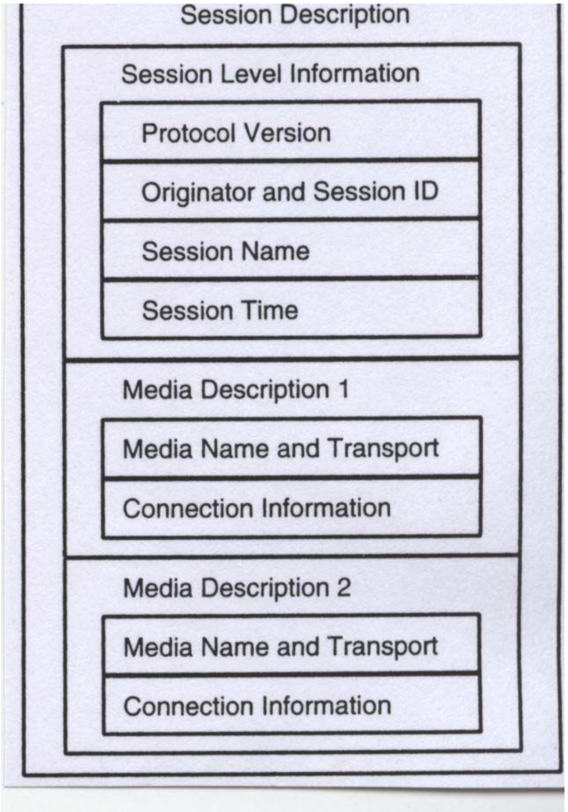
The Session Description Protocol

- The Most Common Message Body
  - Be session information describing the media to be exchanged between the parties
  - SDP, RFC 2327 (initial publication)
- 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, 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

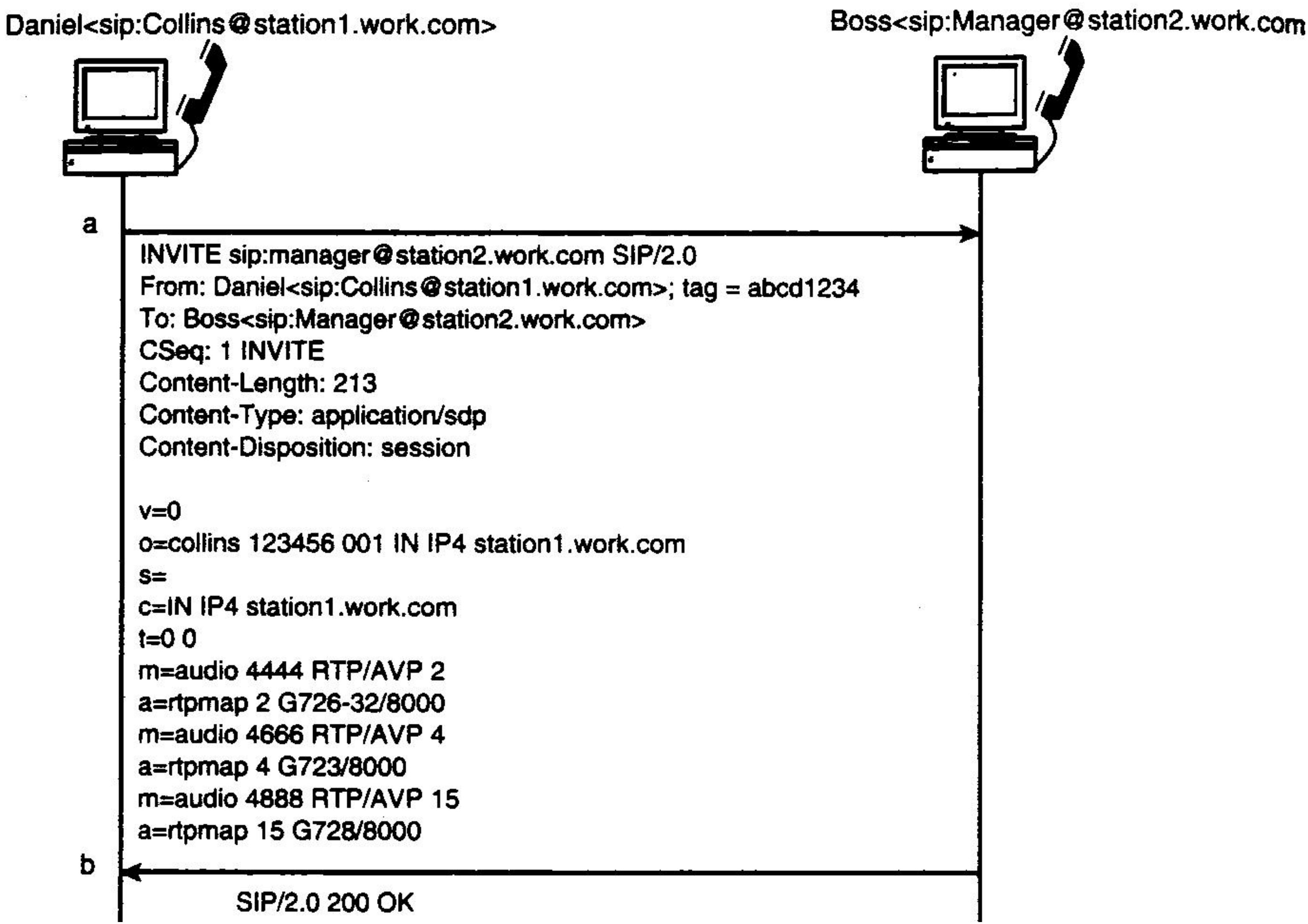
    a=rtpmap 2 G726-32/8000

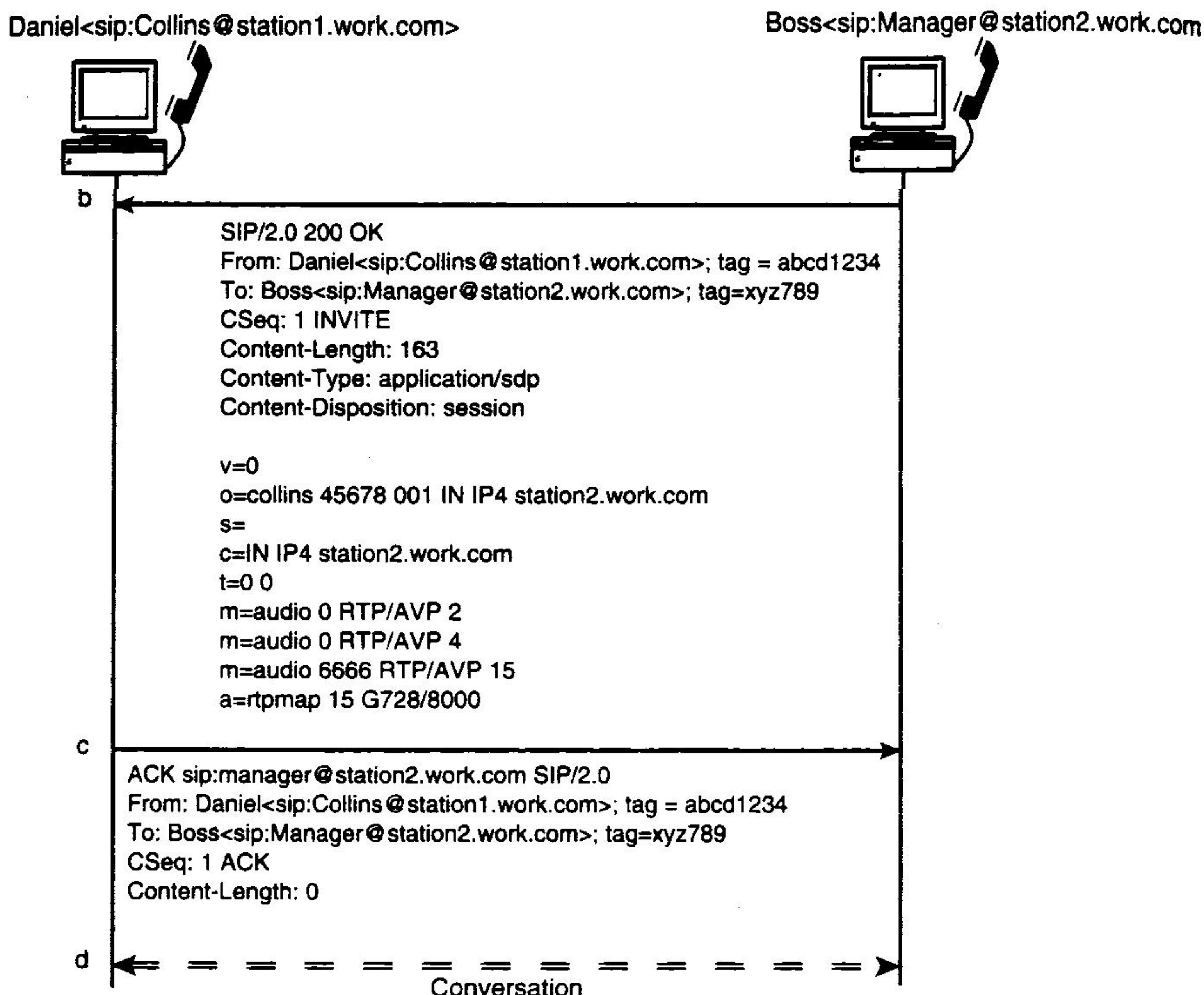
    a=rtpmap 4 G723/8000

    a=rtpmap 15 G728/8000
  - CONNECT

    m=audio 6666 RTP/AVP 15

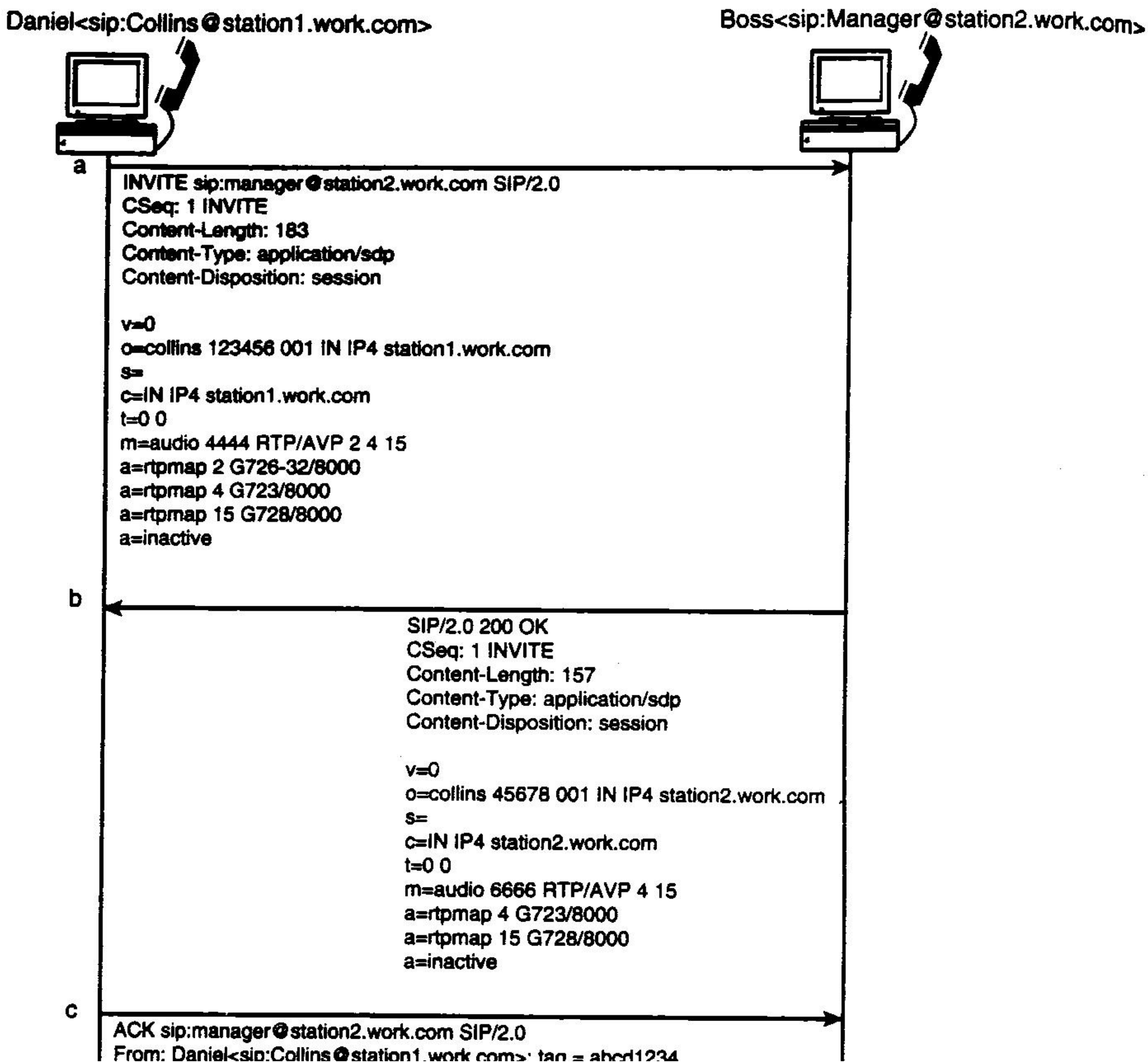
    a=rtpmap 15 G728/8000


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
c
ACK sip:manager@station2.work.com SIP/2.0
From: Daniel<sip:Collins@station1.work.com>; tag = abcd1234

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

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

a=inactive

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

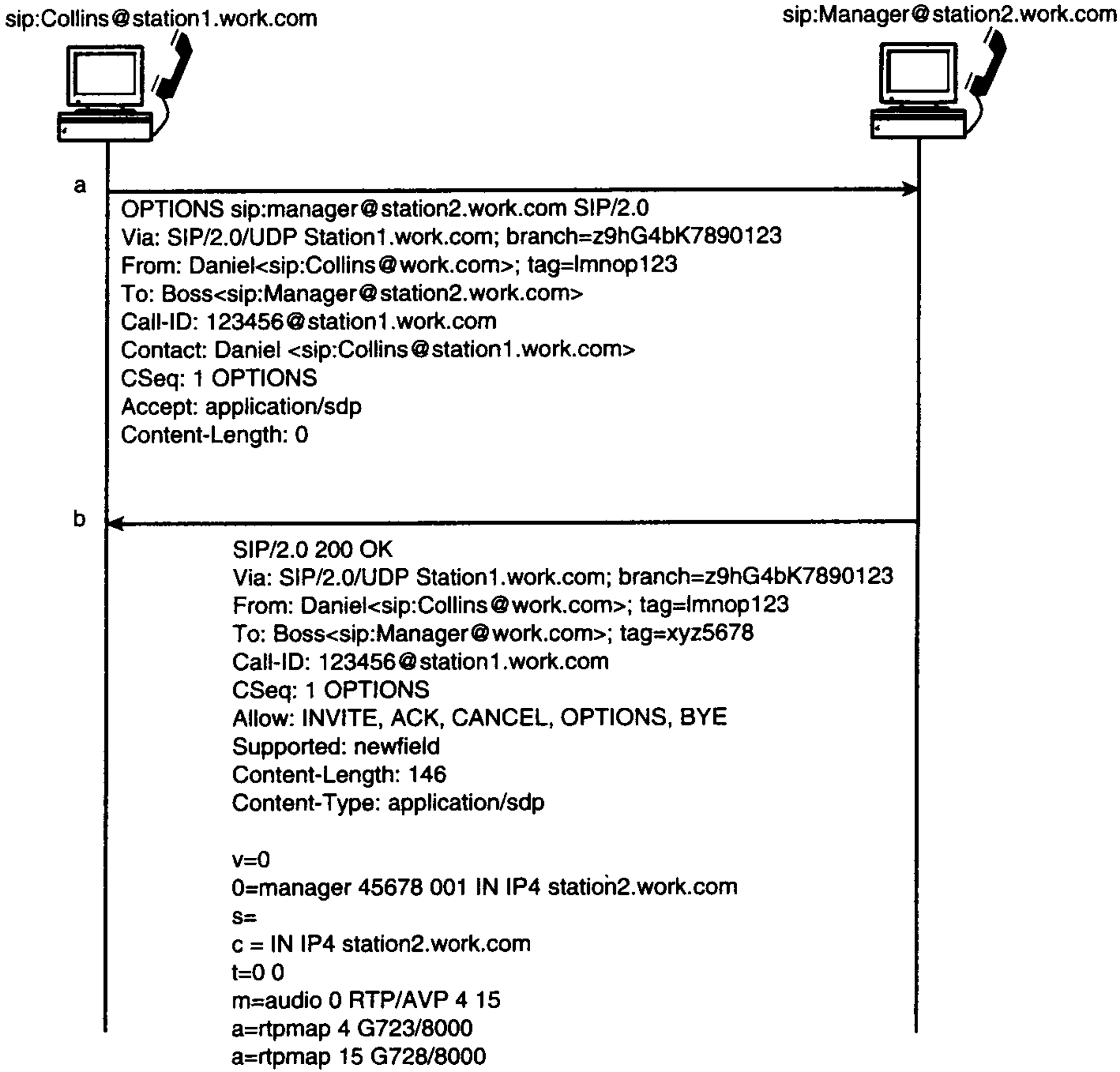
d

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 Boss can handle
- Supported Header
  - Indicate the SIP extensions that can be supported


sip:Collins@station1.work.com
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@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
0=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
  - 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
    - From the called party to calling party
    - 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

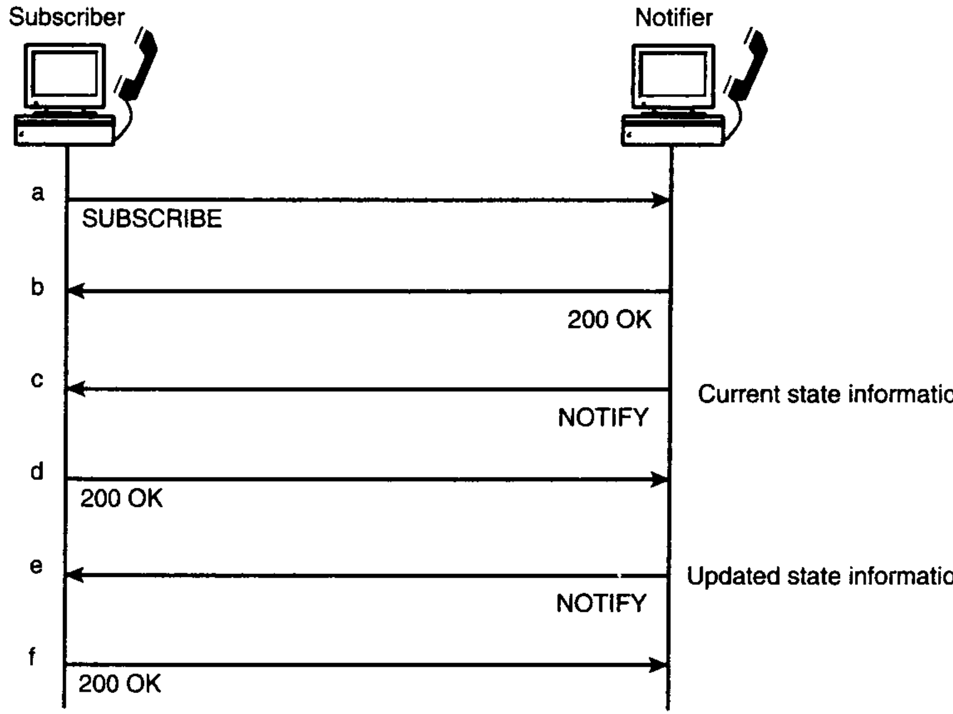
- The Require Header
  - In request
    - A client indicates that a server must support certain extension.
  - In response
    - 421, extension required
- The Supported header
  - RFC 2543 – Require: header (client -> server)
    - 420 (bad extension) – server -> client
  - Can be included in both requests and 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

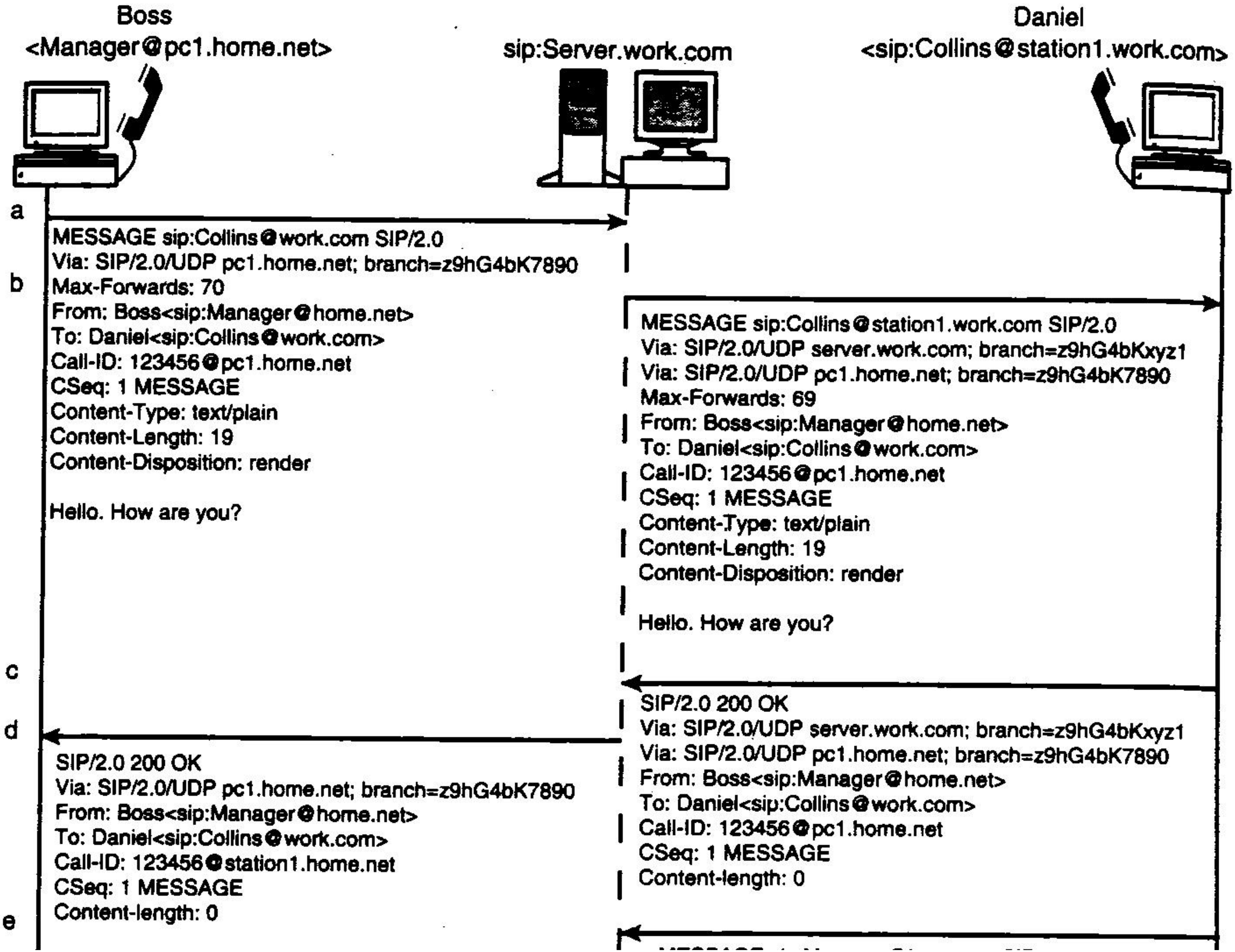
SIP Event Notification

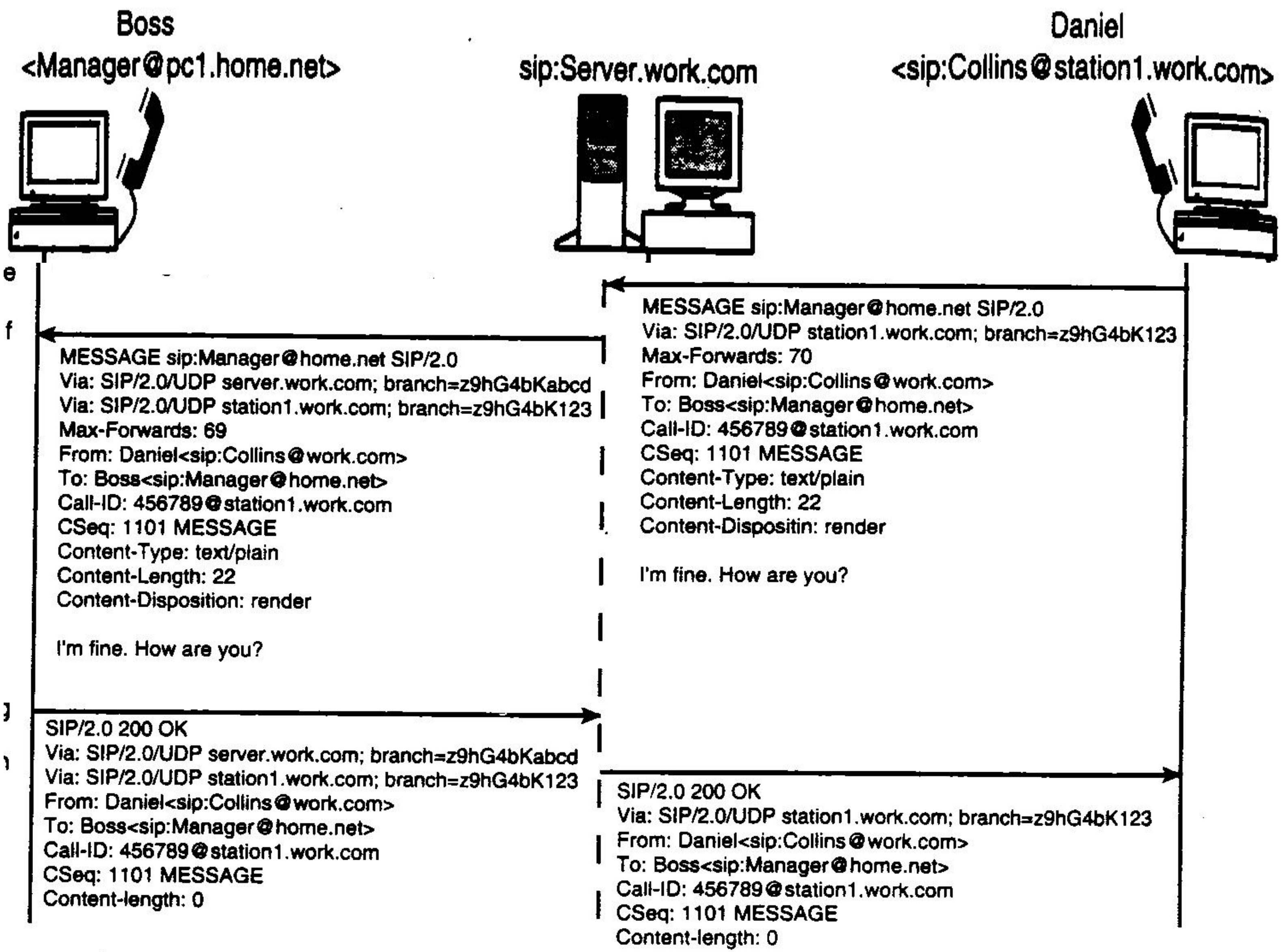
- Several SIP-based applications have been devised based on the concept of a user being informed of some event.
  - E.g., Instant messaging
- RFC 3265 has addressed the issue of event notification.
  - SUBSCRIBE and NOTIFY
  - The Event header



SIP for Instant Messaging

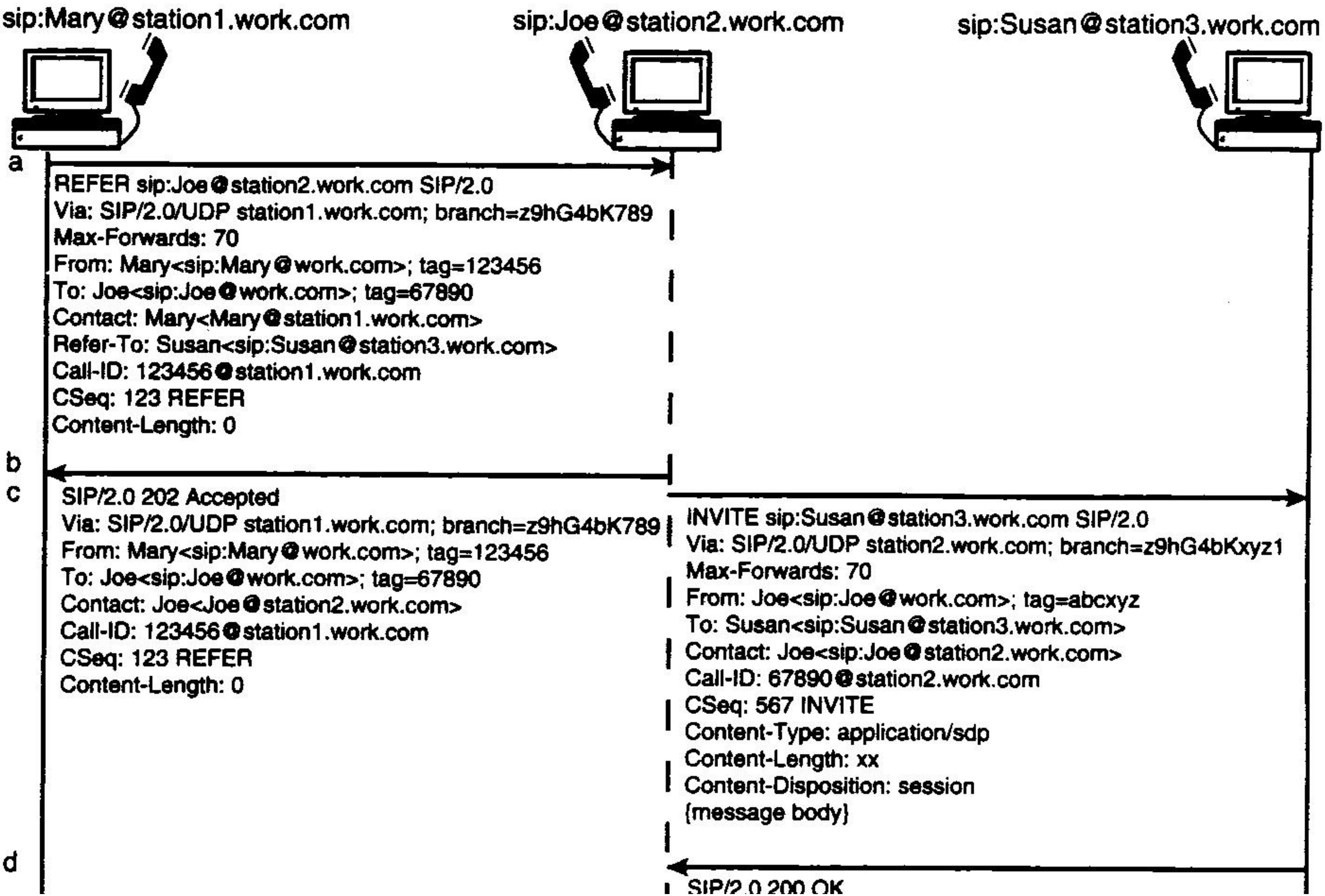
- The IETF working group – SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE)
- A new SIP method – MESSAGE
  - This request carries the actual message in a message body.
  - A MESSAGE request does not establish a SIP dialog.

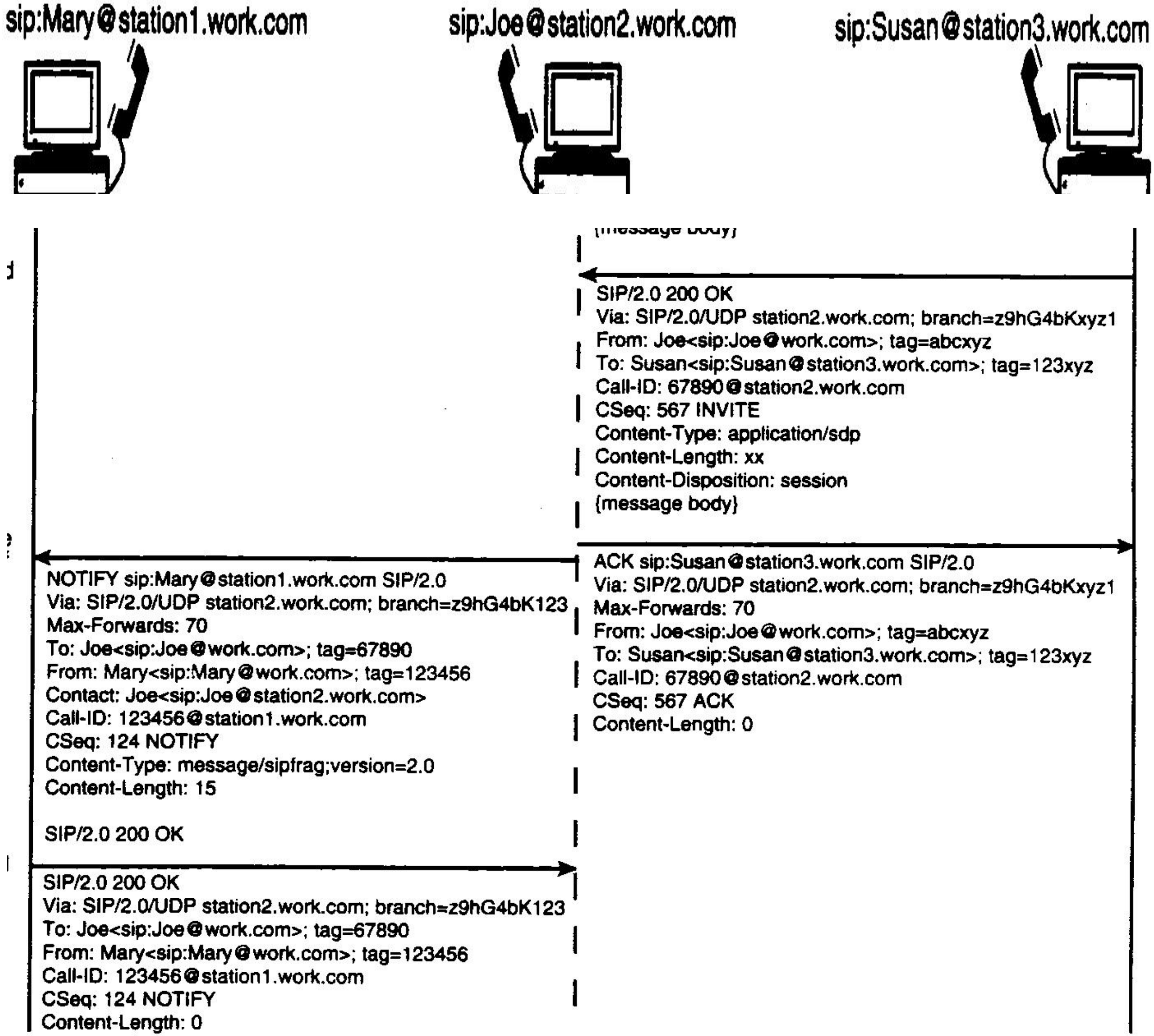

Boss
<Manager@pc1.home.net>
sip:Server.work.com
Daniel
<sip:Collins@station1.work.com>
a
MESSAGE sip:Collins@work.com SIP/2.0
Via: SIP/2.0/UDP pc1.home.net; branch=z9hG4bK7890
b
Max-Forwards: 70
From: Boss<sip:Manager@home.net>
To: Daniel<sip:Collins@work.com>
Call-ID: 123456@pc1.home.net
CSeq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 19
Content-Disposition: render
Hello. How are you?
MESSAGE sip:Collins@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@station1.home.net
CSeq: 1 MESSAGE
Content-length: 0
e


Boss
<Manager@pc1.home.net>
sip:Server.work.com
Daniel
<sip:Collins@station1.work.com>
e
f
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-Dispositin: render
I'm fine. How are you?
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?
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: Susan<sip:Susan@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<sip: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}
d
SIP/2.0 200 OK


sip:Mary@station1.work.com
sip:Joe@station2.work.com
sip:Susan@station3.work.com
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}
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
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<sip:Joe@station2.work.com>
Call-ID: 123456@station1.work.com
CSeq: 124 NOTIFY
Content-Type: message/sipfrag;version=2.0
Content-Length: 15
SIP/2.0 200 OK
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

- 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 → Q931 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.

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

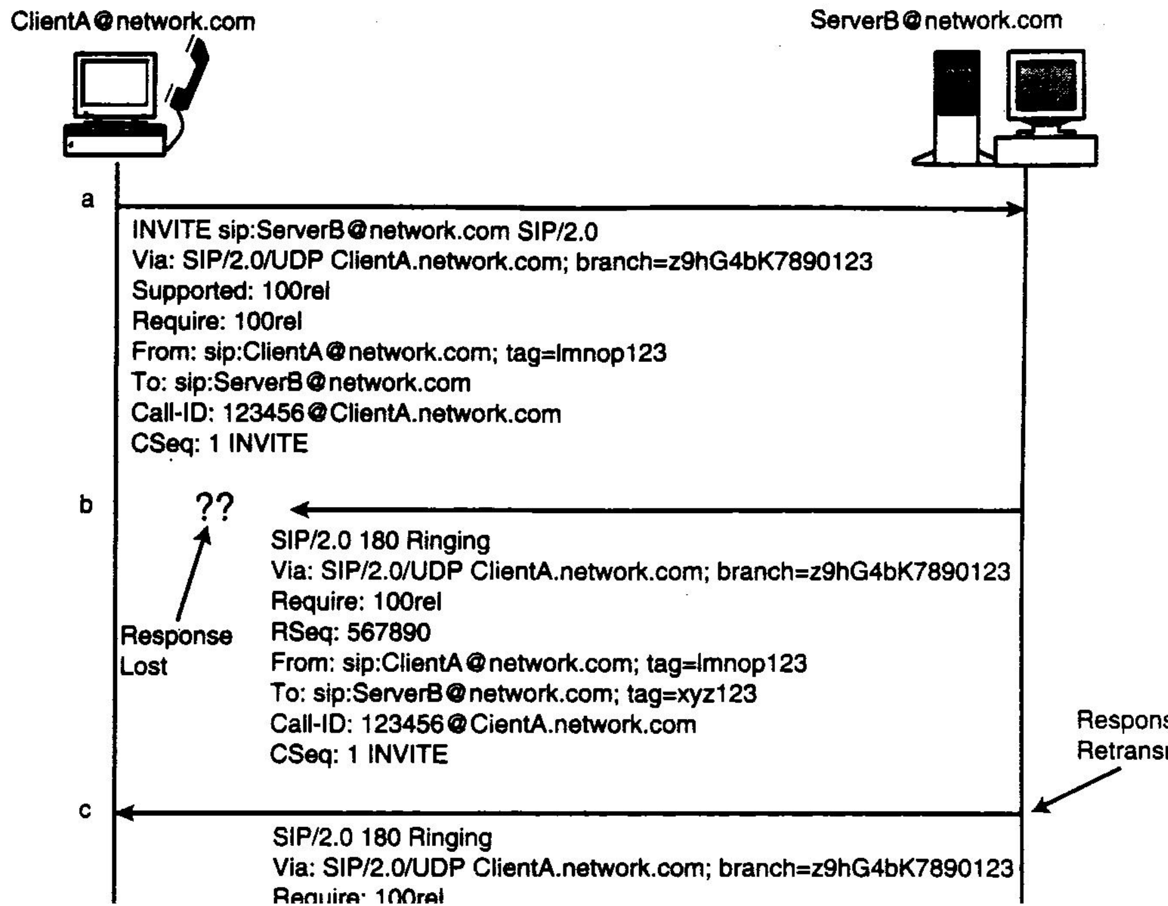
PRACK

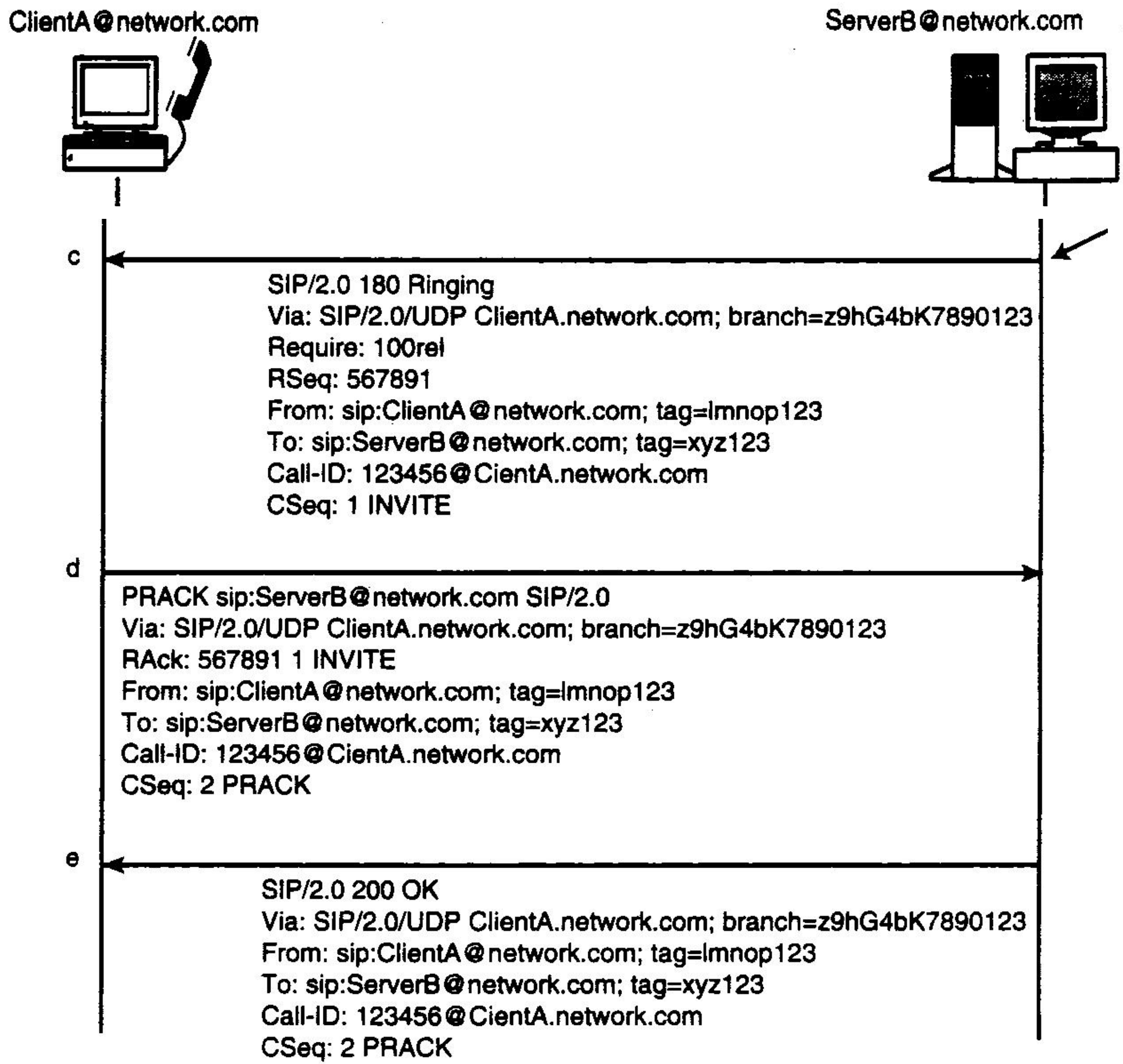
- Prov. Resp. ACK

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@CientA.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@CientA.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@CientA.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
  - E.g., to change the codec
- One important usage is when reserving network resources as part of a SIP session establishment

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

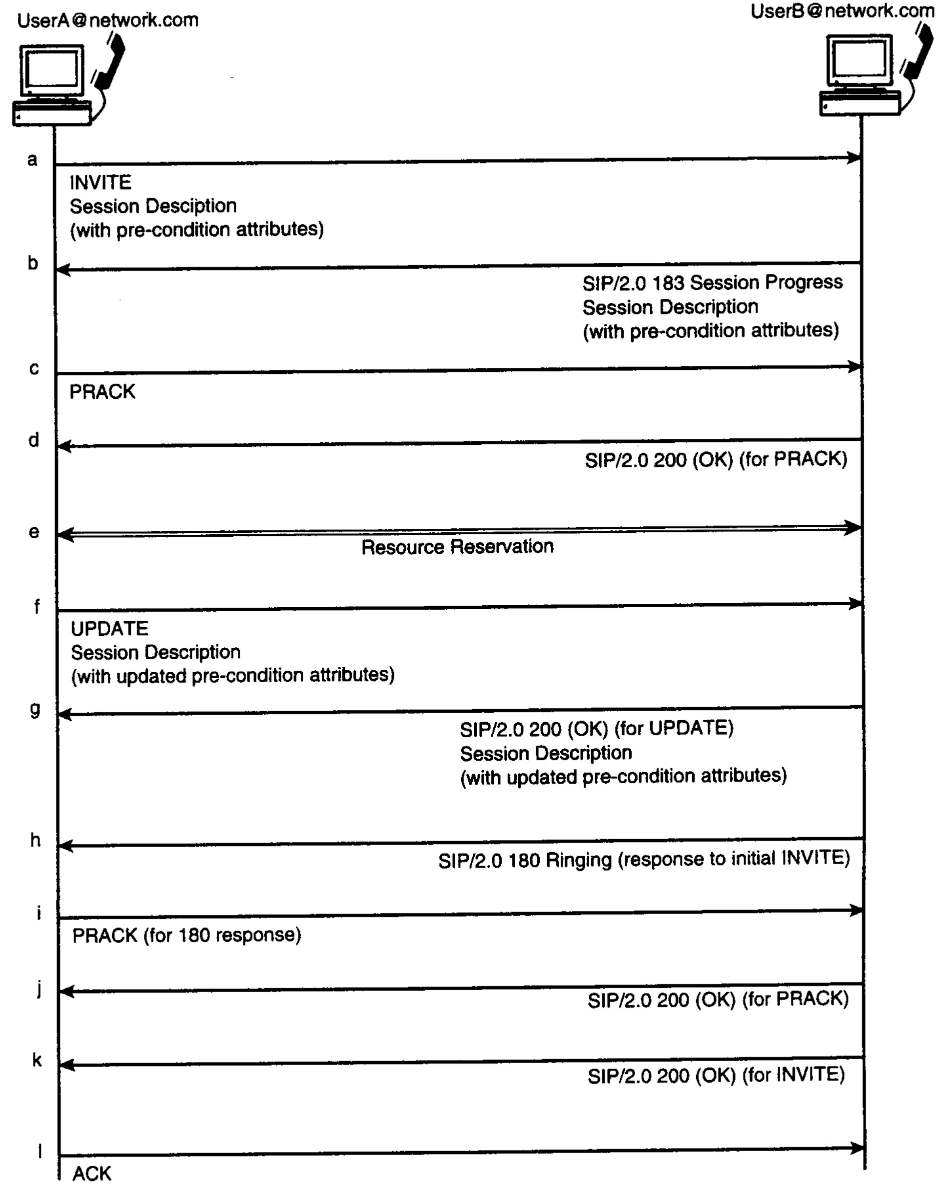
- Ensuring that sufficient resources are available to handle a media stream is a very important.
  - To provide a high-quality service for a carrier-grade network
- The signaling might take a different path from the media.
  - The successful transfer of signaling messages does not imply to a successful 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

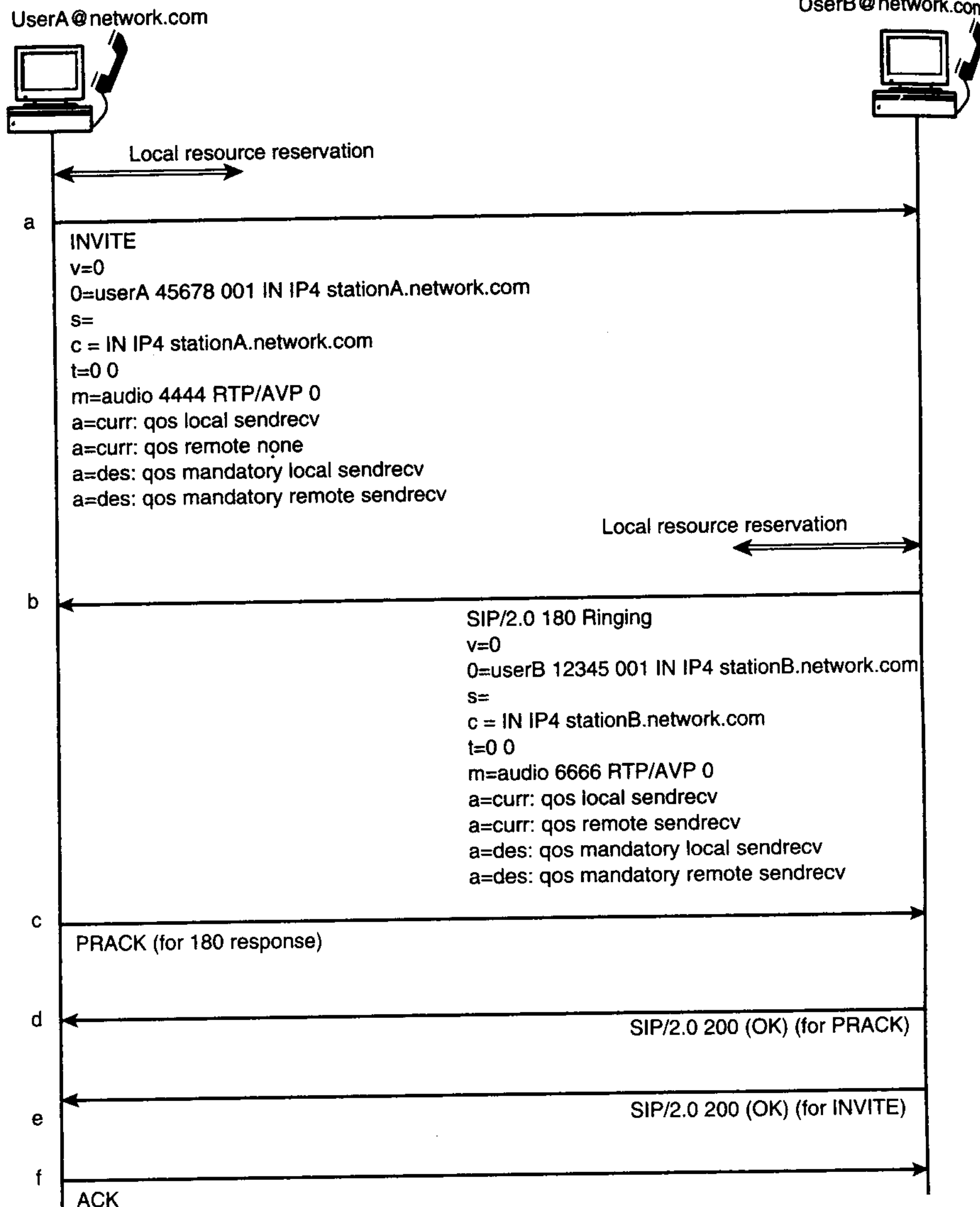
  v=0
  o=userA 45678 001 IN IP4 stationA.network.com
  s=
  c=IN IP4 stationA.nework.com
  t=0 0
  m=audio 4444 RTP/AVP 0
  a=curr: qos e2e send
  a=des: qos mandatory e2e sendrecv

- 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.

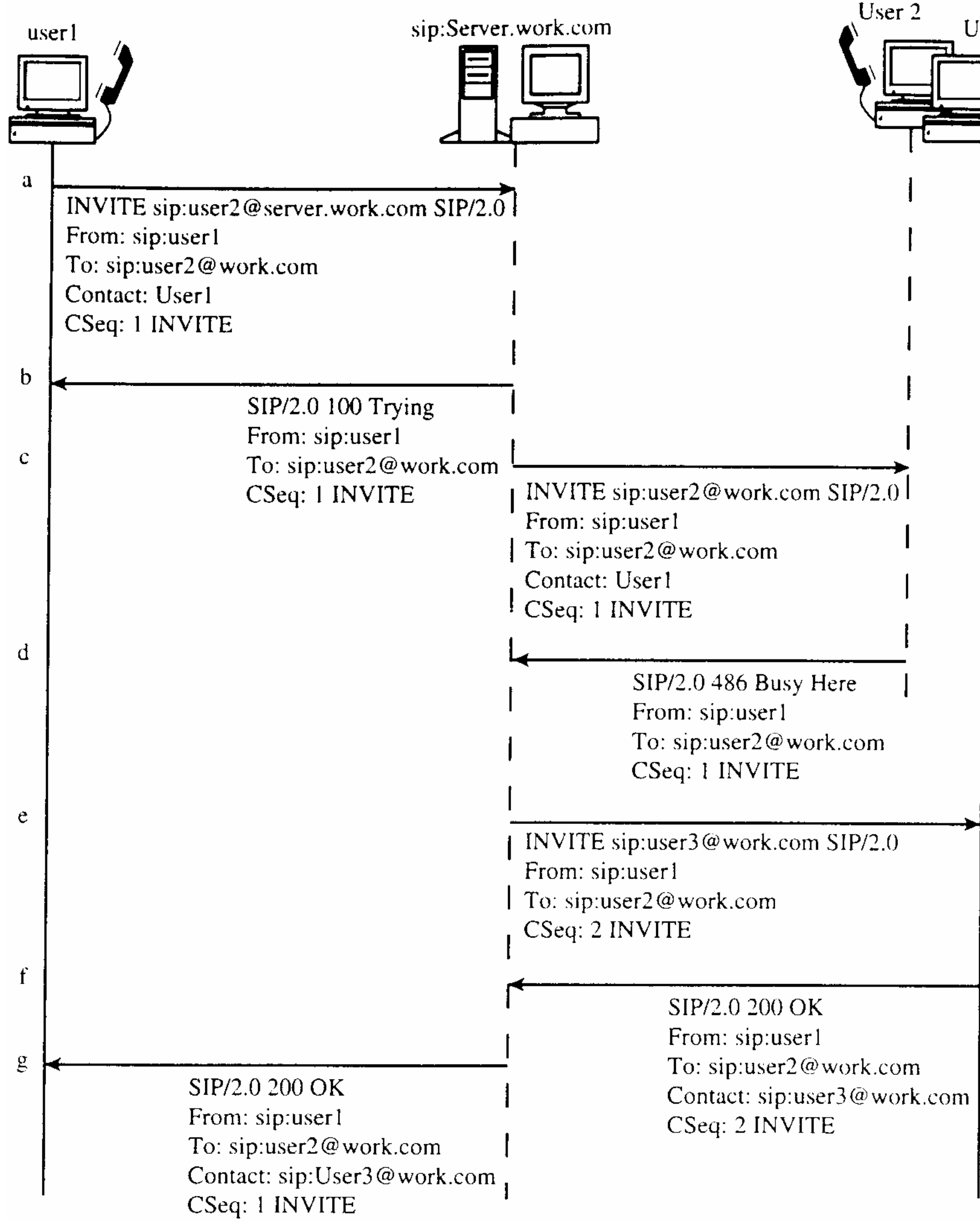


Usage of SIP for Features/Services

- 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
- 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
- Proxy invokes various types of advanced feature logic.
  - Policy server (call-routing, QoS)
  - Authentication server
  - Use the services of an IN SCP over INAP

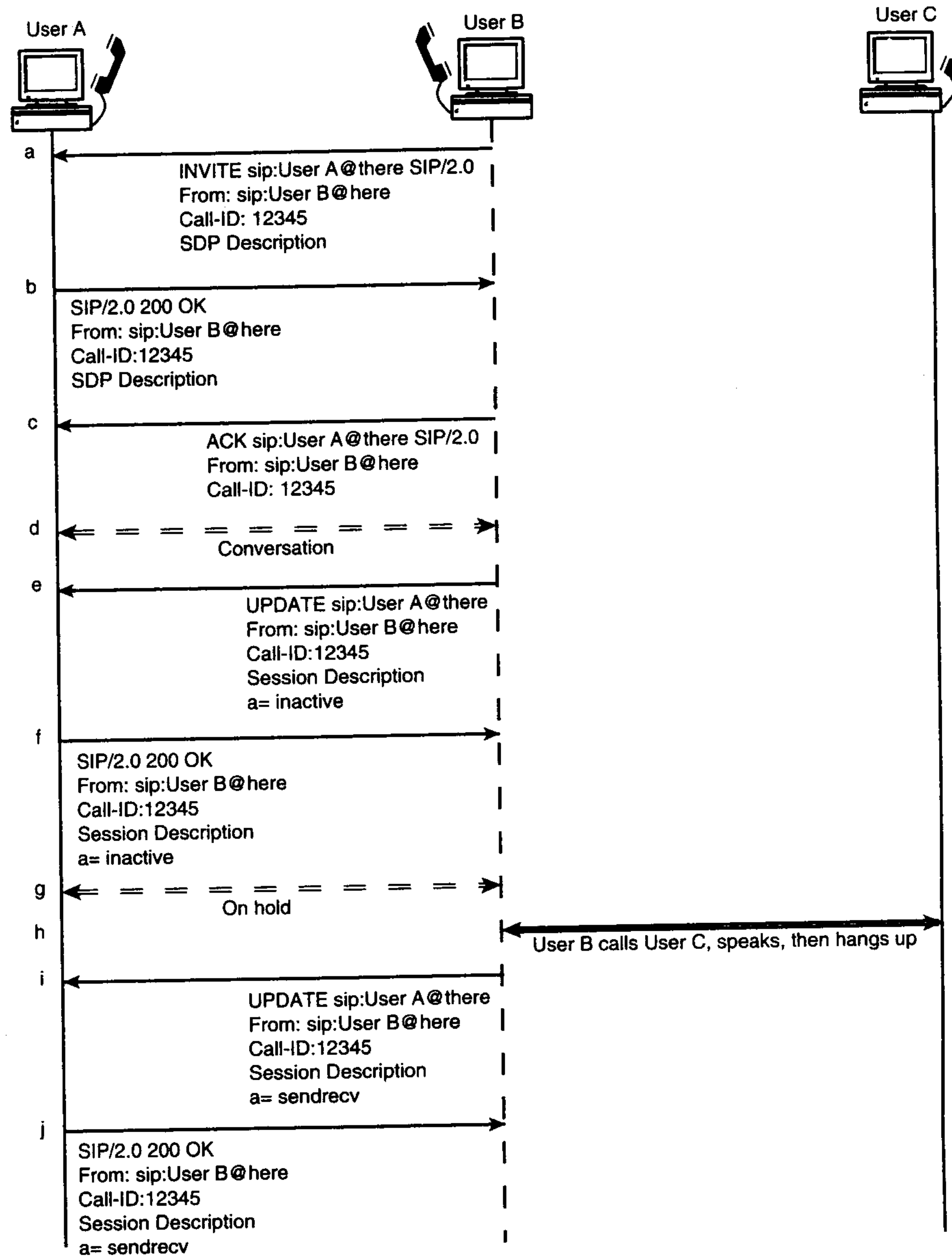
Call Forwarding

- 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



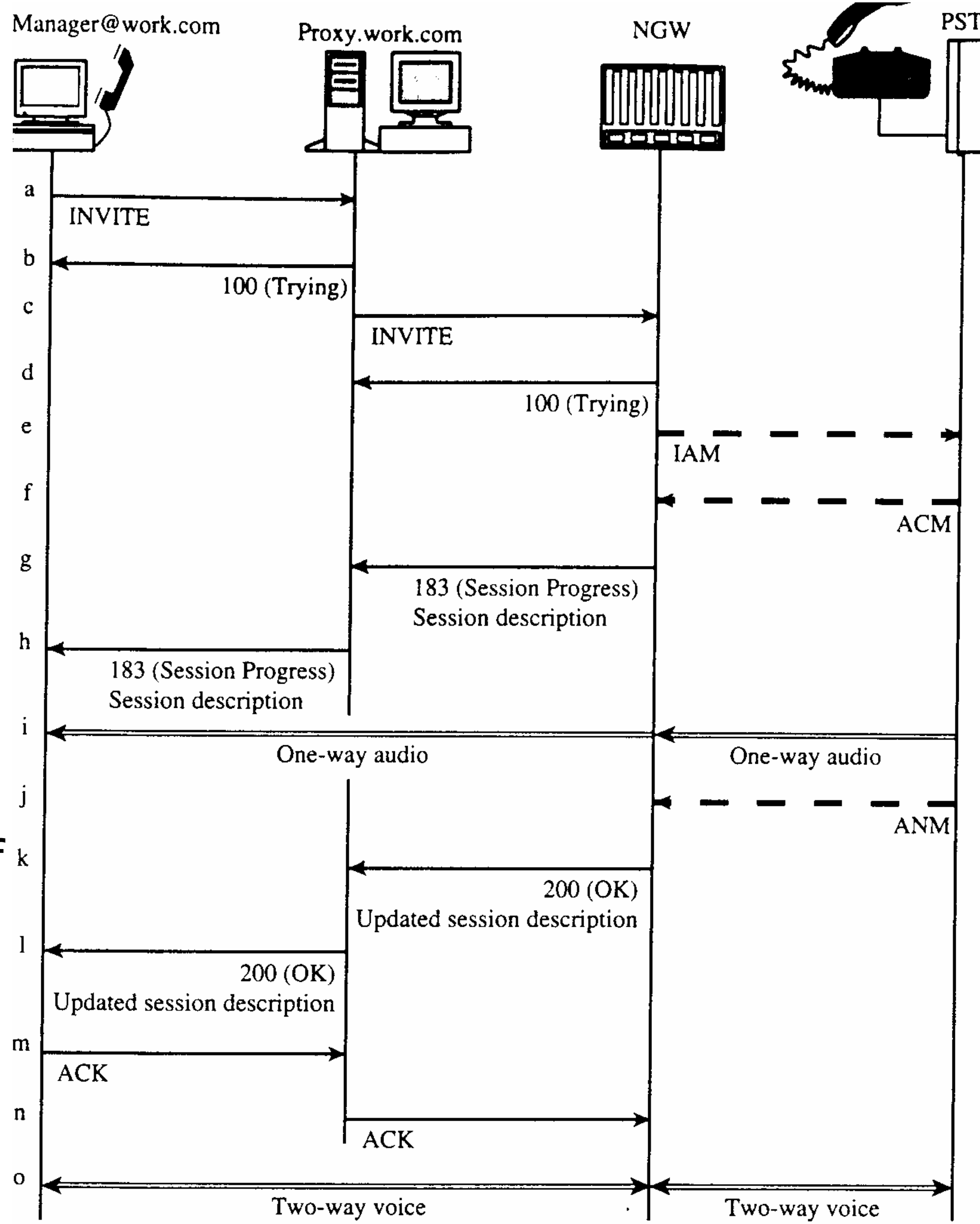
Consultation Hold

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



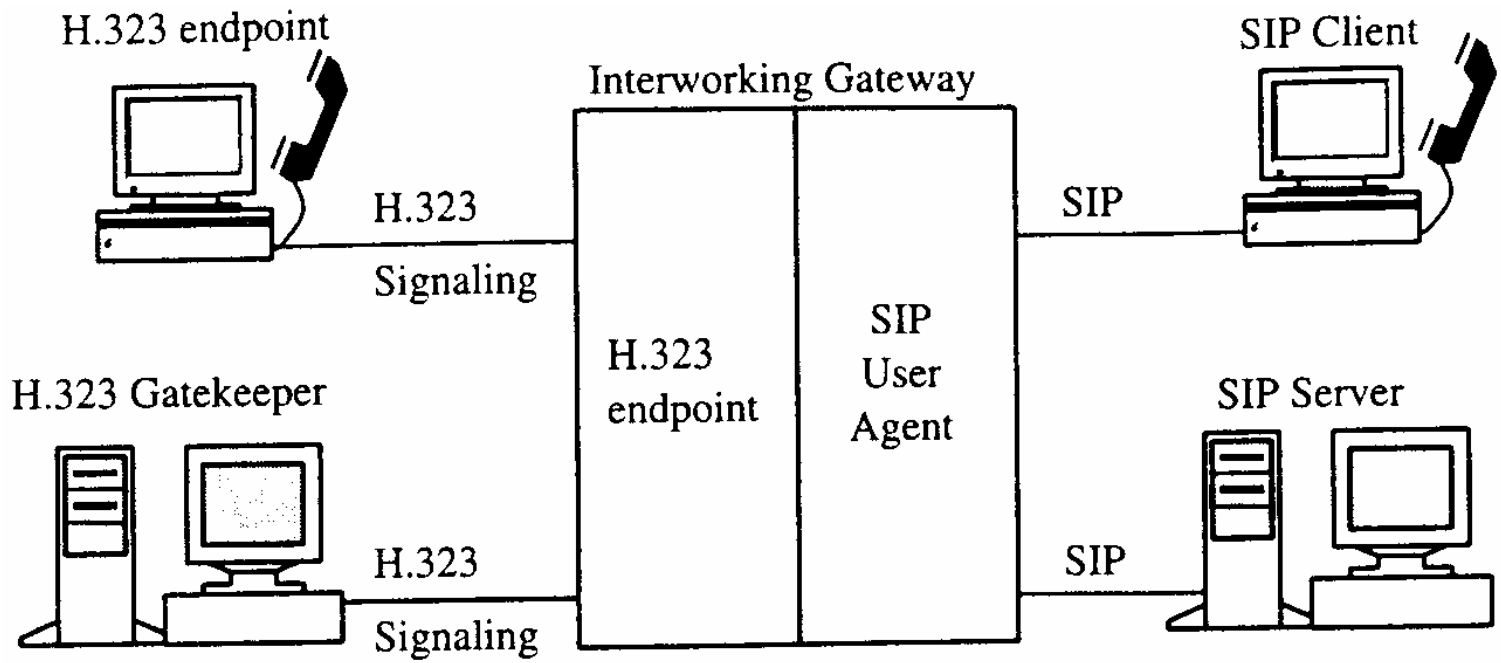
PSTN Interworking

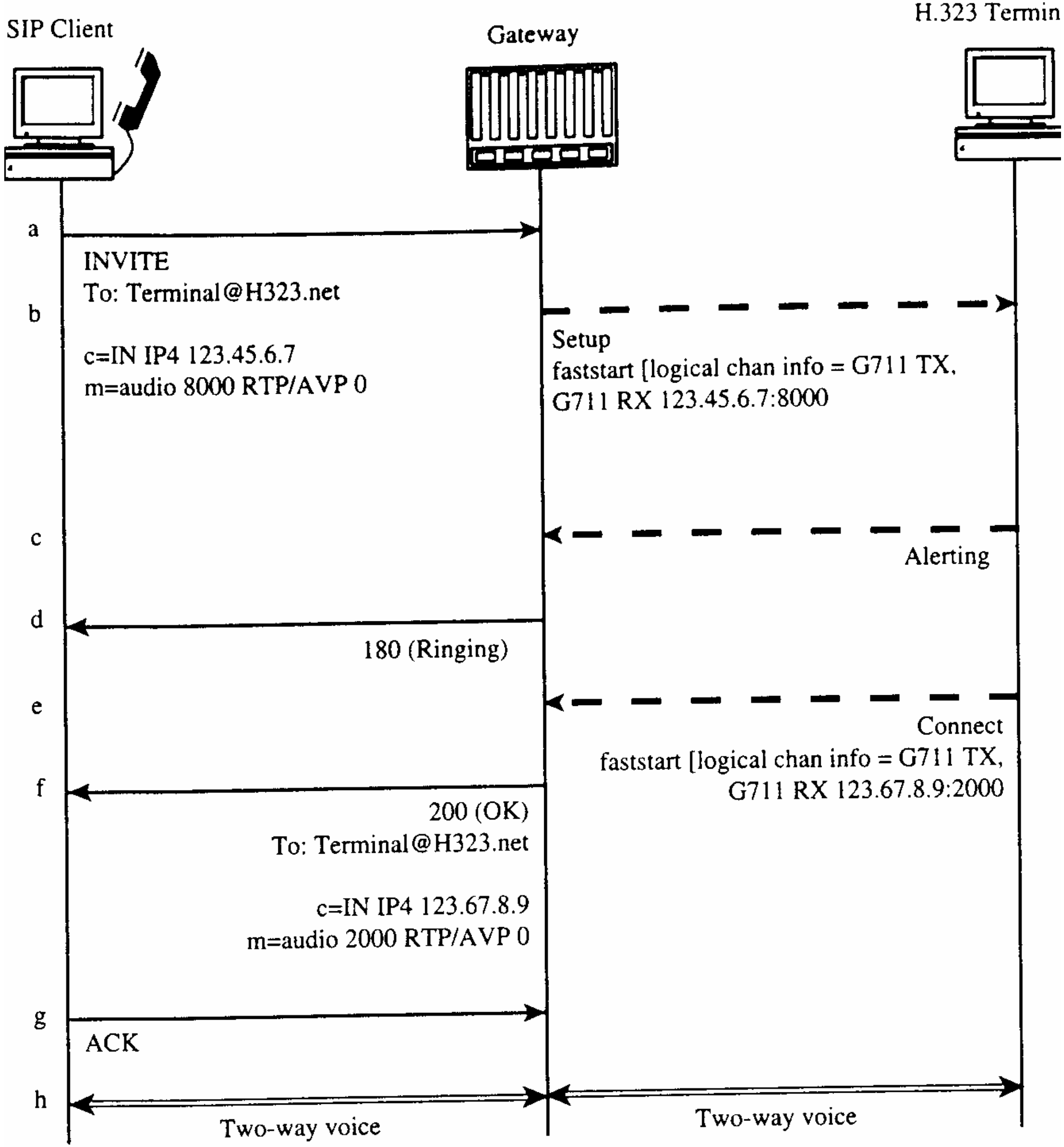
- PSTN Interworking
  - A SIP URL to a telephone number
  - A network gateway
- PSTN – SIP – PSTN
  - MIME media types
  - For ISUP
- SIP for Telephony (SIP-T)
- The whole issue of interworking with SS7 is fundamental to the success of VoIP in the real world.

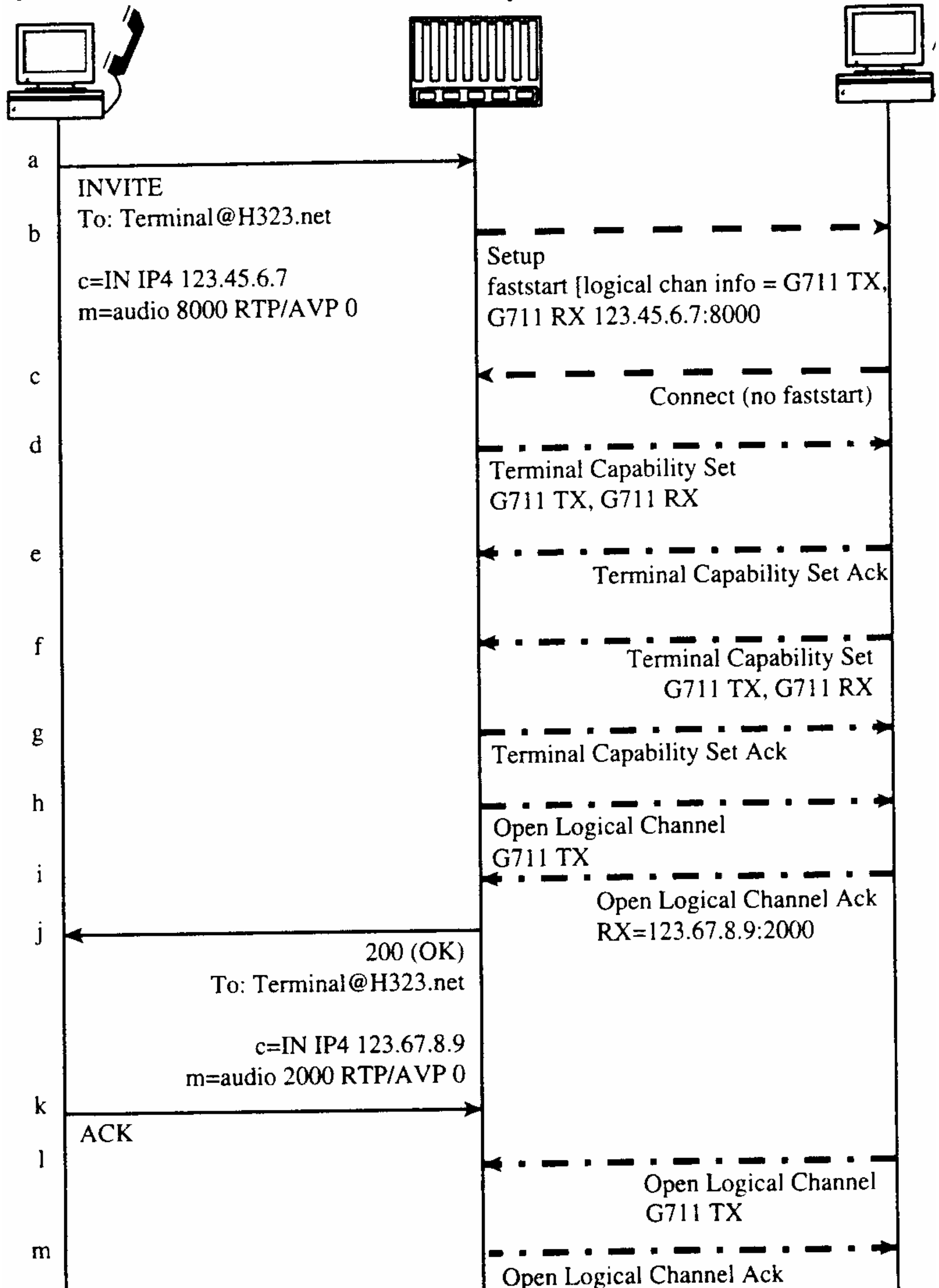


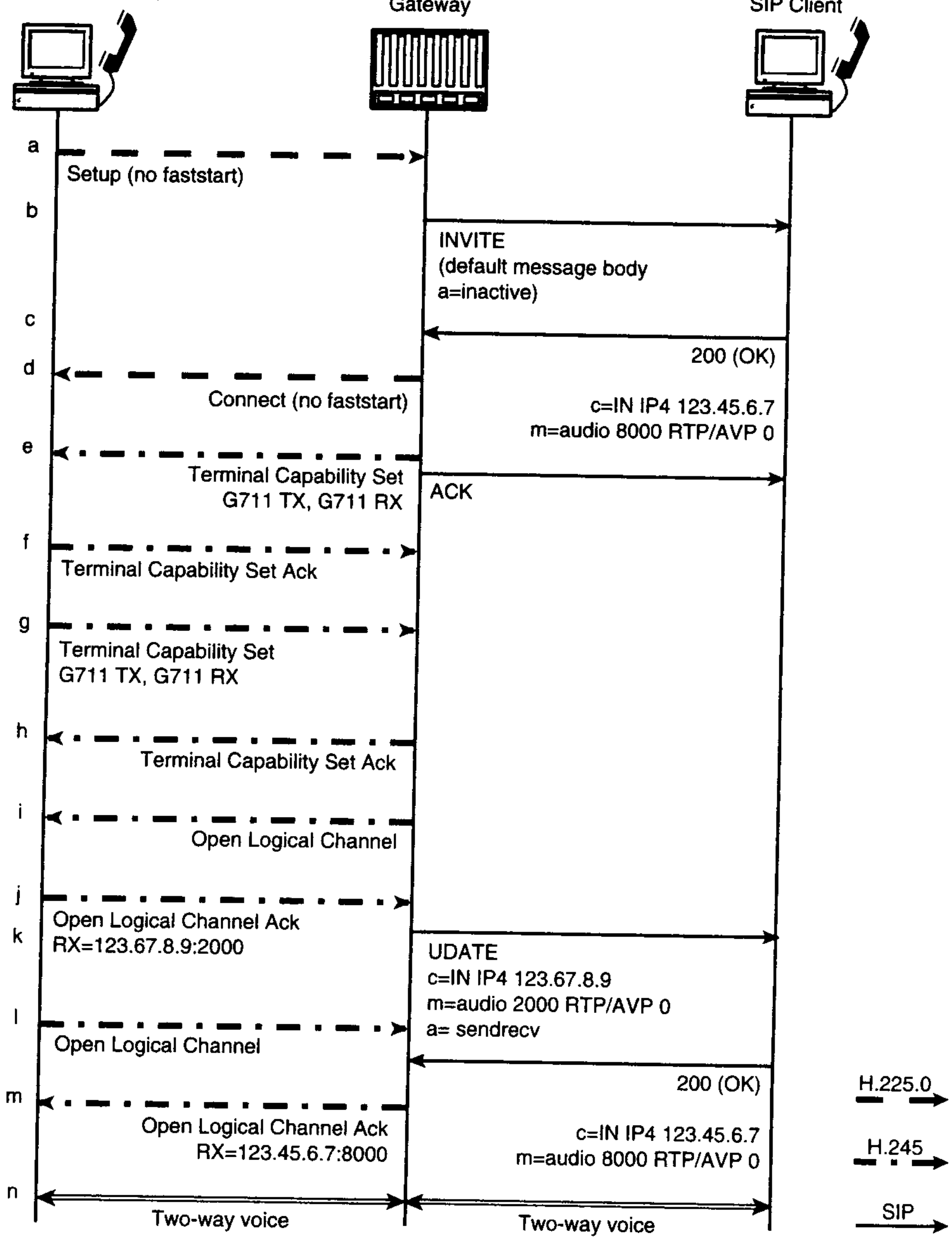
Interworking with H.323

- SIP-H.323 interworking gateway




SIP Client
Gateway
H.323 Termin
a
INVITE
To: Terminal@H323.net
c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0
b
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


a
INVITE
To: Terminal@H323.net
c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0
b
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
G711 TX
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
G711 TX
m
Open Logical Channel Ack


Gateway
SIP Client
a
Setup (no faststart)
b
INVITE
(default message body
a=inactive)
c
200 (OK)
d
Connect (no faststart)
c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0
e
Terminal Capability Set
G711 TX, G711 RX
ACK
f
Terminal Capability Set Ack
g
Terminal Capability Set
G711 TX, G711 RX
h
Terminal Capability Set Ack
i
Open Logical Channel
j
Open Logical Channel Ack
RX=123.67.8.9:2000
k
UDATE
c=IN IP4 123.67.8.9
m=audio 2000 RTP/AVP 0
a= sendrecv
l
Open Logical Channel
200 (OK)
m
Open Logical Channel Ack
RX=123.45.6.7:8000
c=IN IP4 123.45.6.7
m=audio 8000 RTP/AVP 0
n
Two-way voice
Two-way voice
H.225.0
H.245
SIP

Summary

- The future for signaling in VoIP networks
  - Simple, yet flexible
  - Easier to implement
  - Fit well with the media gateway control protocols
- 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.