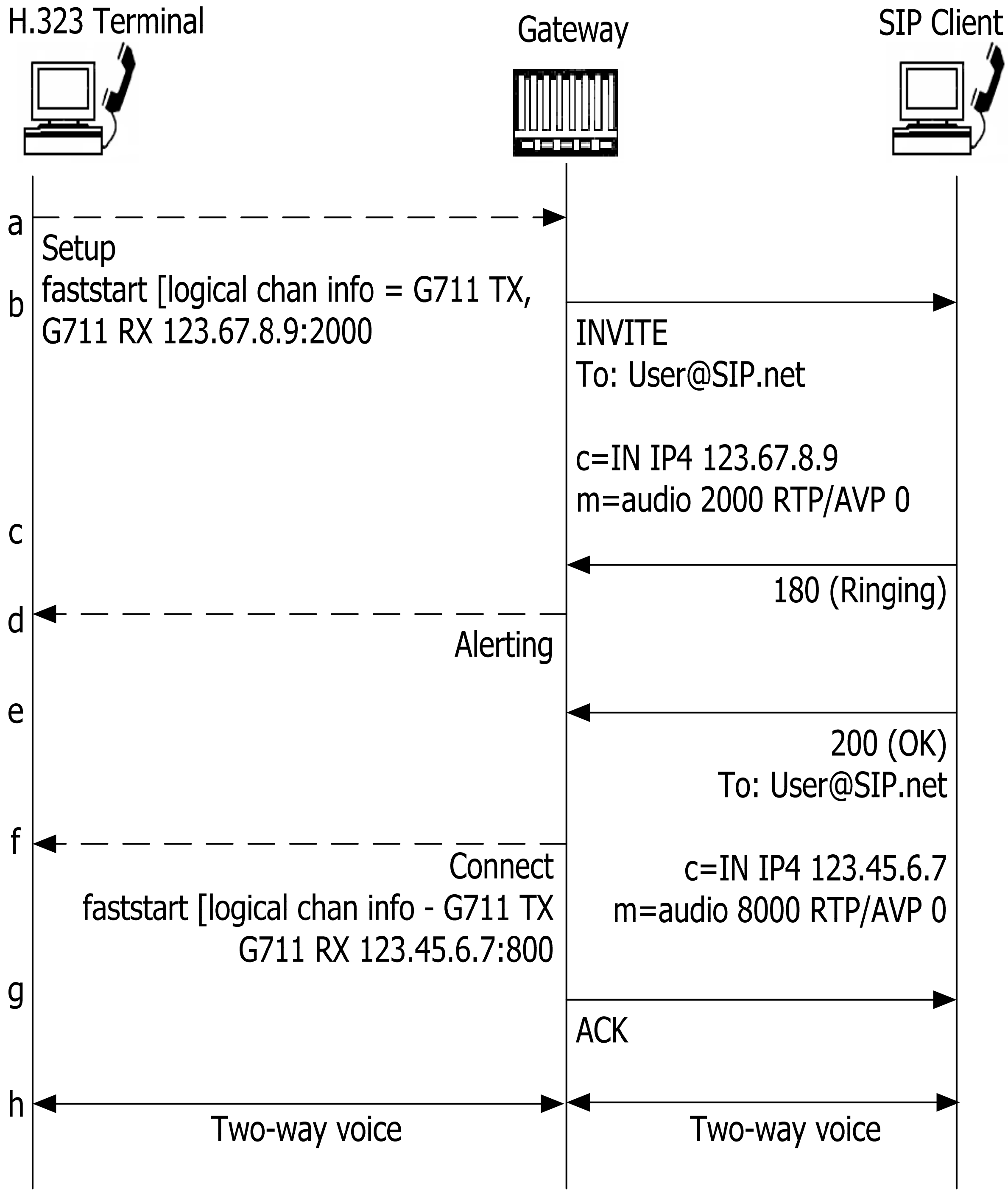
Interworking with H.323

- SIP-H.323 interworking gateway




SIP Client
Gateway
H.323 Terminal
a
INVITE
To: Terminal@H323.net
b
c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0
Setup
faststart [logical chan info = G711 Tx,
G711 RX 123.45.6.7:8000
c
Alerting
d
180 (Ringing)
e
Connect
faststart [logical chan info = G711 Tx,
G711 RX 123.67.8.9:2000
f
200 (OK)
To: Terminal@H323.net
c=IN IP4 123.67.8.9
m=audio 2000 RTP/AVP 0
g
ACK
h
Two-way voice
Two-way voice


SIP Client
Gateway
H.323 Terminal
a
INVITE
To: Terminal@H323.net
b
c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0
Setup
faststart [logical chan info = G711 TX,
G711 RX 123.45.6.7:8000
c
Connect (no faststart)
d
Terminal Capability Set
G711 TX. G711 RX
e
Terminal Capability Set Ack
f
Terminal Capability Set
G711 TX, G711 RX
g
Terminal Capability Set Ack
h
Open Logical Channel
i
Open Logical Channel Ack
RX=123.67.8.9:2000
j
200 (OK)
To: Terminal@H323.net
c=IN IP4 123.67.8.9
m=audio 2000 RTP/AVP 0
k
ACK
l
Open Logical Channel
m
Open Logical Channel Ack
RX=123.45.6.7:8000
H.225.0
H.245
SIP



H.323 Terminal
Gateway
SIP Client
a
Setup
faststart [logical chan info = G711 TX,
G711 RX 123.67.8.9:2000
b
INVITE
To: User@SIP.net
c=IN IP4 123.67.8.9
m=audio 2000 RTP/AVP 0
c
180 (Ringing)
d
Alerting
e
200 (OK)
To: User@SIP.net
c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0
f
Connect
faststart [logical chan info - G711 TX
G711 RX 123.45.6.7:800
g
ACK
h
Two-way voice
Two-way voice

Summary

- The future for signaling in VoIP networks
  - Simple, yet flexible
  - Easier to implement
  - Fit well with the media gateway control protocols
    - Coexisting with PSTN
- SIP is the protocol of choice for the evolution of third-generation wireless networks.
  - SIP-based mobile devices will become available.
  - SIP-based network elements will be introduced within mobile networks.