Communication Systems

SIP
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)
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)
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)
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)
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 one 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
    - authentication, authorization
    - secure routing and rerouting
SIP – server

- SIP server
  - redirect server = information service
  - location server is the request address for the host on which 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 and ending

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

```
Session Initiation Protocol
Request-Line: INVITE sip:2636639@0.131.231.29:5060 SIP/2.0
Method: INVITE

Message Header
Record-Route: <sip:2636639@0.131.231.29:5060 SIP/2.0>
Record-Route: <sip:2636639@0.131.231.29:5060 SIP/2.0>
Max-Forwards: 8
Record-Route: <sip:4920142636039@0.131.231.29:5060 SIP/2.0>
Via: SIP/2.0/UDP 217.10.79.9;branch=z9hG4bKaca7.dea4543.B
Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.dea4543.B
Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.a64faa892.B
Via: SIP/2.0/UDP 217.10.64.66:5868;branch=z9hG4bK2affef745
From: "9911374269" <sip:9911374269@217.10.64.66>;tag=as2c42d09
SIP from address: "9911374269" <sip:9911374269@217.10.64.66>
SIP tag: as2c42d09
To: <sip:4920142636039@sipgate.net>
Contact: <sip:9911374269@217.10.64.66>
Call-ID: 99055e4b1b2c0b20f99055e28516a31@217.10.64.66
CSeq: 192 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
```

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)
SIP – “ringing message”

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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**

<table>
<thead>
<tr>
<th>No.</th>
<th>Time</th>
<th>Source</th>
<th>Destination</th>
<th>Protocol</th>
<th>Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>4:742762</td>
<td>121.16.70.9</td>
<td>180.131.231.29</td>
<td>SIP</td>
<td>Request ACK sip:2636635680:131.231.29:5668</td>
</tr>
<tr>
<td>2</td>
<td>4:720684</td>
<td>180.131.231.29</td>
<td>217.10.70.9</td>
<td>RTP</td>
<td>Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=22668, Time=404144421</td>
</tr>
<tr>
<td>3</td>
<td>4:739956</td>
<td>180.131.231.29</td>
<td>217.10.70.9</td>
<td>RTP</td>
<td>Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=22667, Time=404144451</td>
</tr>
<tr>
<td>4</td>
<td>4:729956</td>
<td>180.131.231.29</td>
<td>217.10.70.9</td>
<td>RTP</td>
<td>Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=22666, Time=404144471</td>
</tr>
</tbody>
</table>

**SIP tag: a2c42d709**

- To: <sip:920142636639@sipgate.net>; tag=9202674db45229f
- SIP to address: <sip:920142636639@sipgate.net>
- SIP tag: 9bd207a4bb45229f
- Call-ID: 99bd556e11b11bcb8b2f0506528510631e17.10.64.86
- CSeq: 162 INVITE
- User-Agent: Grandstream 1.8.4.39
- Contact: <sip:2636635680:131.231.29:5668>; b=rrc=1.0
- 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): 2636635680 8888 8888 IN IP4 10.8.4.20
  - Session Name (n): SIP Call
  - Connection Information (c): IN IP4 10.8.4.20
  - Time Description, active time (t): 0
  - Media Description, name and address (m): audio 5604 RTP/AVP 0
  - Media Attribute (a): rtpmap:0 PCMU/6000
  - Media Attribute (a): ptime:20

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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

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