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Communication Systems

SIP

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Organization

- I. Data and voice communication in IP networks
- II. Security issues in networking
- **III. Digital telephony networks and voice over IP**

Part 3
Digital, Internet Telephony

- 3rd and last part of the communication systems lecture: digital telephony
- For a rather long time telephone and data networks were different entities – remember the network taxonomy
  - packet orientated vs. circuit switched
  - packet orientation is rather efficient in bandwidth using but cannot give any guarantees on packet delivery
  - bandwidth growth and optional QoS helped to offer service quality near to circuit switching
- Why to provide two completely different infrastructures for rather the same services?
  - voice is just another piece of data (with some special requirements)...

Application Layer Protocols – Internet Telephony

- Voice-over-IP is getting more and more ubiquitous
  - every network equipment vendor has some products in its portfolio (even companies like Siemens are able to offer products conforming to standards!!)
  - many new “telephone companies” evolve to offer services, the old providers have to think on new strategies
  - all of them hope for reduce of costs and a source for roaring profits :-)
- That way TCP/IP is just used for another application/service
- This service has to meet some requirements nevertheless

Internet Telephony - Requirements

- Security
  - reduced costs might induce new type of SPAM – spit (spam over Internet telephony)
  - how to know that the caller is the one he claims to, same for the called partner
- Compatibility to existing services
  - routing of emergency calls
  - location of emergency
- Presence
  - robustness of servers and “routes”
  - permanent updates of clients (mobile devices move from network to network)

Internet Telephony - Requirements

- Voice over IP should offer
  - higher robustness (e.g. alternate routes)
  - better voice quality
  - mobility, multimedia and conferencing
  - secure communication
  - gateways to other telephone systems (GSM, UMTS, PSTN)
  - 100% open standards
- Hope of a combination of lower costs with better functionality

Internet Telephony – Infrastructure (idealized)

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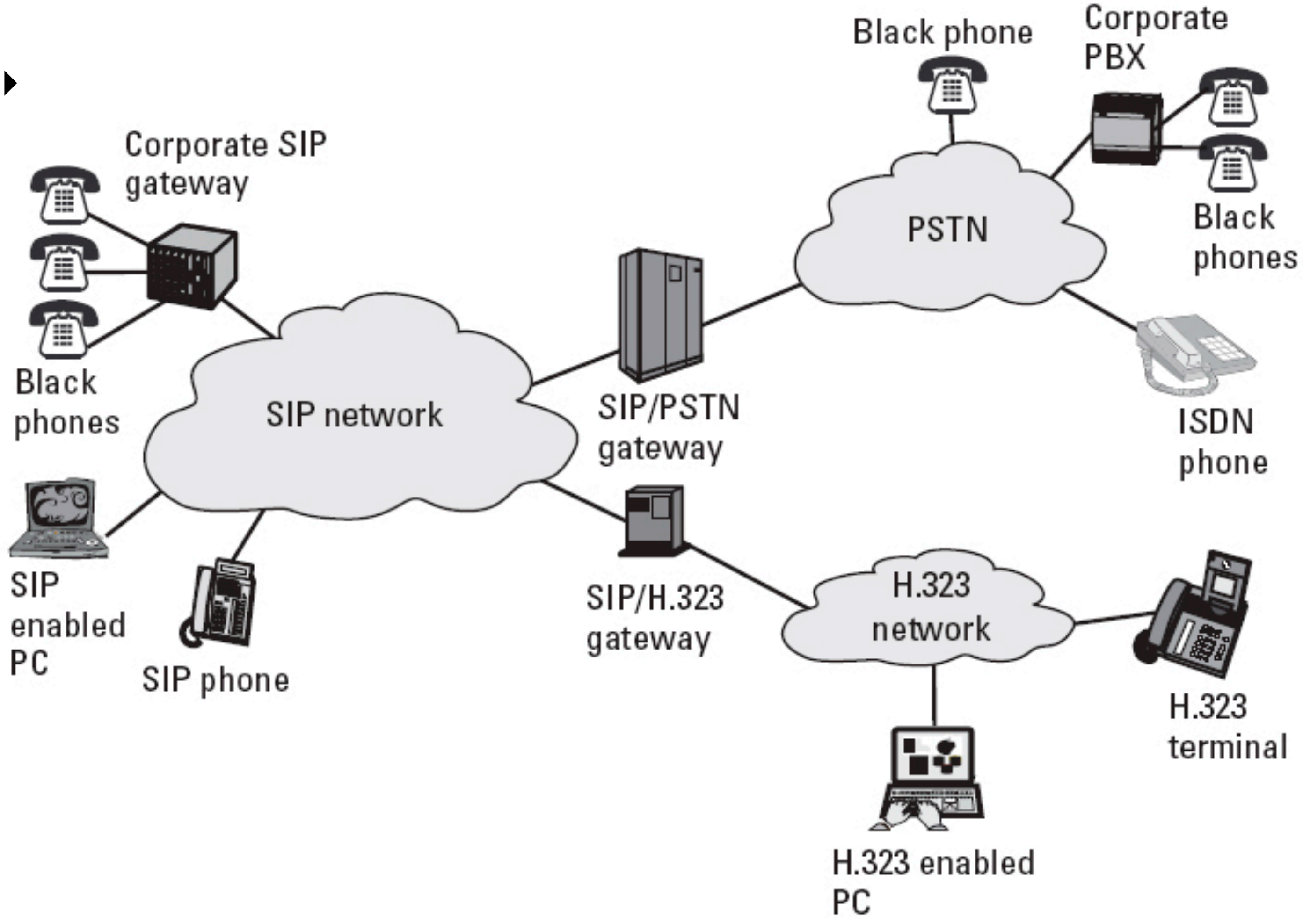
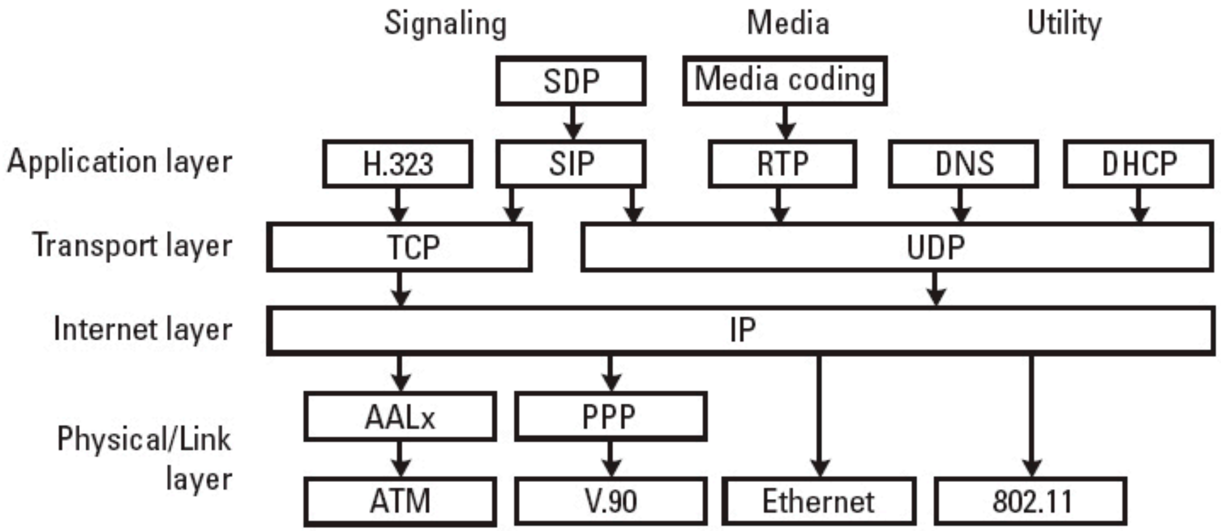


Internet Telephony - Standards

- Requirements by VoIP services
  - enough bandwidth for digitized audio stream (both directions!)
  - minimal jitter and noise
- Two main VoIP standards (in the sense of open, other standards e.g. by Cisco)
  - SIP – internet standard
  - H323 – standard developed by Telcos - ITU (second part of lecture)
- SIP is session initialization protocol
  - developed by Henning Schulzrinne (Feb. 1999)
  - IETF Standard RFC 2543 (March 1999)
  - current: RFC 3261 (June 2002)

Internet Telephony - SIP

- SIP just for session setup not for transport of multimedia streams
- inspired by HTTP
  - text based Peer-to-Peer application layer protocol
  - using requests and replies to set up a connection



Internet Telephony - SIP

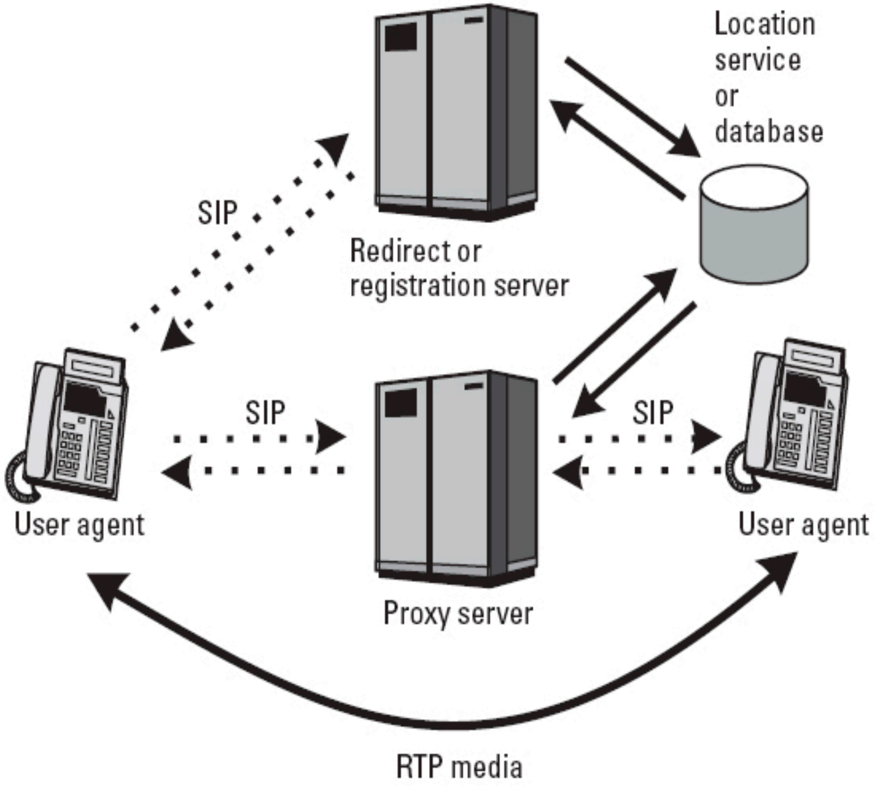
- Requirements toward SIP
  - localization of endpoints
  - setup of connections
  - exchange of media and presence information
  - modification of sessions: rerouting and cancelling of calls
  - complete a session
  - scalability (more than one session should be possible)
- SIP addresses designed same way as email addresses
  - sip: “userID@sipgateway.site”

SIP - entities

- Peers = User Agents (UA)
- a UA can fulfill on of the following roles
  - user agent client (UAC) = initiator of a request
  - user agent server (UAS) = application, which contacts the user and answers requests for him
- SIP clients
  - telephones: as UAC or UAS
  - Gateways: connections to other networks, translates between different audio and video codecs
- SIP server
  - might act as proxy server and could be used for
    - authentification, authorization
    - secure routing and rerouting

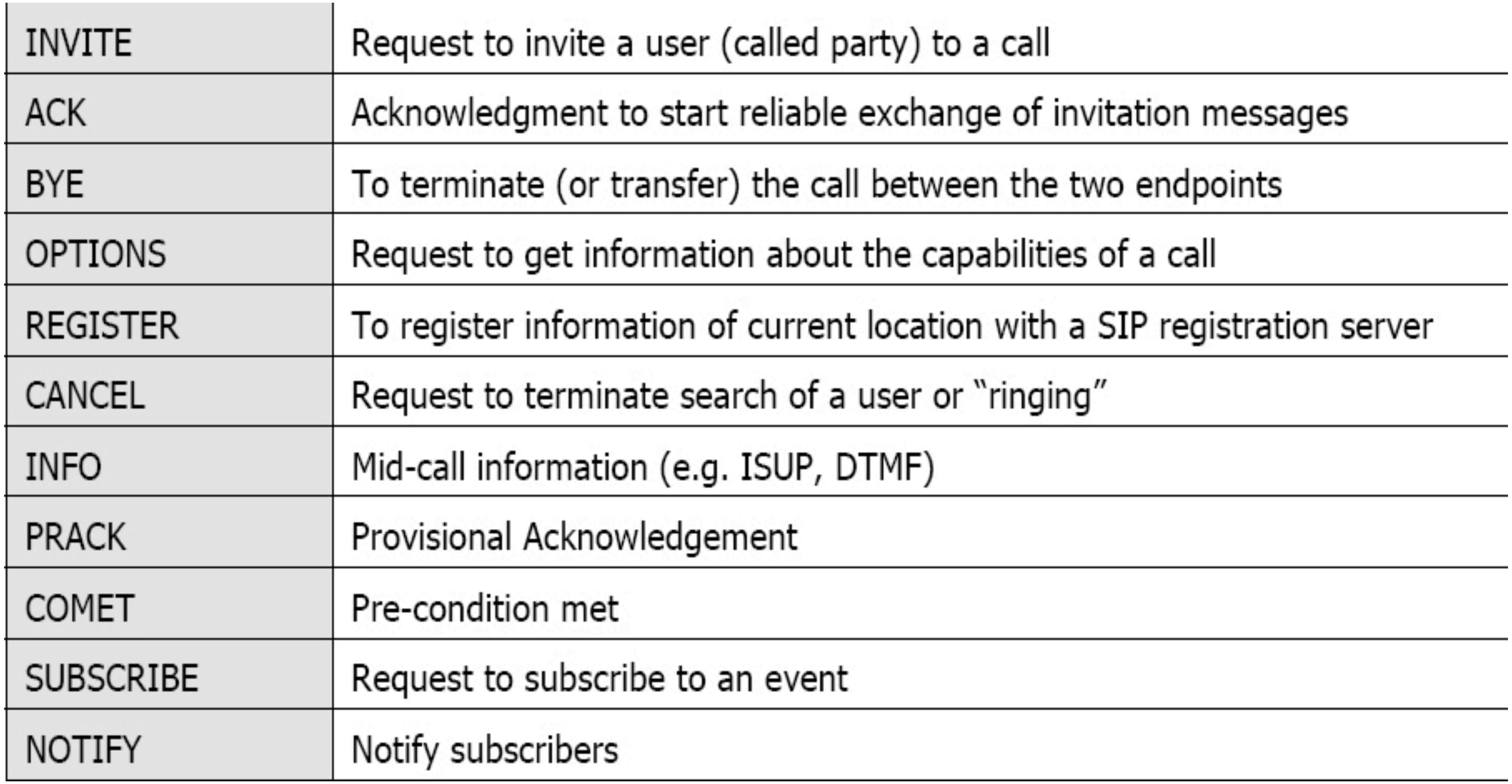
SIP – server

- SIP server
  - redirect server = information service
  - location server is the request address for the host on wich a given user might be reached on
  - registrar server acts as registration service
    - registers the current location of the clients
    - often at the same place as proxy or redirect
    - is not a required component for SIP, but useful in larger setups



SIP – message types

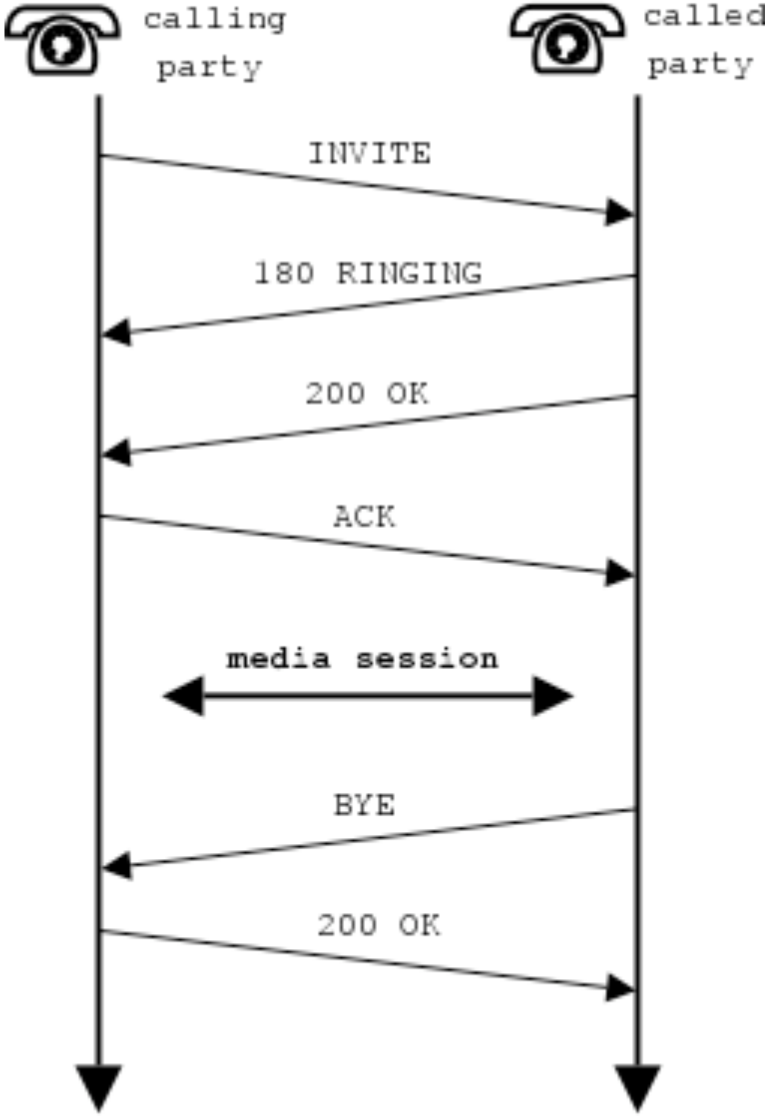
- SIP defines messages for communication setup end ending



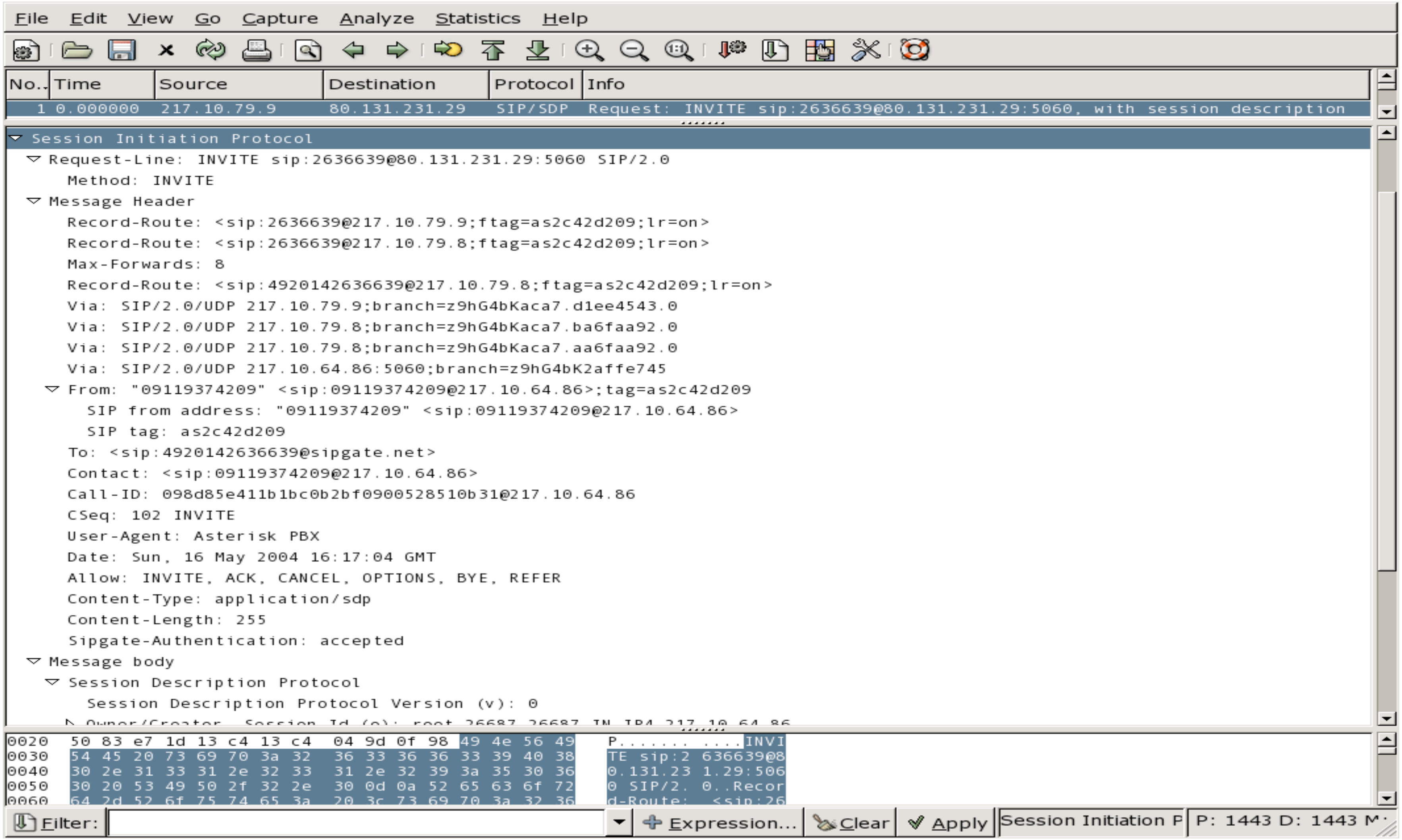
INVITE	Request to invite a user (called party) to a call
ACK	Acknowledgment to start reliable exchange of invitation messages
BYE	To terminate (or transfer) the call between the two endpoints
OPTIONS	Request to get information about the capabilities of a call
REGISTER	To register information of current location with a SIP registration server
CANCEL	Request to terminate search of a user or "ringing"
INFO	Mid-call information (e.g. ISUP, DTMF)
PRACK	Provisional Acknowledgement
COMET	Pre-condition met
SUBSCRIBE	Request to subscribe to an event
NOTIFY	Notify subscribers

SIP – direct example session

- Direct SIP connection
- Disadvantage:
  - the calling party has to know the IP address of called party
- INVITE message contains the details, which type of session is to be initiated



SIP – direct example session



```
No.. Time       Source        Destination    Protocol  Info
   1 0.000000   217.10.79.9   80.131.231.29  SIP/SDP   Request: INVITE sip:2636639@80.131.231.29:5060, with session description
```

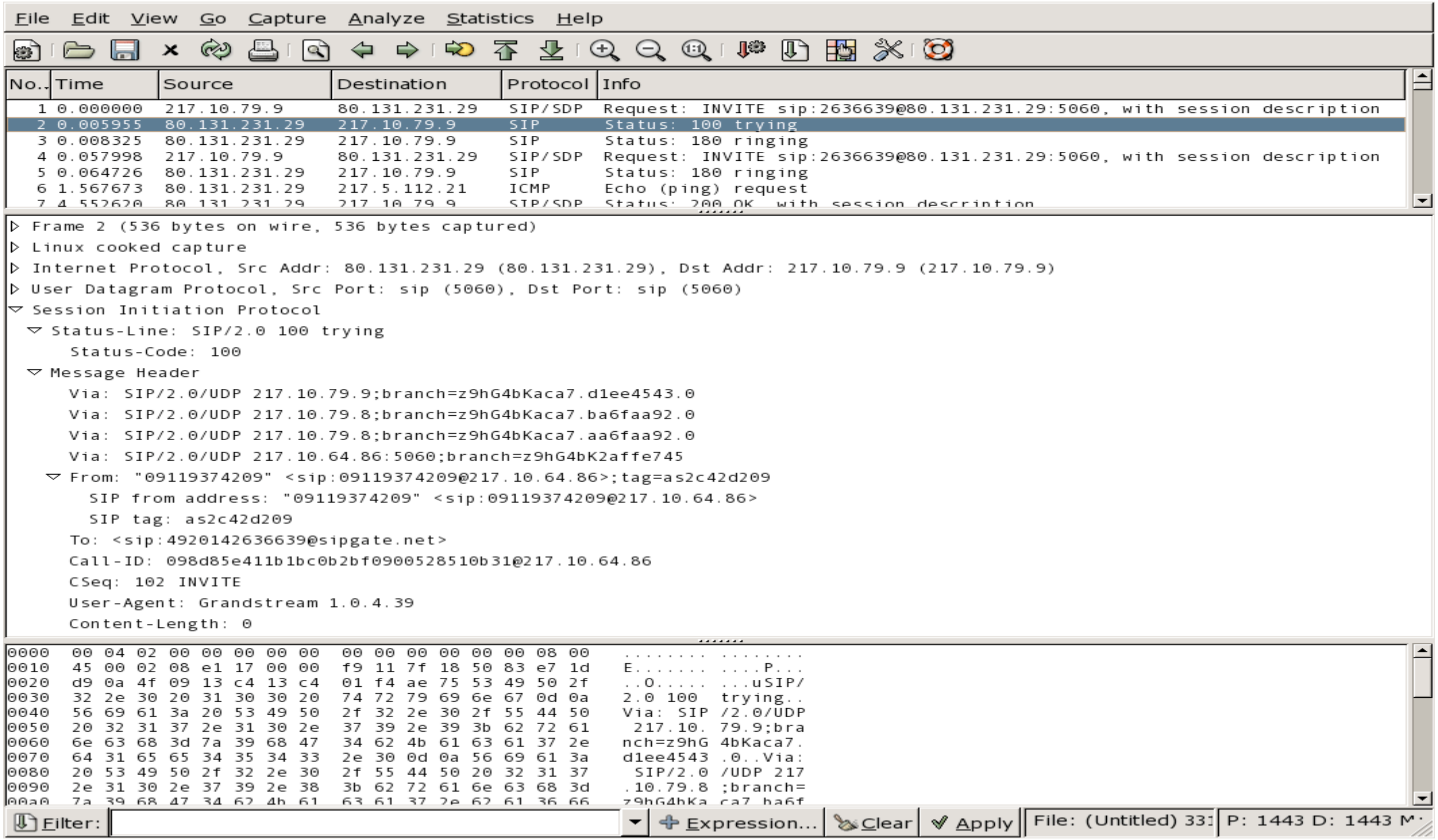
```
▽ Session Initiation Protocol
  ▽ Request-Line: INVITE sip:2636639@80.131.231.29:5060 SIP/2.0
      Method: INVITE
  ▽ Message Header
      Record-Route: <sip:2636639@217.10.79.9;ftag=as2c42d209;lr=on>
      Record-Route: <sip:2636639@217.10.79.8;ftag=as2c42d209;lr=on>
      Max-Forwards: 8
      Record-Route: <sip:4920142636639@217.10.79.8;ftag=as2c42d209;lr=on>
      Via: SIP/2.0/UDP 217.10.79.9;branch=z9hG4bKaca7.d1ee4543.0
      Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.ba6faa92.0
      Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.aa6faa92.0
      Via: SIP/2.0/UDP 217.10.64.86:5060;branch=z9hG4bK2affe745
    ▽ From: "09119374209" <sip:09119374209@217.10.64.86>;tag=as2c42d209
        SIP from address: "09119374209" <sip:09119374209@217.10.64.86>
        SIP tag: as2c42d209
      To: <sip:4920142636639@sipgate.net>
      Contact: <sip:09119374209@217.10.64.86>
      Call-ID: 098d85e411b1bc0b2bf0900528510b31@217.10.64.86
      CSeq: 102 INVITE
      User-Agent: Asterisk PBX
      Date: Sun, 16 May 2004 16:17:04 GMT
      Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER
      Content-Type: application/sdp
      Content-Length: 255
      Sipgate-Authentication: accepted
  ▽ Message body
    ▽ Session Description Protocol
        Session Description Protocol Version (v): 0
      ▷ Owner/Creator, Session Id (o): root 26687 26687 IN IP4 217.10.64.86
```

```
0020  50 83 e7 1d 13 c4 13 c4  04 9d 0f 98 49 4e 56 49   P....... ....INVI
0030  54 45 20 73 69 70 3a 32  36 33 36 36 33 39 40 38   TE sip:2 636639@8
0040  30 2e 31 33 31 2e 32 33  31 2e 32 39 3a 35 30 36   0.131.23 1.29:506
0050  30 20 53 49 50 2f 32 2e  30 0d 0a 52 65 63 6f 72   0 SIP/2. 0..Recor
0060  64 2d 52 6f 75 74 65 3a  20 3c 73 69 70 3a 32 36   d-Route:  <sip:26
```

SIP – header fields

- Request URI, SIP version number
- VIA: SIP version number, protocol, every SIP entity adds host and port, which created or routed the message
- Max-Forwards is decremented at every hop
- To, From: tags as identifier
- Call-ID: sender creates local non-ambiguous identifier which is globally unique in combination with the full qualified domain name
- CSeq: command sequence is incremented with every new request
- More optional fields
- Contact contains the SIP address of the current host, if connected over proxy – messages could be sent directly
- Content-Type and –Length tell the style of the following SDP body

SIP – “trying message” (message before ringing)

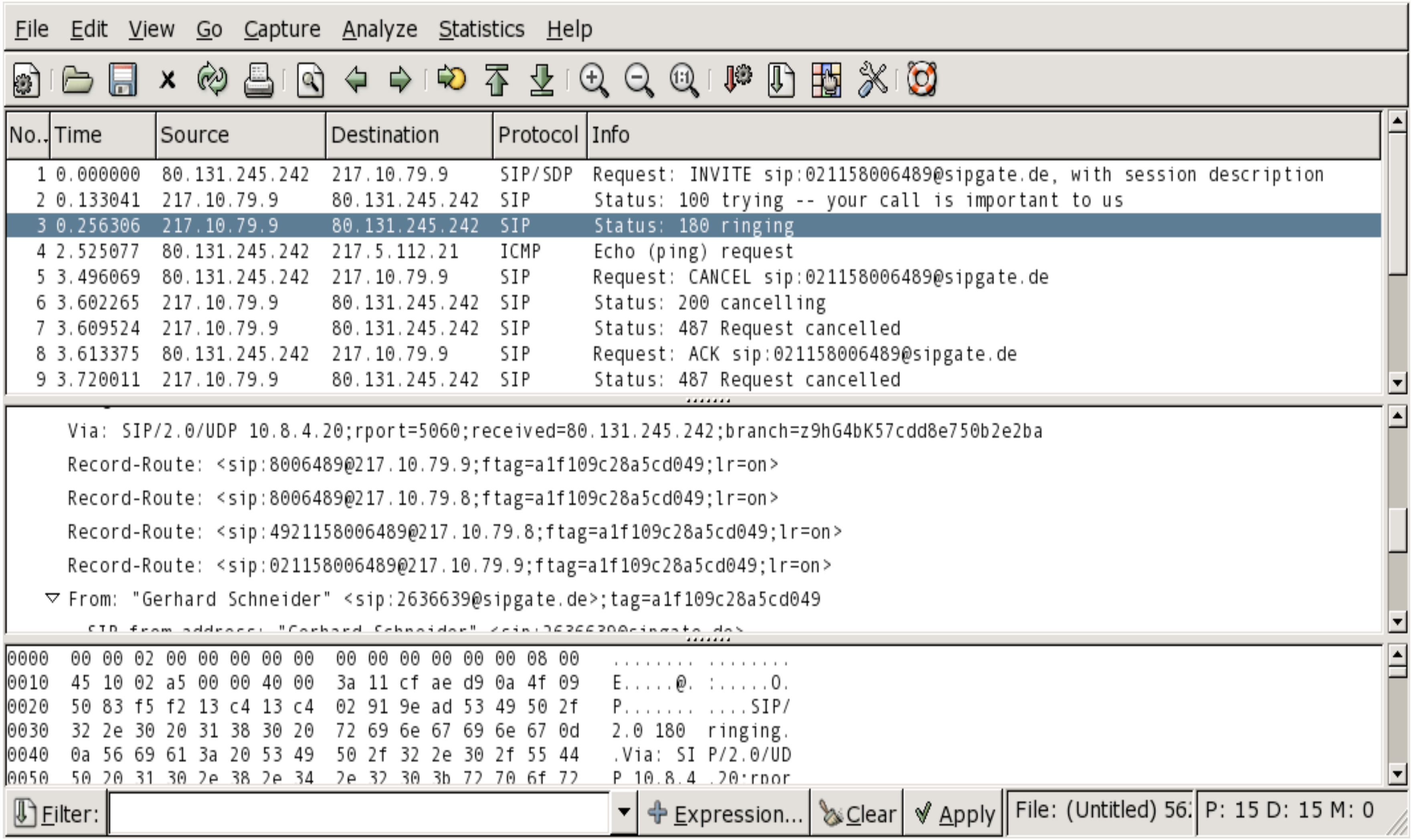


```
No.. Time      Source          Destination     Protocol Info
   1 0.000000  217.10.79.9     80.131.231.29   SIP/SDP  Request: INVITE sip:2636639@80.131.231.29:5060, with session description
   2 0.005955  80.131.231.29   217.10.79.9     SIP      Status: 100 trying
   3 0.008325  80.131.231.29   217.10.79.9     SIP      Status: 180 ringing
   4 0.057998  217.10.79.9     80.131.231.29   SIP/SDP  Request: INVITE sip:2636639@80.131.231.29:5060, with session description
   5 0.064726  80.131.231.29   217.10.79.9     SIP      Status: 180 ringing
   6 1.567673  80.131.231.29   217.5.112.21    ICMP     Echo (ping) request
   7 4.552620  80.131.231.29   217.10.79.9     SIP/SDP  Status: 200 OK, with session description
```

```
▷ Frame 2 (536 bytes on wire, 536 bytes captured)
▷ Linux cooked capture
▷ Internet Protocol, Src Addr: 80.131.231.29 (80.131.231.29), Dst Addr: 217.10.79.9 (217.10.79.9)
▷ User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
▽ Session Initiation Protocol
  ▽ Status-Line: SIP/2.0 100 trying
      Status-Code: 100
  ▽ Message Header
      Via: SIP/2.0/UDP 217.10.79.9;branch=z9hG4bKaca7.d1ee4543.0
      Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.ba6faa92.0
      Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.aa6faa92.0
      Via: SIP/2.0/UDP 217.10.64.86:5060;branch=z9hG4bK2affe745
    ▽ From: "09119374209" <sip:09119374209@217.10.64.86>;tag=as2c42d209
        SIP from address: "09119374209" <sip:09119374209@217.10.64.86>
        SIP tag: as2c42d209
      To: <sip:4920142636639@sipgate.net>
      Call-ID: 098d85e411b1bc0b2bf0900528510b31@217.10.64.86
      CSeq: 102 INVITE
      User-Agent: Grandstream 1.0.4.39
      Content-Length: 0
```

```
0000   00 04 02 00 00 00 00 00  00 00 00 00 00 00 08 00   ........ ........
0010   45 00 02 08 e1 17 00 00  f9 11 7f 18 50 83 e7 1d   E....... ....P...
0020   d9 0a 4f 09 13 c4 13 c4  01 f4 ae 75 53 49 50 2f   ..O..... ...uSIP/
0030   32 2e 30 20 31 30 30 20  74 72 79 69 6e 67 0d 0a   2.0 100  trying..
0040   56 69 61 3a 20 53 49 50  2f 32 2e 30 2f 55 44 50   Via: SIP /2.0/UDP
0050   20 32 31 37 2e 31 30 2e  37 39 2e 39 3b 62 72 61    217.10. 79.9;bra
0060   6e 63 68 3d 7a 39 68 47  34 62 4b 61 63 61 37 2e   nch=z9hG 4bKaca7.
0070   64 31 65 65 34 35 34 33  2e 30 0d 0a 56 69 61 3a   d1ee4543 .0..Via:
0080   20 53 49 50 2f 32 2e 30  2f 55 44 50 20 32 31 37    SIP/2.0 /UDP 217
0090   2e 31 30 2e 37 39 2e 38  3b 62 72 61 6e 63 68 3d   .10.79.8 ;branch=
00a0   7a 39 68 47 34 62 4b 61  63 61 37 2e 62 61 36 66   z9hG4bKa ca7.ba6f
```

SIP – “ringing message”



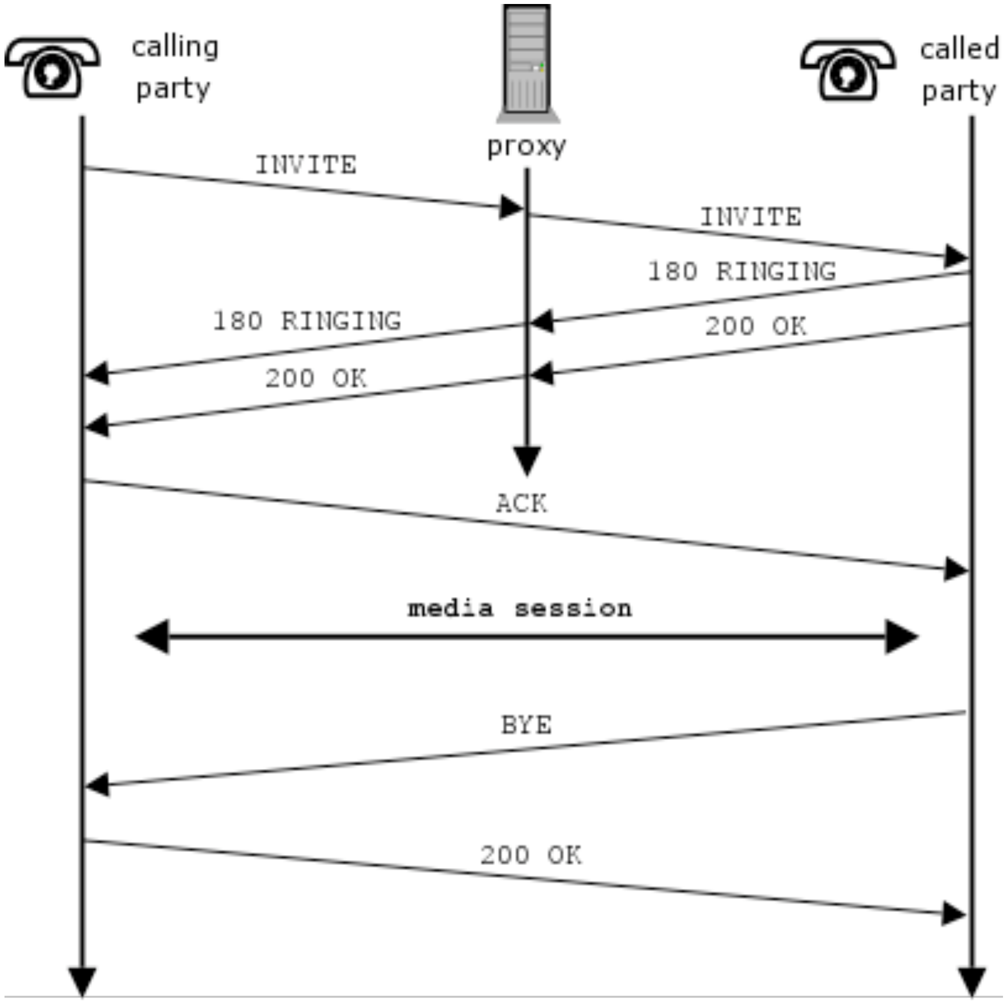
```
No.. Time      Source          Destination     Protocol Info
   1 0.000000  80.131.245.242  217.10.79.9     SIP/SDP  Request: INVITE sip:021158006489@sipgate.de, with session description
   2 0.133041  217.10.79.9     80.131.245.242  SIP      Status: 100 trying -- your call is important to us
   3 0.256306  217.10.79.9     80.131.245.242  SIP      Status: 180 ringing
   4 2.525077  80.131.245.242  217.5.112.21    ICMP     Echo (ping) request
   5 3.496069  80.131.245.242  217.10.79.9     SIP      Request: CANCEL sip:021158006489@sipgate.de
   6 3.602265  217.10.79.9     80.131.245.242  SIP      Status: 200 cancelling
   7 3.609524  217.10.79.9     80.131.245.242  SIP      Status: 487 Request cancelled
   8 3.613375  80.131.245.242  217.10.79.9     SIP      Request: ACK sip:021158006489@sipgate.de
   9 3.720011  217.10.79.9     80.131.245.242  SIP      Status: 487 Request cancelled
```

```
    Via: SIP/2.0/UDP 10.8.4.20;rport=5060;received=80.131.245.242;branch=z9hG4bK57cdd8e750b2e2ba
    Record-Route: <sip:8006489@217.10.79.9;ftag=a1f109c28a5cd049;lr=on>
    Record-Route: <sip:8006489@217.10.79.8;ftag=a1f109c28a5cd049;lr=on>
    Record-Route: <sip:4921158006489@217.10.79.8;ftag=a1f109c28a5cd049;lr=on>
    Record-Route: <sip:021158006489@217.10.79.9;ftag=a1f109c28a5cd049;lr=on>
  ▽ From: "Gerhard Schneider" <sip:2636639@sipgate.de>;tag=a1f109c28a5cd049
      SIP from address: "Gerhard Schneider" <sip:2636639@sipgate.de>
```

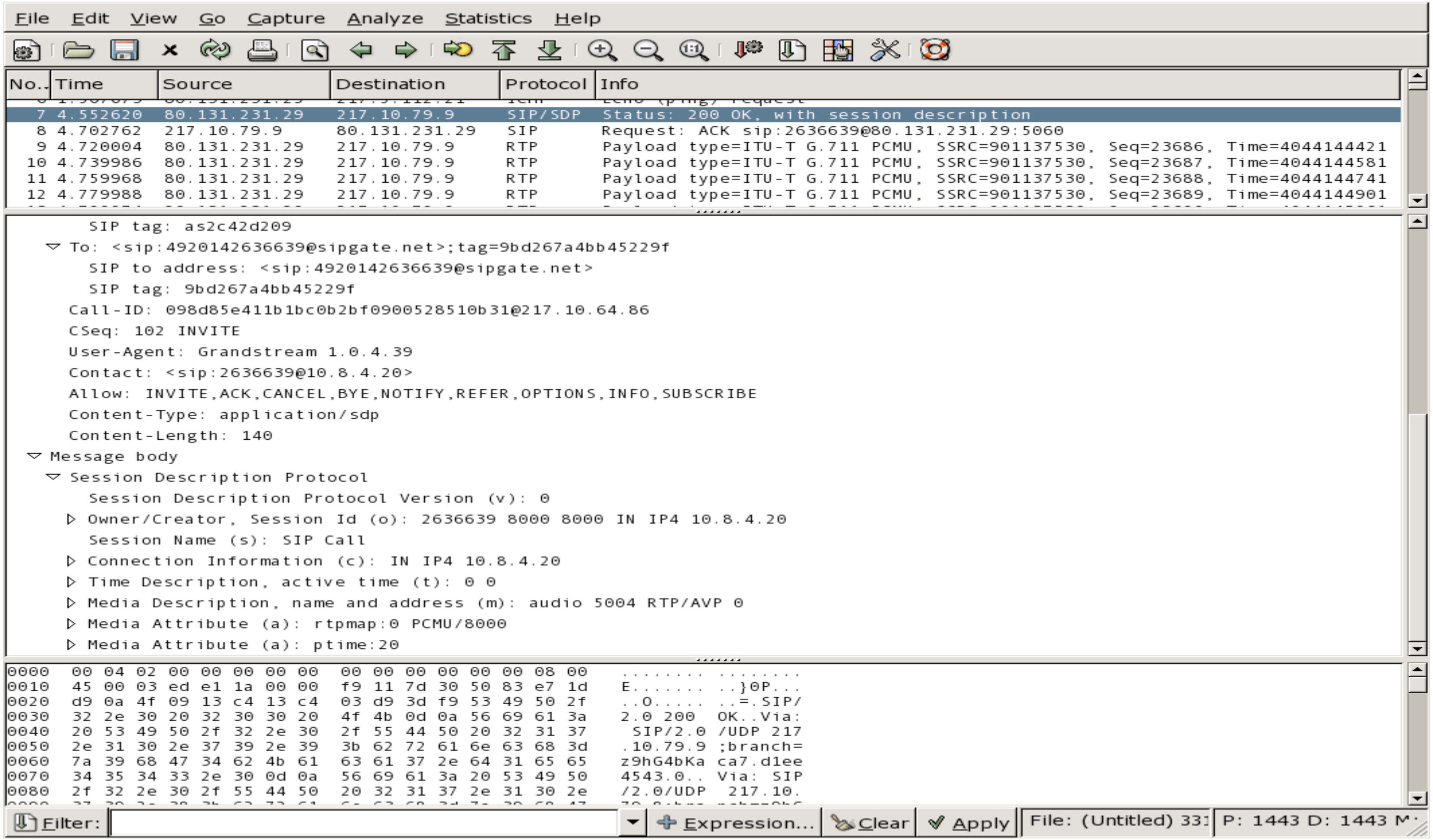
```
0000  00 00 02 00 00 00 00 00  00 00 00 00 00 00 08 00   ........ ........
0010  45 10 02 a5 00 00 40 00  3a 11 cf ae d9 0a 4f 09   E.....@. :.....O.
0020  50 83 f5 f2 13 c4 13 c4  02 91 9e ad 53 49 50 2f   P....... ....SIP/
0030  32 2e 30 20 31 38 30 20  72 69 6e 67 69 6e 67 0d   2.0 180  ringing.
0040  0a 56 69 61 3a 20 53 49  50 2f 32 2e 30 2f 55 44   .Via: SI P/2.0/UD
0050  50 20 31 30 2e 38 2e 34  2e 32 30 3b 72 70 6f 72   P 10.8.4 .20;rpor
```

SIP – “ringing” (cont.)

- To and From fields are the same as in INVITE
  - direction of the initiating request is important
- Connection over a proxy
  - only answers to requests, does not send requests by itself
  - no media abilities (does not handle media sessions)
  - reads header and does not analyse body+
- Proxy may send request for clients location to location server



SIP – OK (200) message



```
No.. Time      Source          Destination     Protocol  Info
  7 4.552620  80.131.231.29   217.10.79.9     SIP/SDP   Status: 200 OK, with session description
  8 4.702762  217.10.79.9     80.131.231.29   SIP       Request: ACK sip:2636639@80.131.231.29:5060
  9 4.720004  80.131.231.29   217.10.79.9     RTP       Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=23686, Time=4044144421
 10 4.739986  80.131.231.29   217.10.79.9     RTP       Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=23687, Time=4044144581
 11 4.759968  80.131.231.29   217.10.79.9     RTP       Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=23688, Time=4044144741
 12 4.779988  80.131.231.29   217.10.79.9     RTP       Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=23689, Time=4044144901
```

```
      SIP tag: as2c42d209
  ▽ To: <sip:4920142636639@sipgate.net>;tag=9bd267a4bb45229f
      SIP to address: <sip:4920142636639@sipgate.net>
      SIP tag: 9bd267a4bb45229f
    Call-ID: 098d85e411b1bc0b2bf0900528510b31@217.10.64.86
    CSeq: 102 INVITE
    User-Agent: Grandstream 1.0.4.39
    Contact: <sip:2636639@10.8.4.20>
    Allow: INVITE,ACK,CANCEL,BYE,NOTIFY,REFER,OPTIONS,INFO,SUBSCRIBE
    Content-Type: application/sdp
    Content-Length: 140
▽ Message body
  ▽ Session Description Protocol
      Session Description Protocol Version (v): 0
    ▷ Owner/Creator, Session Id (o): 2636639 8000 8000 IN IP4 10.8.4.20
      Session Name (s): SIP Call
    ▷ Connection Information (c): IN IP4 10.8.4.20
    ▷ Time Description, active time (t): 0 0
    ▷ Media Description, name and address (m): audio 5004 RTP/AVP 0
    ▷ Media Attribute (a): rtpmap:0 PCMU/8000
    ▷ Media Attribute (a): ptime:20
```

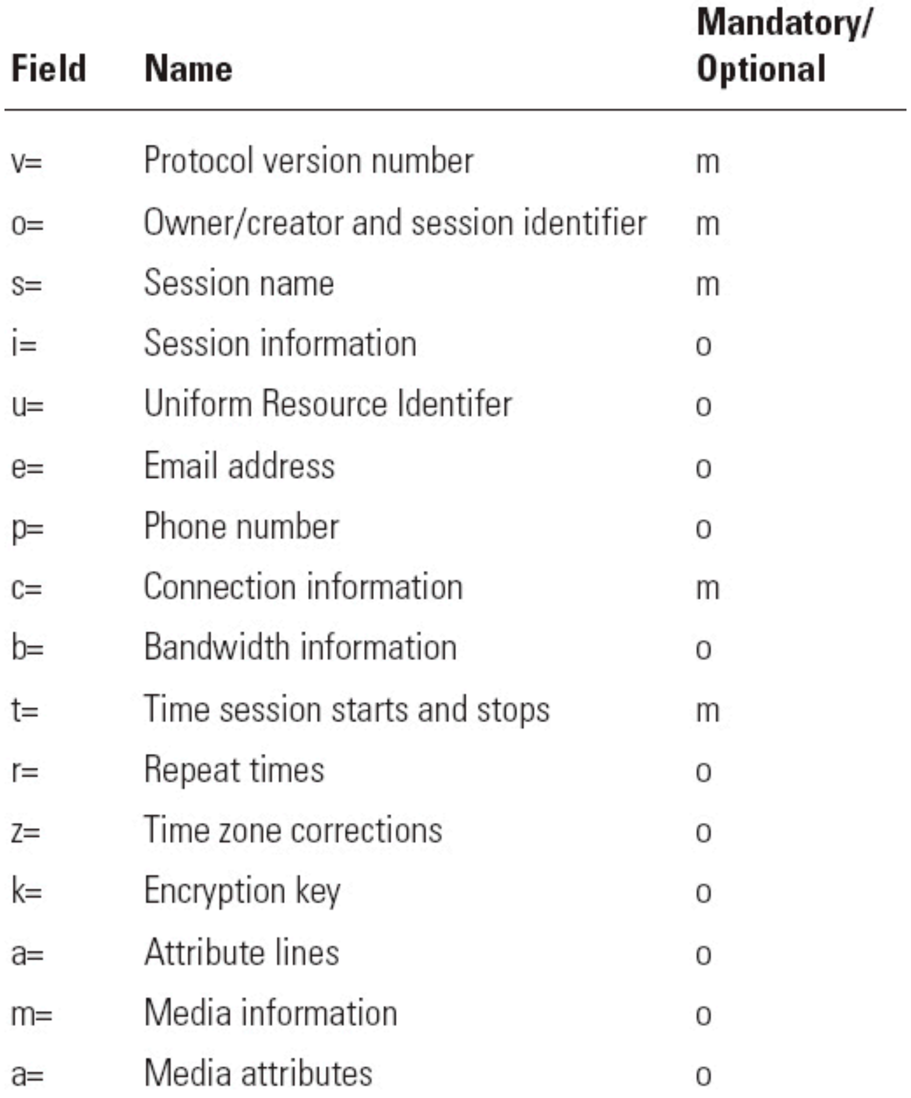
```
0000  00 04 02 00 00 00 00 00  00 00 00 00 00 00 08 00   ........ ........
0010  45 00 03 ed e1 1a 00 00  f9 11 7d 30 50 83 e7 1d   E....... ..}0P...
0020  d9 0a 4f 09 13 c4 13 c4  03 d9 3d f9 53 49 50 2f   ..O..... ..=.SIP/
0030  32 2e 30 20 32 30 30 20  4f 4b 0d 0a 56 69 61 3a   2.0 200  OK..Via:
0040  20 53 49 50 2f 32 2e 30  2f 55 44 50 20 32 31 37    SIP/2.0 /UDP 217
0050  2e 31 30 2e 37 39 2e 39  3b 62 72 61 6e 63 68 3d   .10.79.9 ;branch=
0060  7a 39 68 47 34 62 4b 61  63 61 37 2e 64 31 65 65   z9hG4bKa ca7.d1ee
0070  34 35 34 33 2e 30 0d 0a  56 69 61 3a 20 53 49 50   4543.0.. Via: SIP
0080  2f 32 2e 30 2f 55 44 50  20 32 31 37 2e 31 30 2e   /2.0/UDP  217.10.
```

SIP – redirect, registering & instant messaging

- Redirection
  - client sends INVITE to the SIP redirect server
  - redirect server sends a request to the location server or requests the IP of the client to call
  - current data is sent to the client, which ACK's
  - from now on further on like direct connection
- Registration
  - REGISTER message to SIP registration server
  - binding of the SIP URI with IP the users client/machine
  - 200 OK
- Instant messaging like the wellknown tools in that sector
  - online status, buddy lists ...

SDP – service description protocol

- Session Description Protocol (SDP)
  - IETF standard RFC 2327
  - text coded like SIP
  - description syntax
- But unclean design
  - IP layer information on higher protocol levels



Field	Name	Mandatory/ Optional
v=	Protocol version number	m
o=	Owner/creator and session identifier	m
s=	Session name	m
i=	Session information	o
u=	Uniform Resource Identifer	o
e=	Email address	o
p=	Phone number	o
c=	Connection information	m
b=	Bandwidth information	o
t=	Time session starts and stops	m
r=	Repeat times	o
z=	Time zone corrections	o
k=	Encryption key	o
a=	Attribute lines	o
m=	Media information	o
a=	Media attributes	o

SDP – service description protocol

- example:
  ```
  v=0
  o=calling 2890844526 2890844526 IN IP4 10.8.4.254
  s=Phone Call
  c=IN IP4 100.101.102.103
  t=0
  m=audio 49170 RTP/AVP
  a=rtpmap:0 PCMU/8000
  ```
- Version is 0 (at the moment no other versions available)
- Origin o=username session-id version network-type adress-type adress
- Subject s=subject

SDP – service description protocol (cont.)

- Connection Data c=network-type address-type connection-adress
- Time t=start-time stop-time
- Media Announcements m=media port transport format-list
- Attributes a=…
- This setup defines the multimedia session
  - which usually uses RTP / RTCP

SIP – firewalls, NAT, ...

- NAT
  - SIP messages contain IP addresses in the data segments of its packets
  - internal network addresses from the NATted network are not visible from the „outside“ world
  - A calls B, B gets the message from A, but not vice versa
  - problem could be solved with a proxy server sitting in the internal and external LAN
- Firewalls
  - RTP does not use fixed layer 4 port numbers
  - variable in the range of 1024 - 65534

SIP – firewalls, NAT, ... (cont.)

- stun protocol
  - simple traversal of UDP through NATs
  - returning public's IP port
  - can help to determine which kind of NAT is used
  - most clients implement that protocol to produce the relevant SDP messages
  - stun server will send its response to the IP:port the initial packet was sent to
    - if change-ip flag, then sends from different IP
    - if change-port flag from different port

Literature/End

- Kurose & Ross: Computer Networking - Section on SIP
- Tanenbaum: Computer Networks, Section on Voice over IP
- Plenty of online resources

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