Session Initiation Protocol (SIP)

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

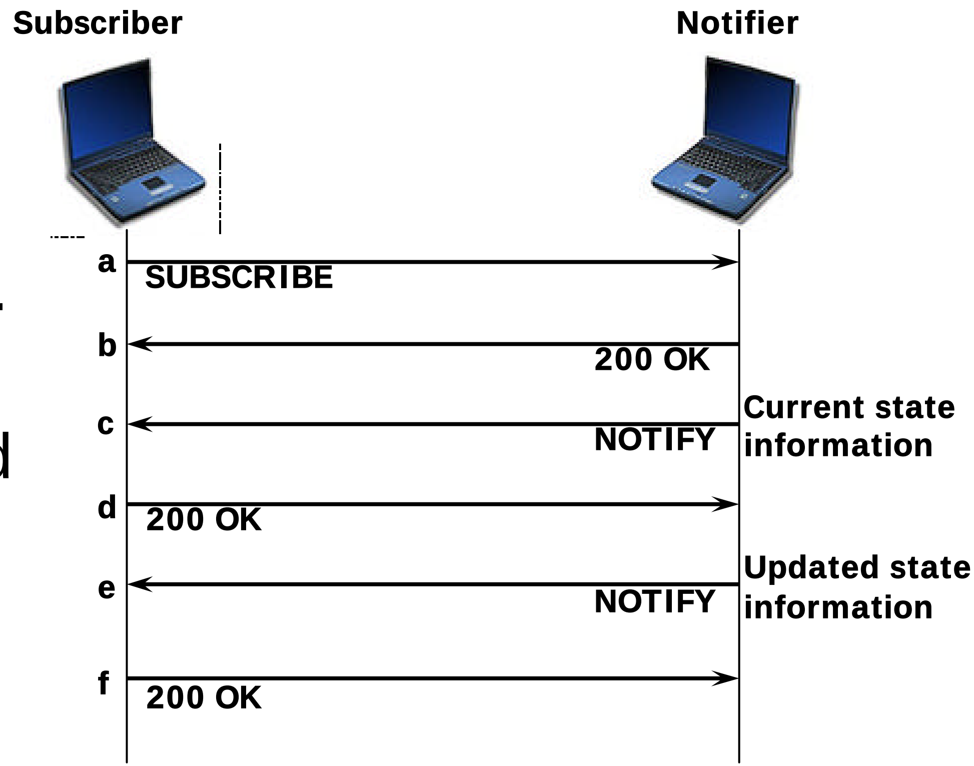
- The Base RFC 2543
  - The Require: Header
    - In request (client ->server)
      - A client indicates that a server must support certain extension.
  - The Unsupported Header
    - In response (server -> client)
      - 420 (bad extension)
  - A cumbersome way of determining what extensions a server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

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.
  - A MESSAGE request does not establish a SIP dialog.

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

a

b

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?

MESSAGE sip:Collins@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>

e

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?

f

MESSAGE sip:Manager@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?

g

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

h

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.co

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=z9hG4bKxyz
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.co


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=z9hG4bKxyz
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=z9hG4bKxyz
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
To: Joe<sip:Joe@work.com>; tag=67890
From: Mary<sip:Mary@work.com>; tag=123456
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

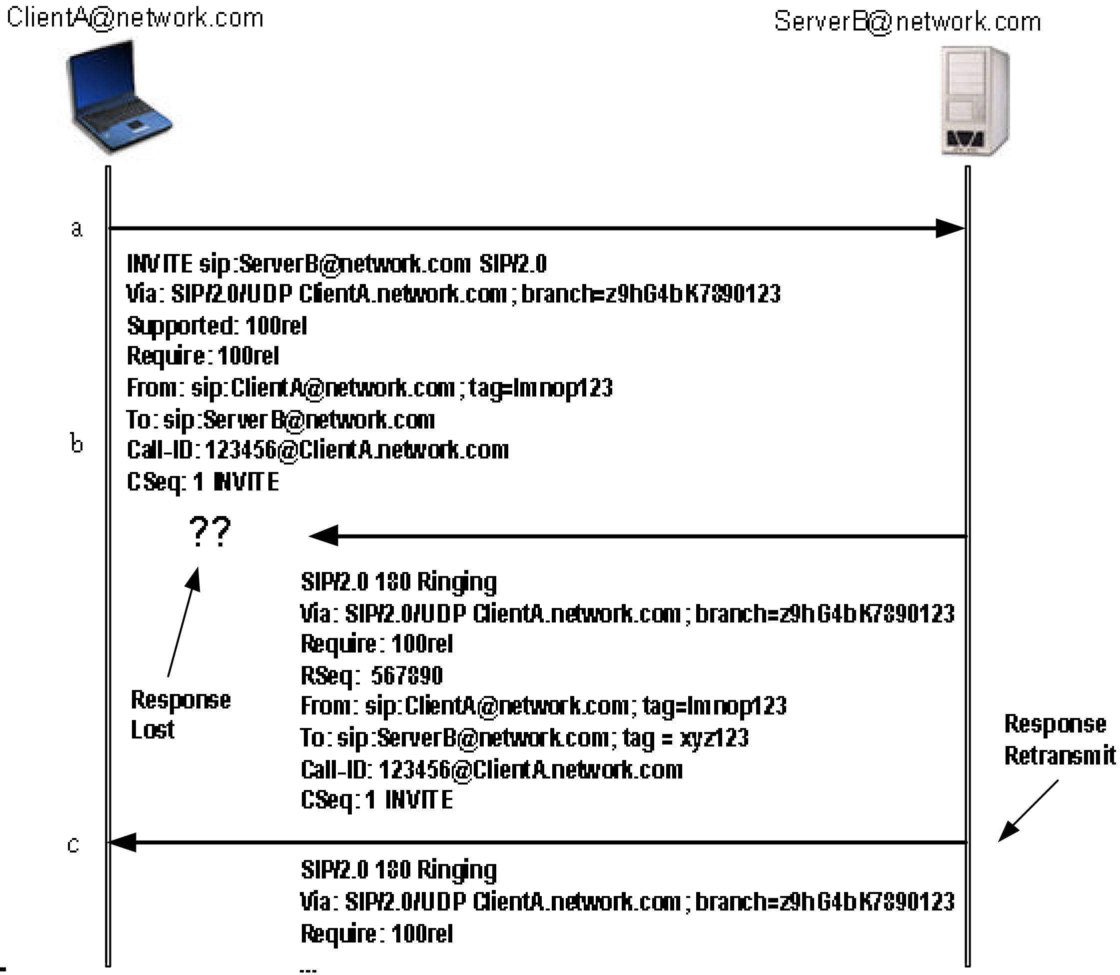
SIP/2.0 200 OK
Via: SIP/2.0/UDP station2.work.com; branch=z9hG4bK123
To: Joe<sip:Joe@work.com>; tag=67890
From: Mary<sip:Mary@work.com>; tag=123456
Call-ID: 123456@station1.work.com
CSeq: 124 NOTIFY
Content-Length: 0

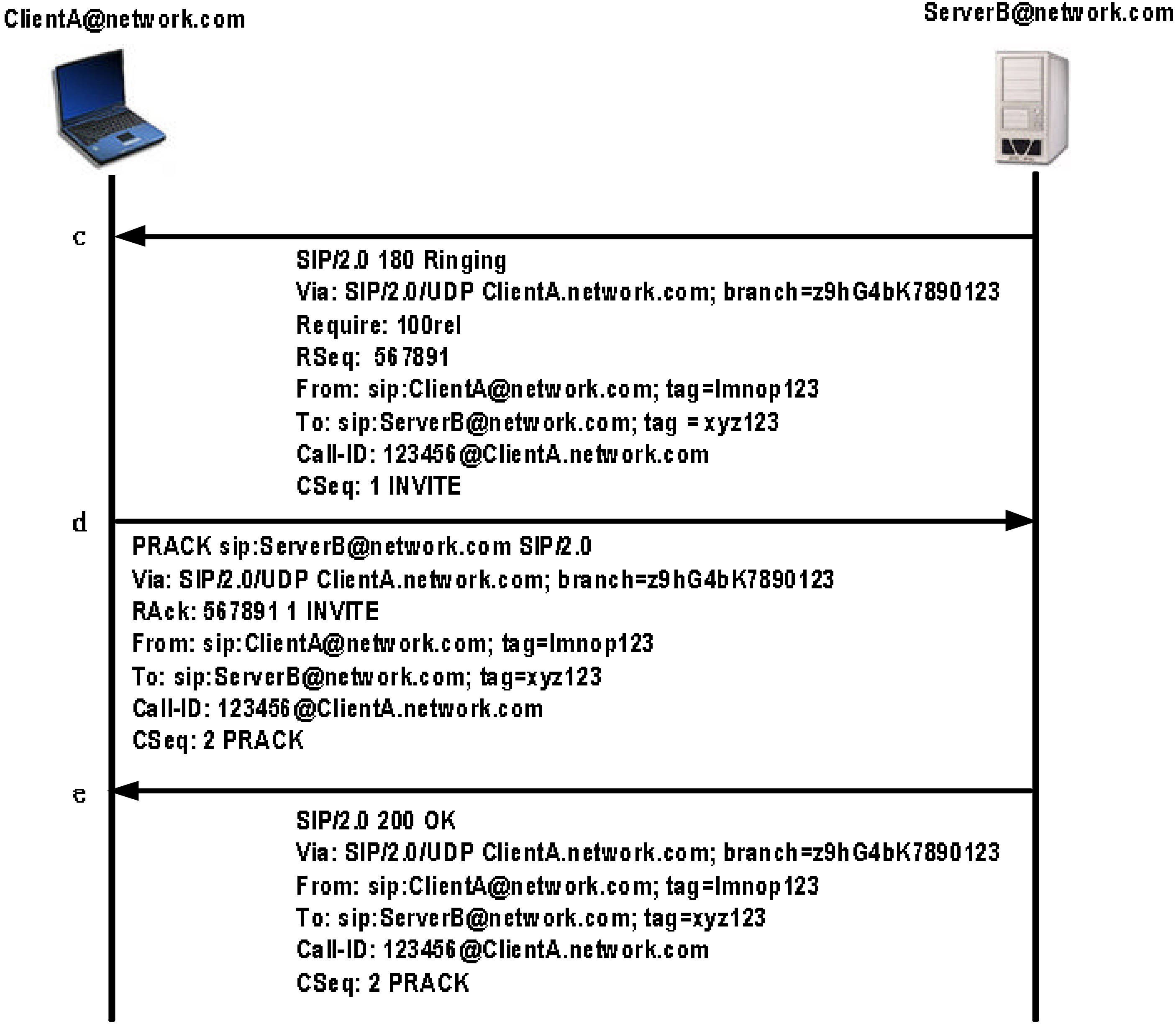
Reliability of Provisional Responses [1/2]

- 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

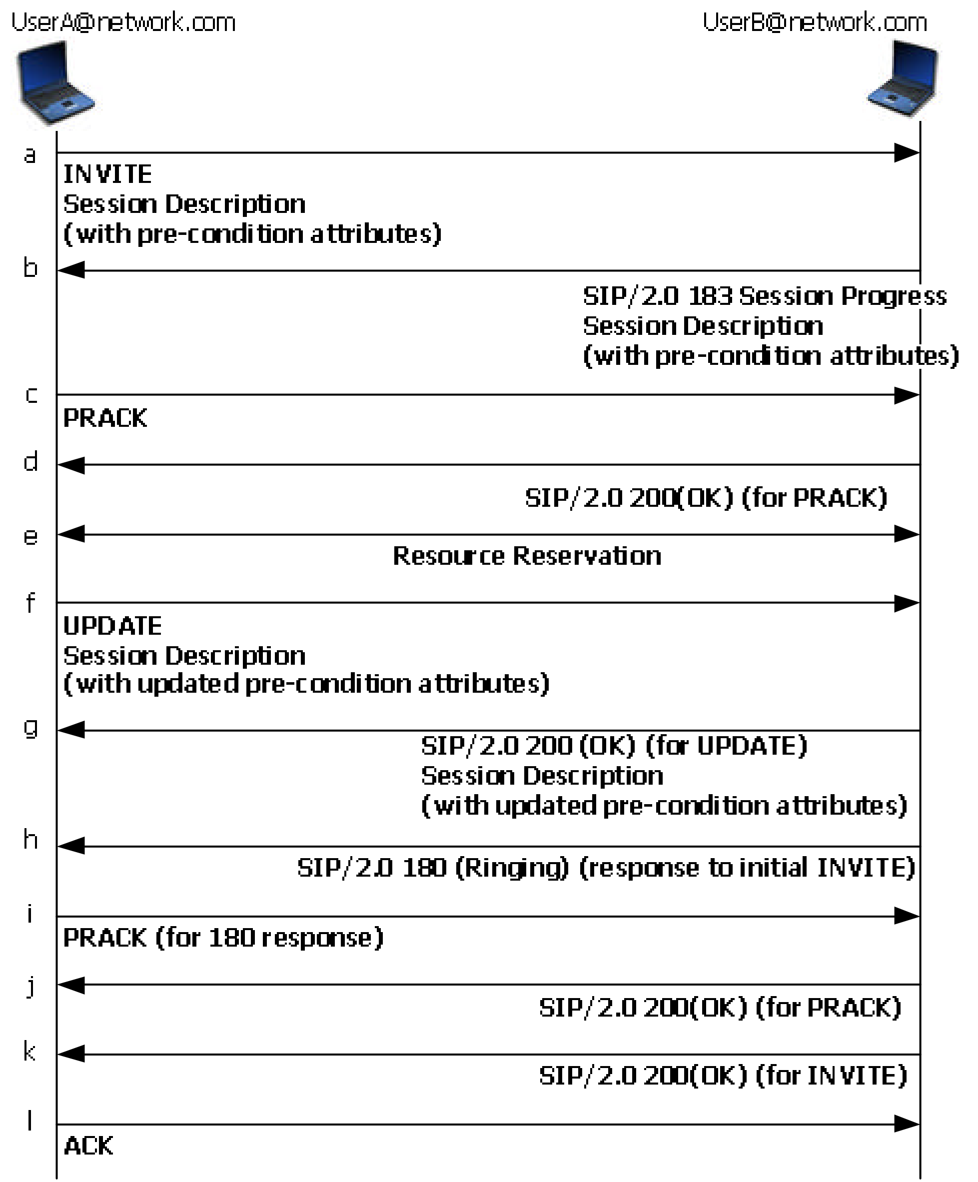
- To enable the modification of session information before a final response to an INVITE is received
  - The dialog is in the early state (An INVITE that receives a 183 response that includes a message body)
    - The message body might establish a media stream from callee to caller for sending a ring 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

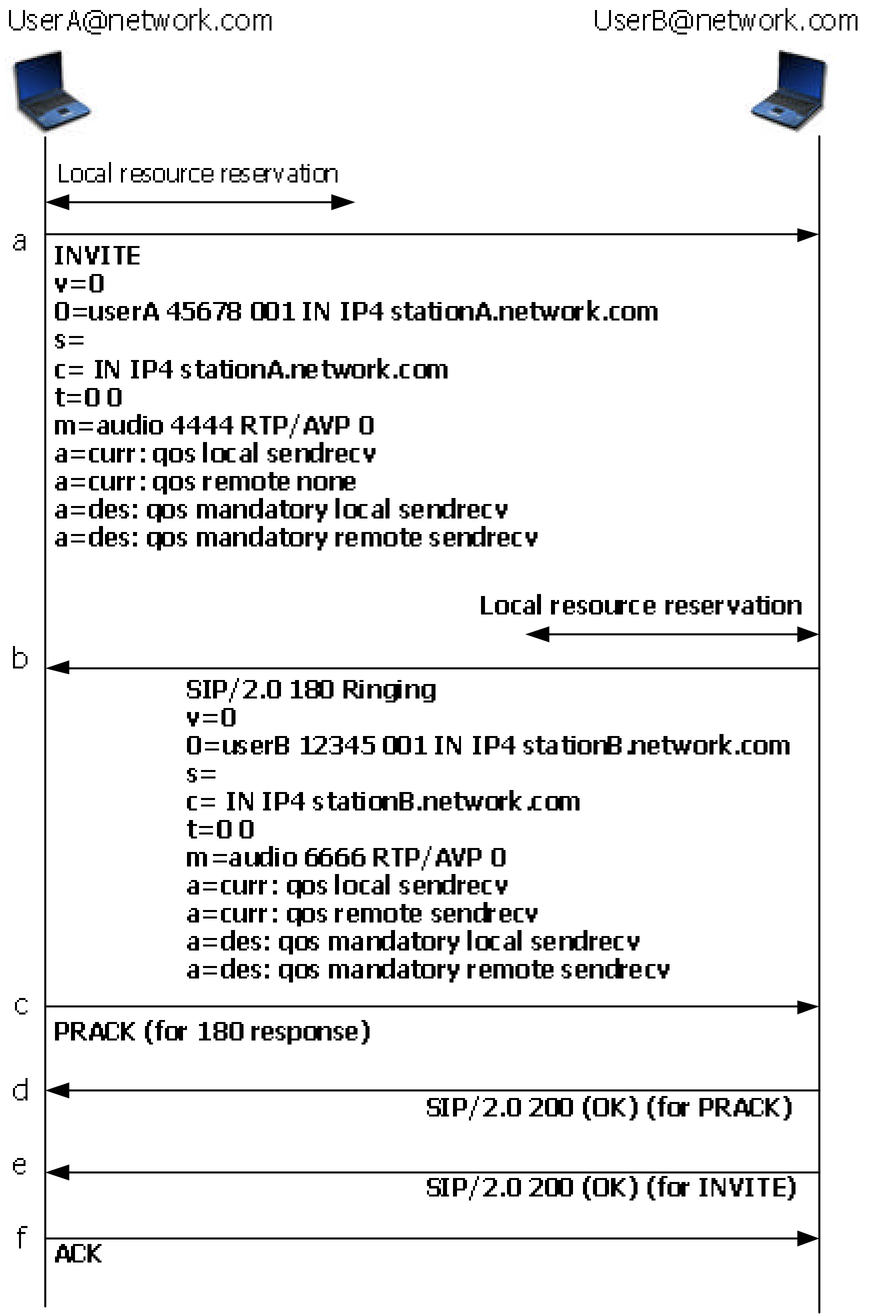
Example of e2e Resource Reservation [2/2]

- 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)

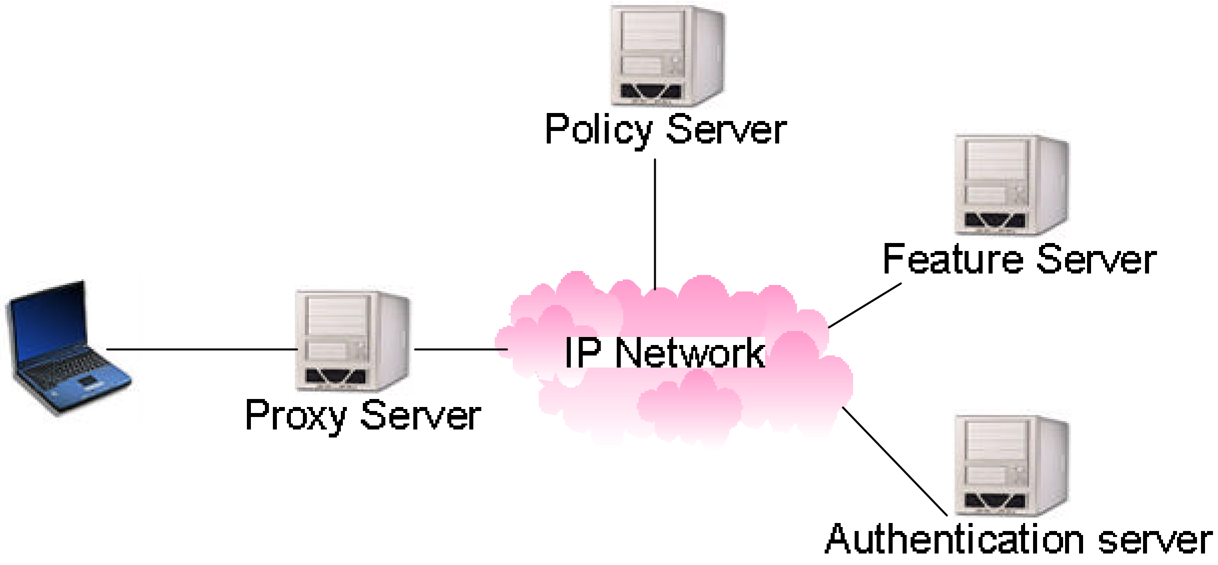


Usage of SIP for Features/Services [1/2]

- 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

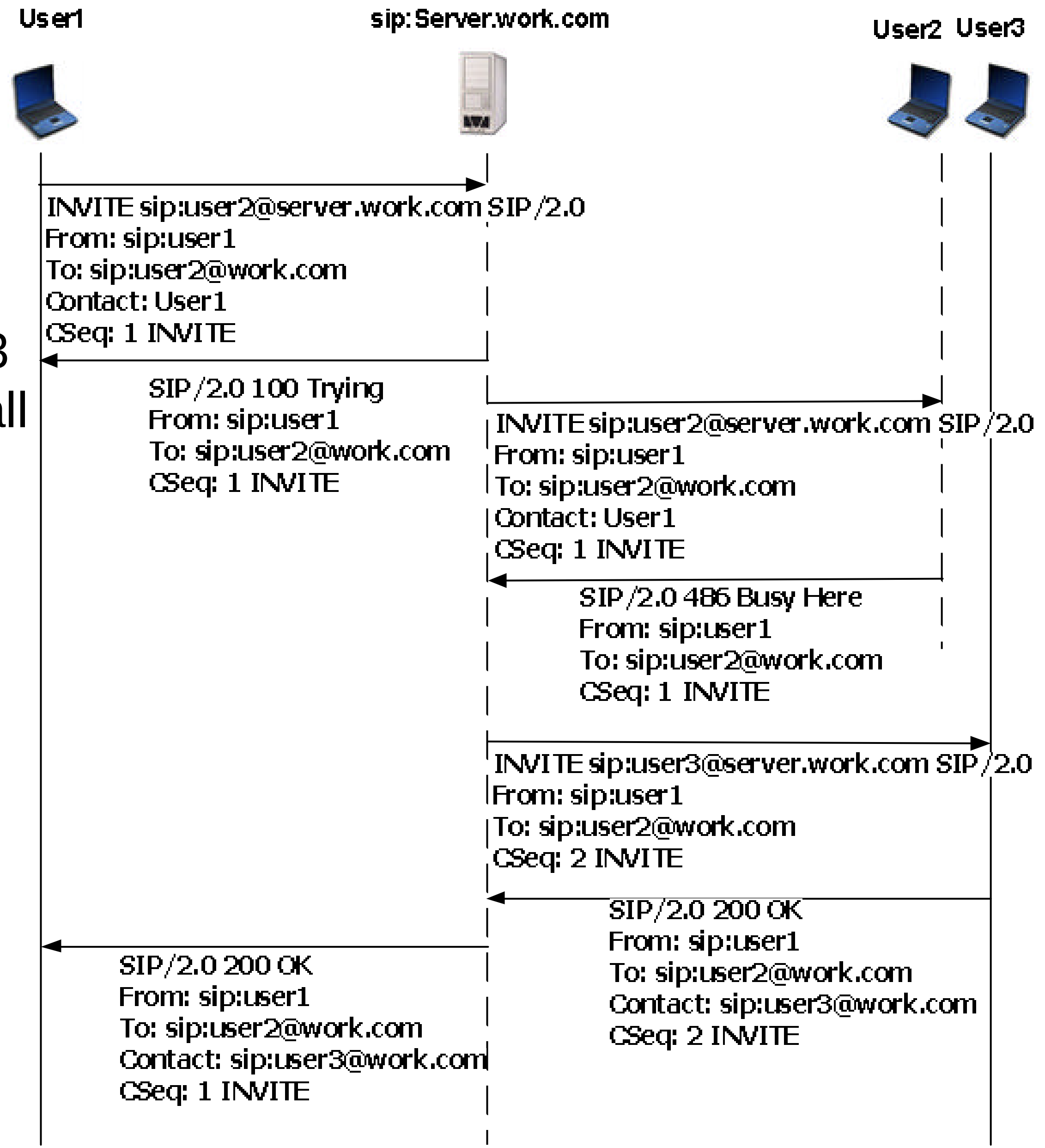
Usage of SIP for Features/Services [2/2]

- 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.



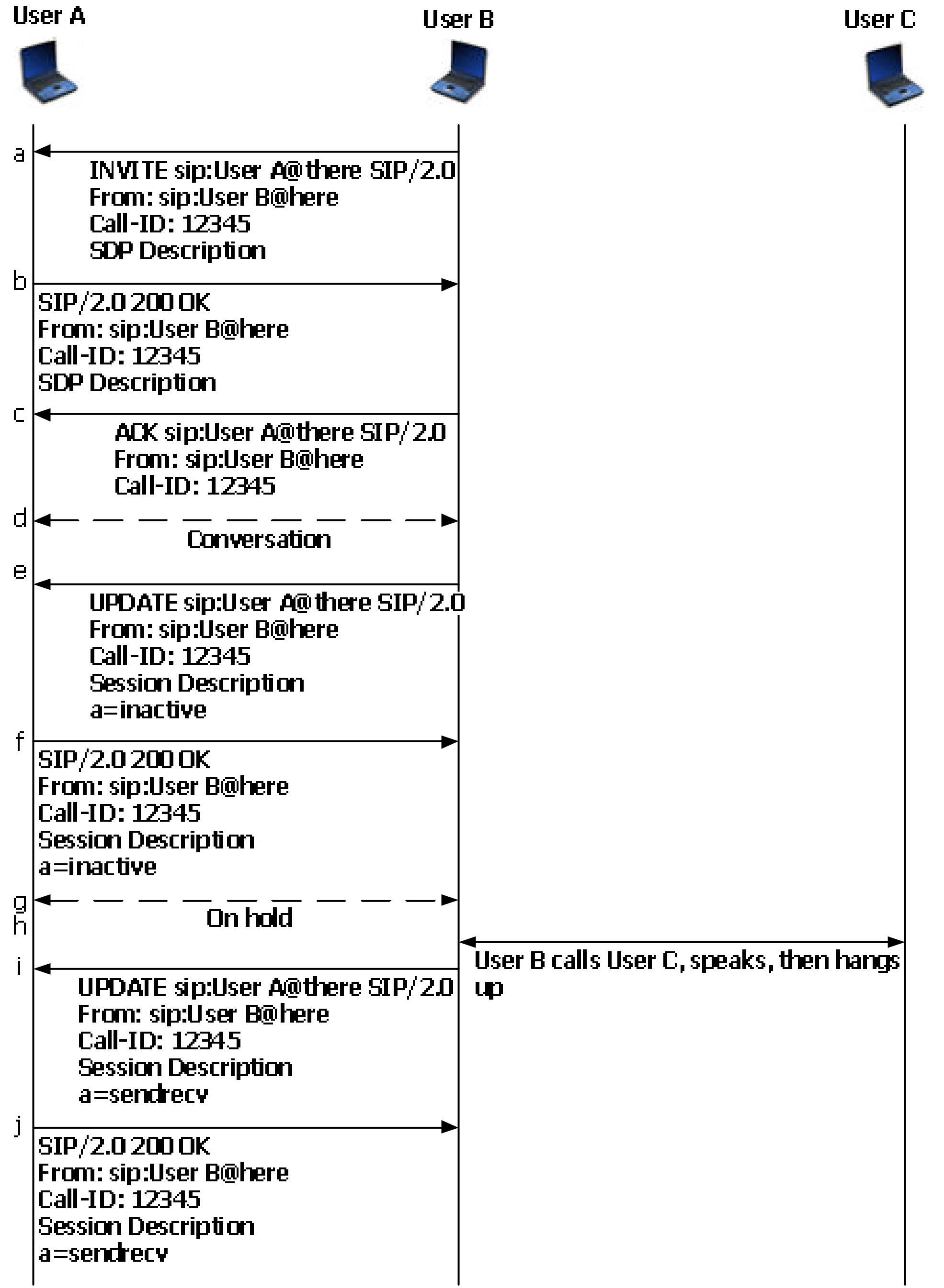
Call Forwarding

- 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



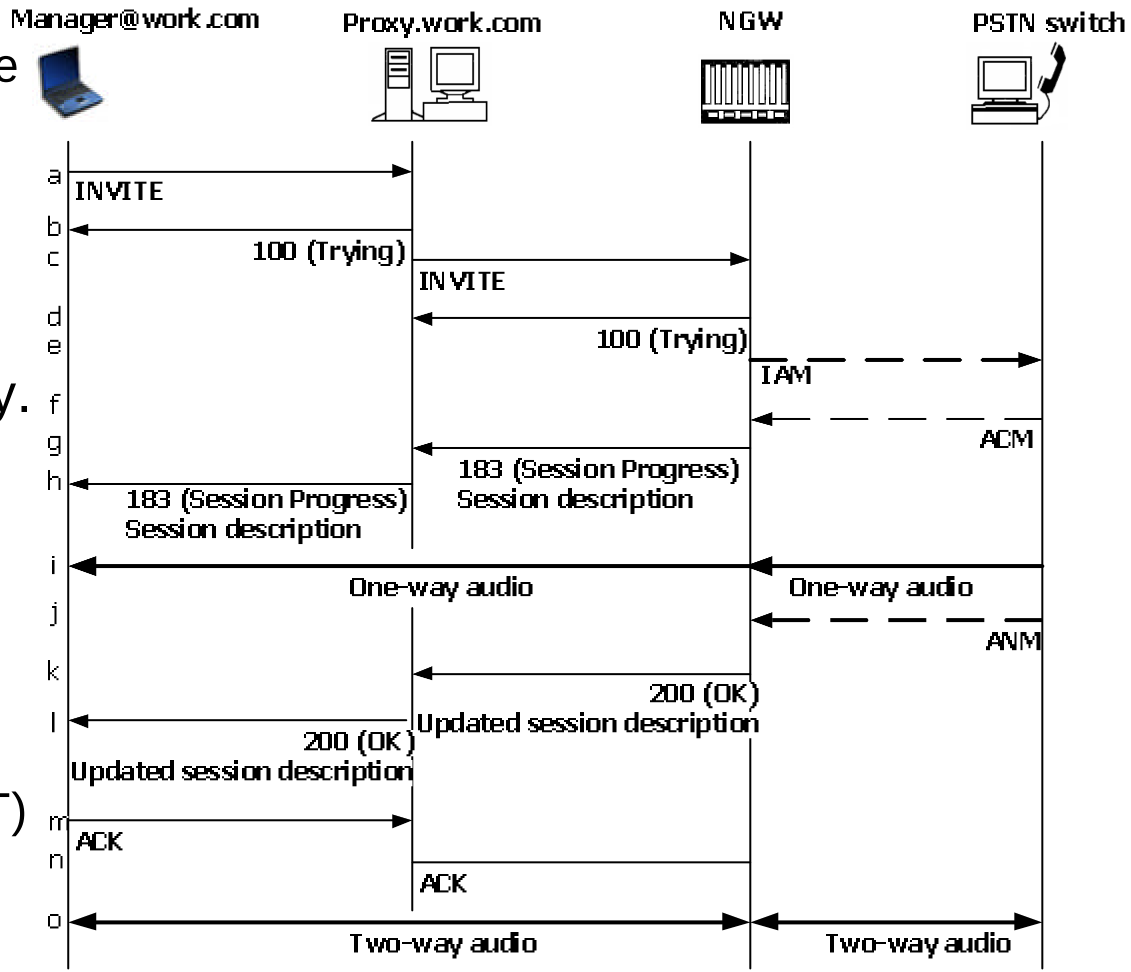
Consultation Hold

- 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 talke to User A directly, User B could use the REFER method to transfer the call to User C.



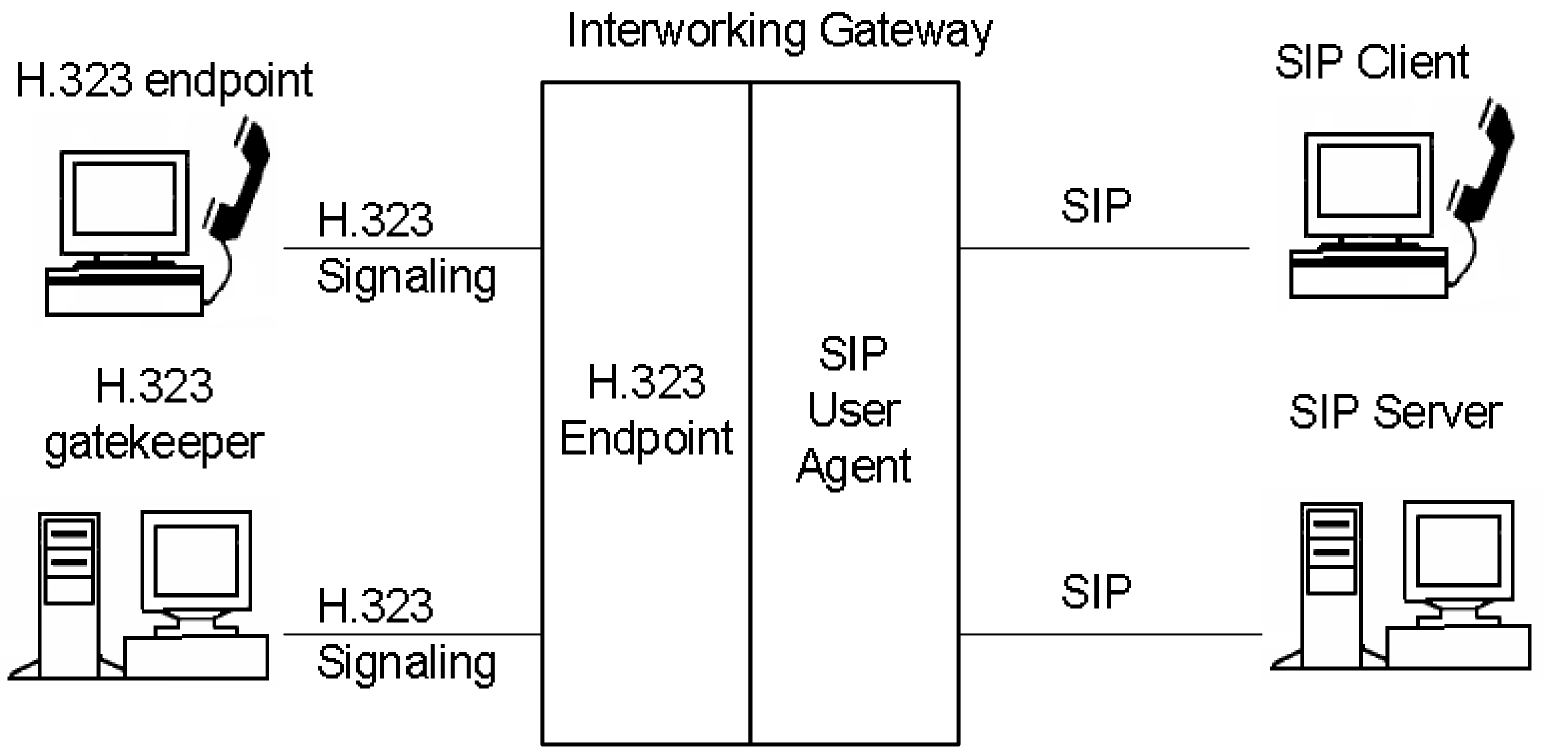
PSTN Interworking

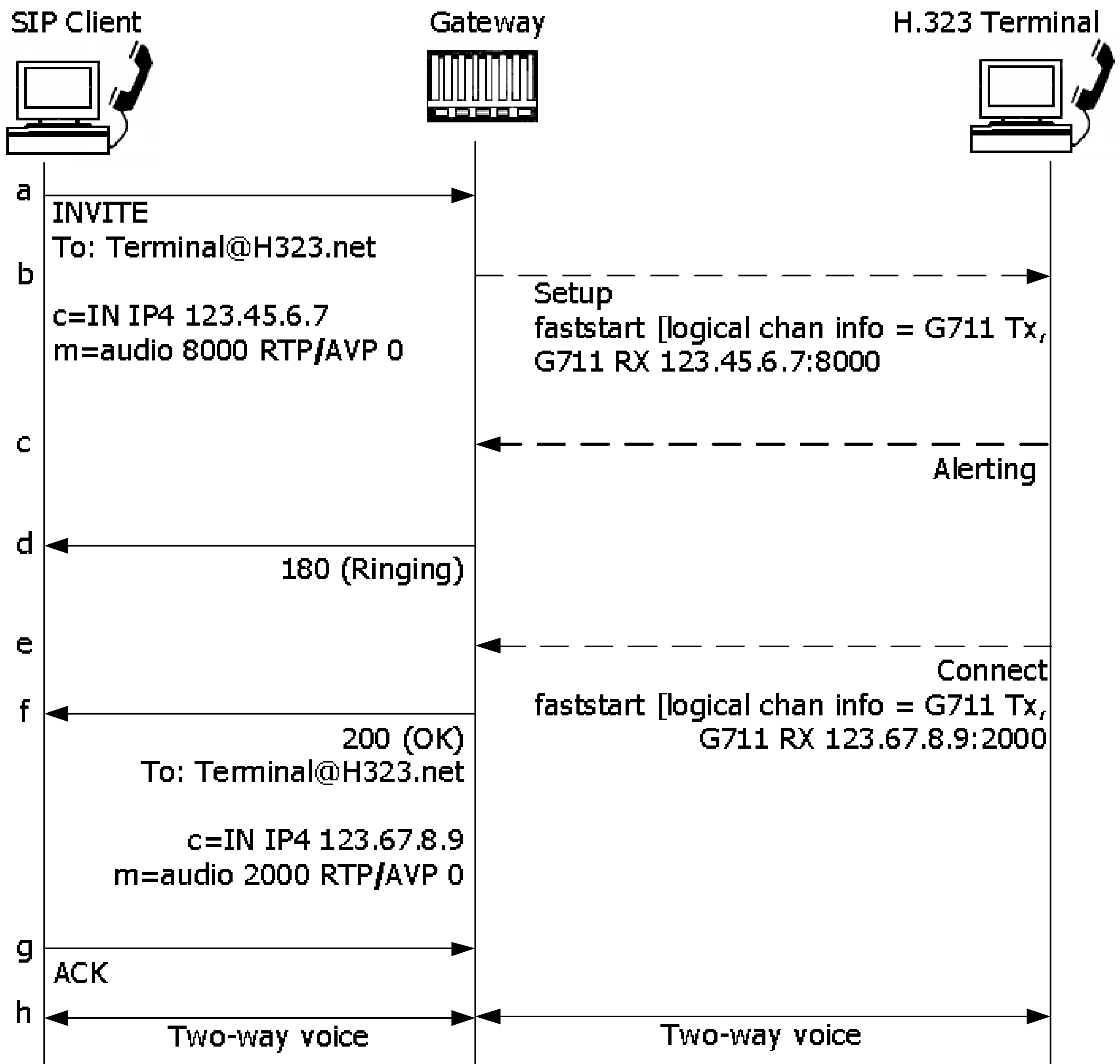
- PSTN Interworking
  - A SIP URL to a telephone number
  - A network gateway
- Seamless interworking between two different protocols is not quite easy.
  - One-to-one mapping between these protocols
- PSTN – SIP – PSTN
  - MIME media types
  - For ISUP
  - SIP for Telephony (SIP-T)
- The whole issue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

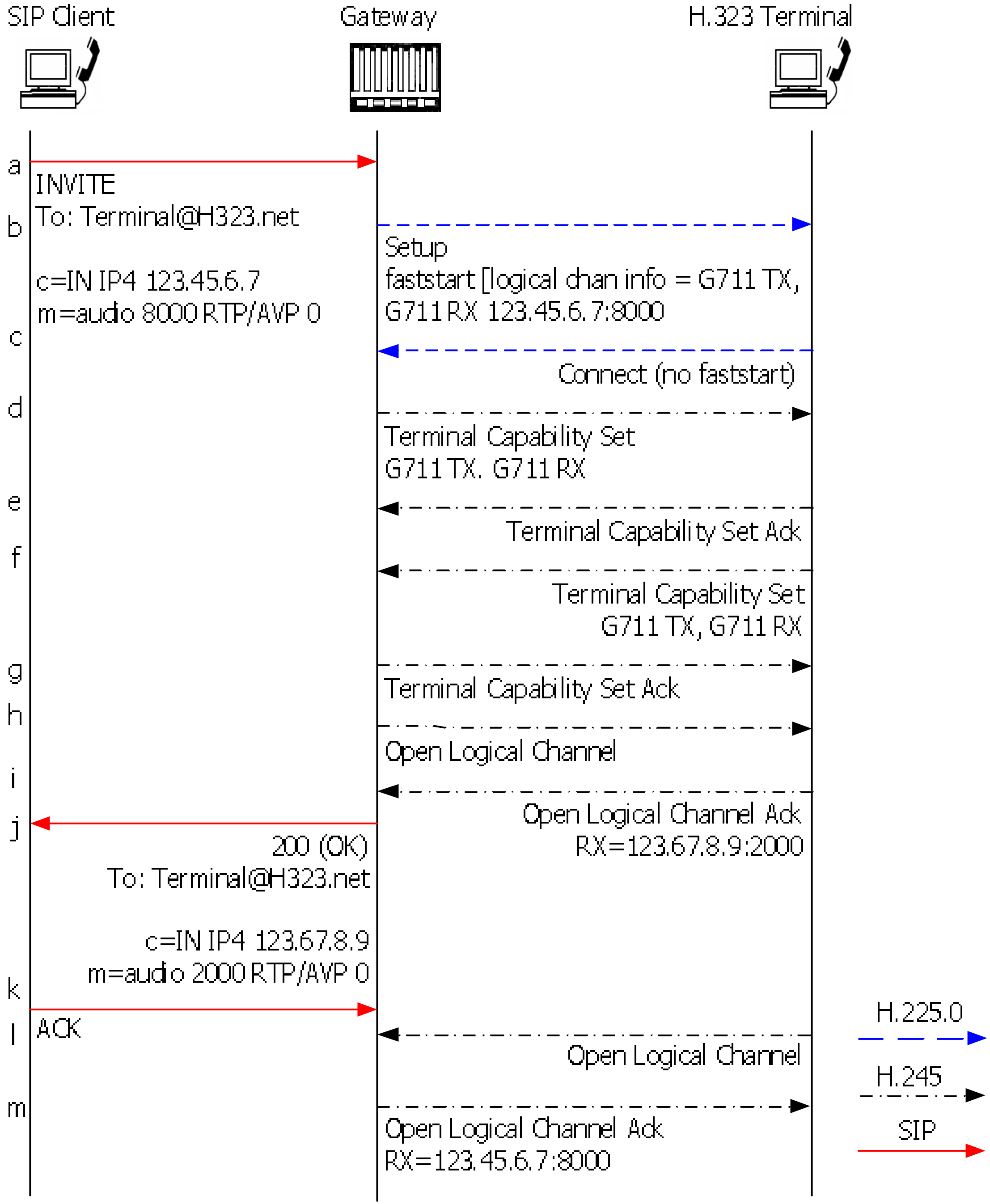


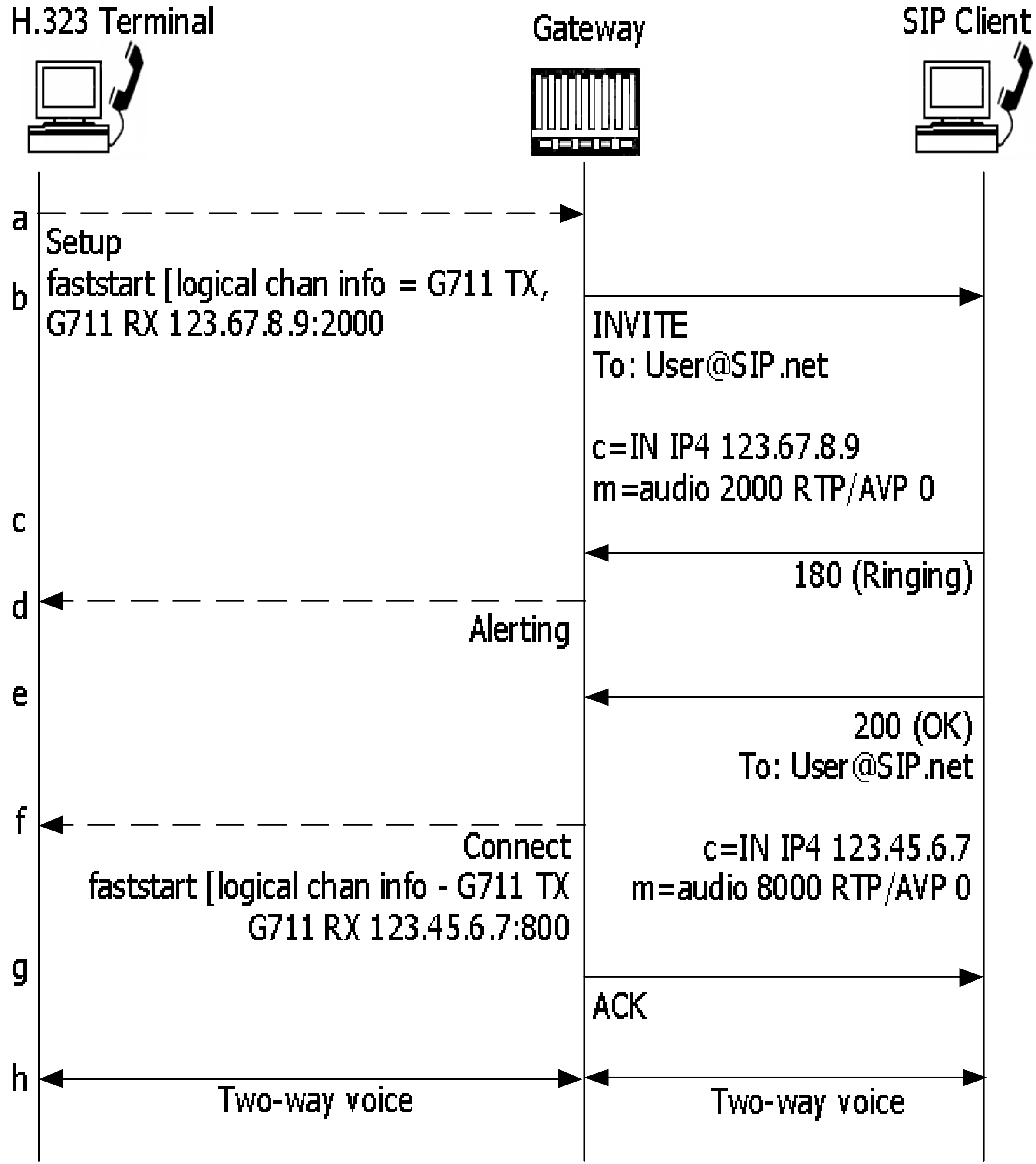
Interworking with H.323

- 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.