Dialogue
Systems

Luděk Bártek

User –
Dialogue
System Com-
munication

VoIP

SIP

Dialogue Systems

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- Voice:
  - mostly communicate using telephone network PSTN, VoIP).
  - Voice digitization is performed:
    - On the user side – VoIP communication.
    - On the side of PBX – DS uses VoIP, user PSTN.
    - On the DS side – As well as user and the DS uses PSTN.
  - Speech recognition is mostly performed on the DS side.
    - When to use speech recognition on the client side?
    - What can be advantages of speech recognition on the client side?

- Text:
    - User communicates either using a specialized client or using some of the common TCP/IP protocols.
    - There is no need of speech recognition
    - Used mostly at development and debugging.
- Voice+Text
    - communication to DS
        - VoIP – text using a DTMF (like writing SMS).
        - special client.

- VoIP – family of protocols to control voice communication over Internet (IP based network).
- Used in VoIP.
- Used protocols:
  - UDP (ISO/OSI transport layer):
    - Transmits packets over network between two endpoints.
    - Does not guarantee neither packet delivery nor ordering.
    - Advantages – low data transfer costs.
    - Disadvantages – possible data lost and packet deliver speed jitter.
  - RTP (ISO/OSI session Layer):
    - Used to transmit multimedia data..
    - Guarantee data delivery..
    - Allows to define transmission parameters – guarantee the packet delivery speed.

- VoIP – many implementations.
- Differs in:
  - used standards
    - H.323 (not commonly used, standard ITU, complex, relatively complicated)
    - SIP (simpler H.323 replacement, commonly used in present time)
    - proprietary – Skinny (Cisco), HFA (Siemens), . . .
  - offered services – telephony, TV (DVB), fax, message transmission, . . .
  - signalization – depends on selected standard and used protocols.
  - . . .

Session Initiation Protocol (SIP)

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- VoIP signalization control protocol on application ISO/OSI layer.
- Client–server text protocol offering:
  - call forwarding
  - participant identification
  - personal mobility
  - both parties authentication
  - conference call support using multicast.
  - . . .

SIP

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- Participant identification – URI in the form
  sip:caller_id@server_address
    - caller_id – ID assigned to the user by his PBX
    - server address – PBX address (FQDN/IP), where is the user registered.
- SIP session may be:
    - direct – formed by communicating parties directly
    - using SIP proxy server(s) – serves as participants registrators.

SIP Actions

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- Participant localization – using his SIP ID
- Checking participant state – ready to receive call vs. busy/forwarded
- Checking participant communication possibilities – available codecs, bitrates, audio/video support, . . .
- Connection establishing – uses the SDP protocol.
    - describes formed connection,
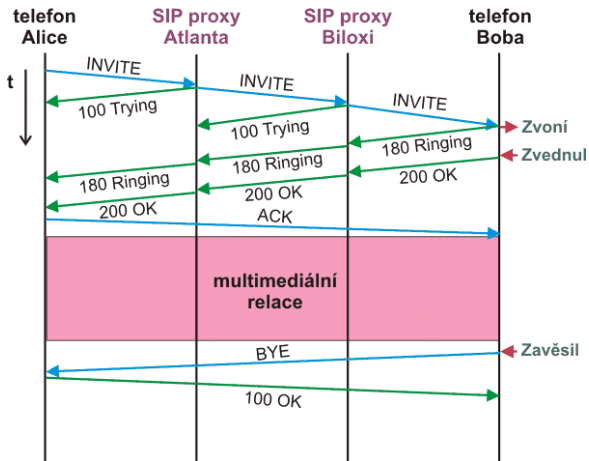   - references RTP/UDP data flow used to participants communication.

SIP Session Control

Obrázek: See Wikipedia