Outline (Preliminary)

10 Signaling Protocols for Multimedia Communication

10.1 Signaling and Sessions

10.2 SIP Basics *

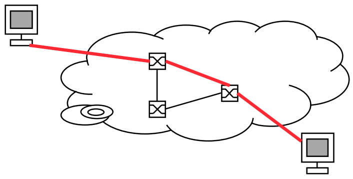
Literature:

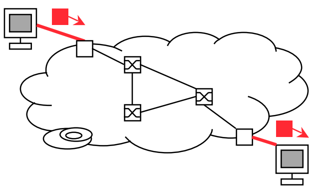
Stephan Rupp, Gerd Siegmund, Wolfgang Lautenschlager:
SIP – Multimediale Dienste im Internet, dpunkt.Verlag 2002

* Hinweis: Überlappung mit „Rechnernetze II“ (Hegering)

Communication networks

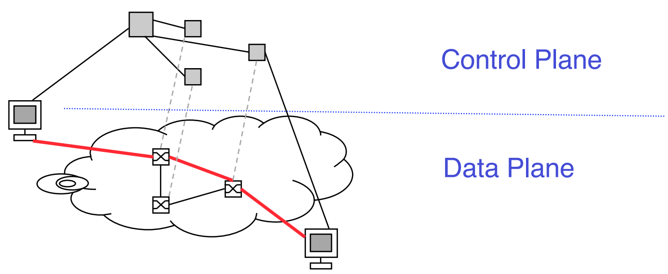
- Classification of communication networks:
  - Circuit-switched *(Leitungsvermittlung)*: Physical connection between communicating end systems (for limited duration)
    - » Traditional telephone networks
    - » *Virtual connections* in advanced digital networks (e.g. ATM)
  - Packet-switched *(Paketvermittlung)*: Transmission of packets to addressed end system
    - » Internet Protocol (IP)





Control Plane and Data Plane

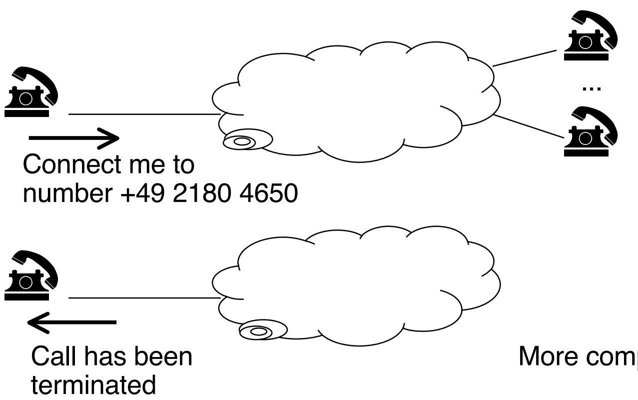
- Classification of network mechanisms:
  - *Control Plane:* Mechanisms of the network to establish, modify and remove connections
  - *Data Plane:* Mechanisms of the network to transmit data over established connections
  - Strict separation of Control and Data planes in traditional telephone networks (e.g. ISDN)



Signaling

- *Signaling (Signalisierung, Zeichengabe)* originates from circuit-switched networks
- Signaling = Protocols of the Control Plane
  - User-to-Network Signaling: From end system to network interface
  - Network-to-Network Signaling: From one network node to another network node
  - End-to-End Signaling: From one end system to another end system
- Examples:
  - Call setup in ISDN
  - Call setup in ATM (Q.2931)
  - Resource reservation in IP networks (RSVP)

Signaling in Telephone Networks



More complex signaling:

Add 3rd party to call
Forward incoming calls
Route calls according to
time and origin
...

Call Control and Bearer Control

- Signaling can be further separated in
  - *Call Control (Rufsteuerung):*
    - » Determining the partners to be connected
    - » Defining properties of connections
    - » Logical establishment of connection
  - *Bearer Control (Wegbereitstellung):*
    - » Determining the actual route in the network
    - » Establishment of connections in the network
- Call Control is relatively independent of network technology
- Bearer Control always depends heavily on the network technology

Signaling and the Internet – Why?

- *Convergence* of network technologies
  - To establish phone conversations over the Internet *(IP telephony)*
    - » Phone sets interconnected through the Internet
    - » Mixed conversation, e.g. calling a normal phone from a PC
    - » Gateways Internet/Telephone networks
  - To support Bearer Control in the Internet
    - » E.g. by sophisticated resource management
    - » *Quality-of-Service* support
- On plain Internet:
  - Support of mobility
    - » User mobility: Forwarding to dynamically changing end system
    - » Terminal mobility: Forwarding traffic to end system in dynamically changing network location
    - » Service mobility: Support for services from foreign networks
  - To provide information on *status* of user or terminal (e.g. online/offline)

Signaling and the Internet – How?

- Internet is based on packet-switching
  - Classical Internet does not provide the concept of routes
  - Bearer control cannot be realized in plain Internet
- Signaling
  - Either restricted to Call Control
    - » Just informing the end systems of their current state
    - » SIP is essentially Call Control
  - Or involving advanced network features
    - » Support for Quality of Service
    - » E.g. by adjusting resources in routers
    - » E.g. driven by the RSVP resource reservation protocol

Session

- *Session:*
  - Information about the partners in a communication activity and the connections existing among them, including the characteristic properties of party participation and connections (important for multimedia sessions)
  - A session exists only for a limited period of time, typically ranging between several seconds and several hours
- Examples:
  - Video on Demand Service
    - » Partners: Server, User terminal
    - » Connections:
      (a) Control connection (bidirectional, low bandwidth)
      (b) Video transfer connection (unidirectional, high bandwidth)
  - Videoconference Service
    - » Partners: *n* User terminals (one is *master*)
    - » Connections:
      (a) e.g. one control connection per partner to master (*n* connections)
      (b) fully meshed A/V connections between partners ($O(n^2)$ connections)

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SIP - The Context

- SIP = *Session Initiation Protocol*,
  standardized by IETF *(Internet Engineering Task Force)*
  - Signaling protocol independent of underlying network technology
  - Text-based client/server protocol, similar to HTTP
  - Covers broad range from traditional telephony to multimedia conferencing
  - Peer-to-peer style architecture:
    - » Client contains *User Agent* (UA) in client and server roles (UAC, UAS)
- Developed based on proposals by Mark Handley and Henning Schulzrinne, 1999
- Related other protocols:
  - SDP = *Session Description Protocol*
  - SAP = *Session Announcement Protocol*
  - SCCP = *Simple Conference Control Protocol*
  - RTSP = *Real Time Streaming Protocol*
  - RTP = *Real Time Transport Protocol*
- *MMUSIC = Multiparty Multimedia Session Control*

Main Features & Components of SIP

- SIP Proxy Servers for forwarding of control messages
  - Including “redirect” and “location” servers
- Support of user, terminal and service mobility
- Gateways to traditional networks (e.g. telephone networks)
  - Including services of the so-called “Intelligent Network” (IN), i.e. advanced network features
- Status observation for users and terminals (e.g. online/offline, busy/free)
- Service creation and execution tools
  - Call Processing Language CPL
  - XML-Scripts in SIP server
  - SIP-Java-Servlets

- In the following: Focus (first) on audio connections = “IP telephony”

Adressing in SIP

- SIP supports Email-style addresses as well as addresses based on phone numbers
  - ITU standard for international phone number format: E.164
- Email style addresses:
  `sip:Heinrich.Hussmann@ifi.lmu.de`
  `sip:hussmann@cip.ifi.lmu.de`
- IP-based addresses:
  `sip:hussmann@141.84.8.6`
- Phone number style addresses:
  `sip:+49-89-2180-4650@net2phone.com`
- Mapping of E.164 telephone numbers to IP domain names
  - +49-89-2180-4650 is mapped to domain name 0.5.6.4.0.8.1.2.9.8.9.4.E164.arpa

SIP Terminals

- PCs with SIP-enabled applications
  - For audio connections: “softphones”
- Special phone sets
- Special mobile devices

Mitel 5055 SIP Phone

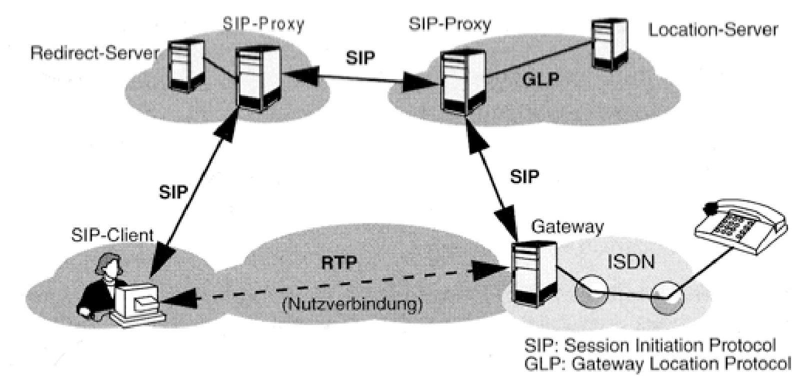
The Mitel 5055 SIP Phone is a full-featured, standards-based business telephone that delivers superior audio quality and session initiation protocol (SIP) services to the end-user's desktop. The 5055 SIP Phone is a versatile, highly interoperable phone that can function as a standalone product connected to a hosted solution, as part of a Mitel communications solution, or in PBX environments that support SIP. As a SIP-compliant appliance, it is interoperable with all voice, data, video and Internet applications and services that are SIP-enabled and/or provide full SIP protocol support.

SIP Servers

- Each SIP system can act as a SIP client *(User Agent Client, UAC)* or as a SIP server *(User Agent Server, UAS)*
- Functions of a SIP server:
  - Registration of SIP terminals
  - Registration of users including their profiles
  - Authentication, authorization and accounting (AAA)
  - Determination of end address (mapping of symbolic to current physical address)
  - Forwarding of requests
  - Call control (e.g. suspend and resume of connections)
  - Collecting and presenting information of user presence
  - Forwarding of QoS requests to network elements

Proxy Servers

- Proxy servers act on behalf of other terminals
- Proxy servers may modify SIP messages (headers)
- Proxy servers route SIP messages over the network



SIP Messages

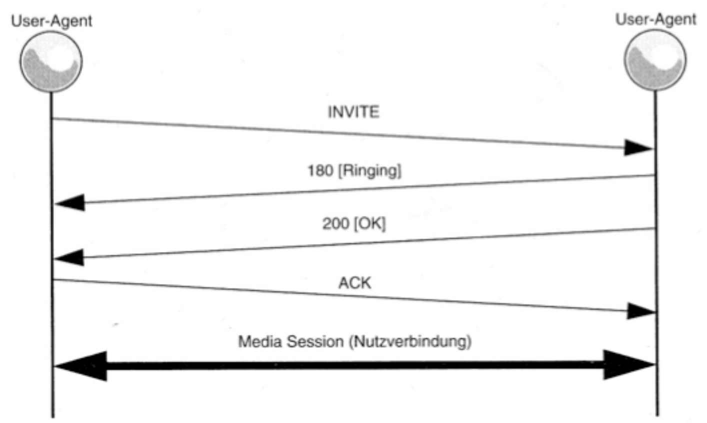
- Text-based peer-to-peer protocol
- Modelled after HTTP
  - *Header* contains connection parameters and service information
  - *Body* contains description of connection (using *Session Description Protocol* SDP)
- Requests:
  - From client (agent) to server (agent)
  - INVITE, BYE, OPTIONS, STATUS, CANCEL, ACK, REGISTER, …
- Responses:
  - Status information, e.g.
    - » Informational: 100 Trying, 180 Ringing, 181 Call is forwarded, …
    - » Success: 200 OK
    - » Redirection: 300 Multiple Choices, 301 Moved Permanently, …
    - » Client Error: 400 Bad Request, 404 Not Found, 486 Busy Here, …
    - » Server Error: 500 Internal Server Error, 504 Gateway Timeout, …

Example: SIP Message

`INVITE sip:john@domain.com SIP/2.0`	***Start Line***
`VIA:SIP/2.0/UDP 169.130.12.5`	***General Header***
`Call-ID:187602141351@worchester.bell-telephone.com`	
`From:<sip:a.g.bell@bell-telephone.com>`	
`To:T.A.Watson<sip:watson@bell-telephone.com>`	
`CSeq:1 INVITE`	*Sequence Number*
`Subject:Mr. Watson, come here`	***Request Header***
`Content-Type:application/sdp`	***Entity Header***
`Content-Length:885`	
`v=0`	***Body: SDP Data***
`o=bell 536557652353687637 IN IP4 128.3.4.5`	
`c=IN IP4 135.180.144.94`	
`m=audio 3456 RTP/AVP 0 3 4 5`	

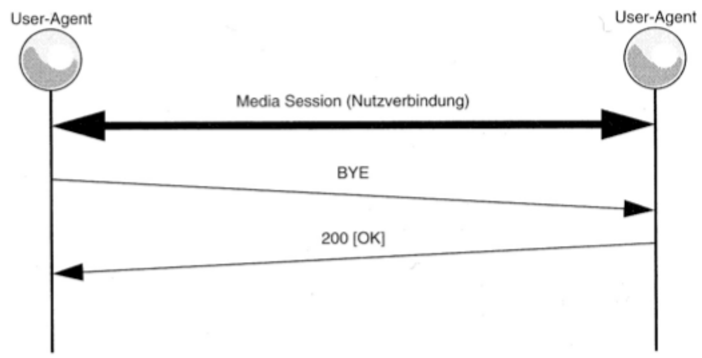
Call Setup

- Direct connection establishment between two SIP terminals (left: UAC, right: UAS)



Call Release

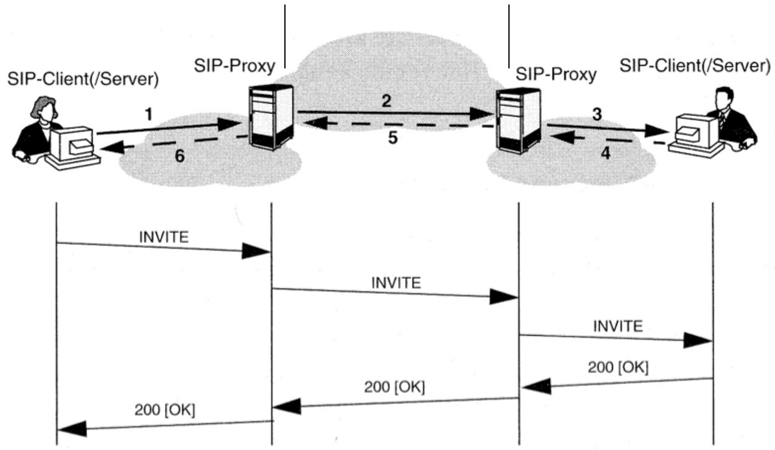
- Teardown of the connection initiated by client (initiator)



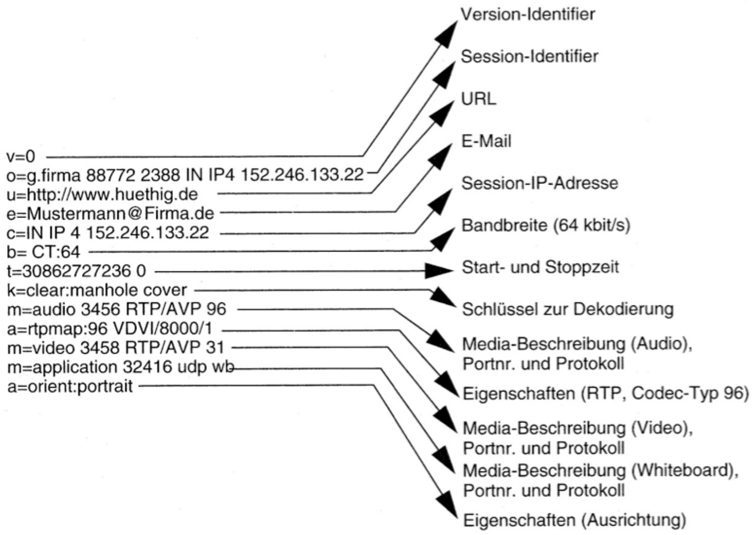
Proxies, Redirect and Location Server

May contact *Redirect server*

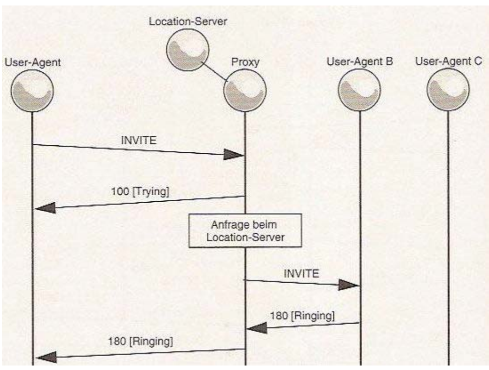
May contact *Location server*



SDP Information



Example: Parallel Call Forking (e.g. Call Center)



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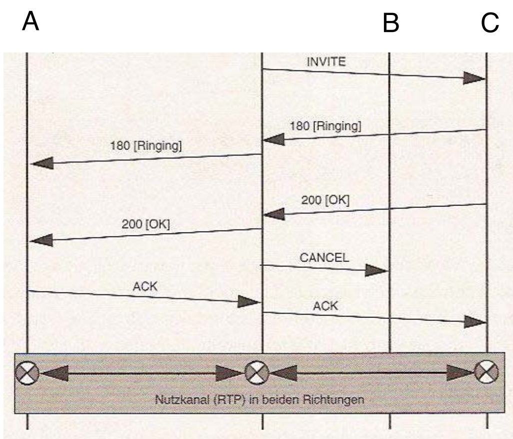
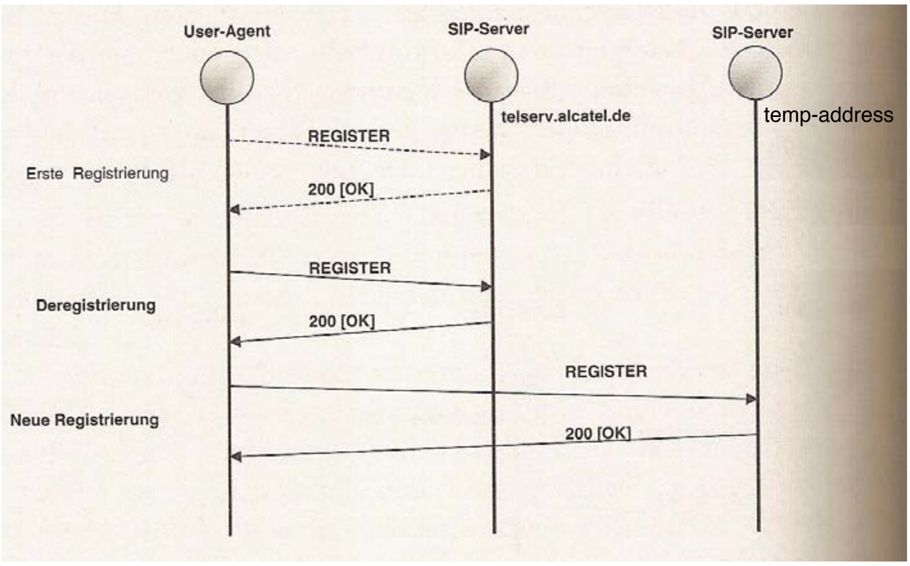


Example: Personal Number

- Incoming call for personal number triggers selection software in proxy server
- Mon–Fri 8–18:
  - Laptop online? If yes: Call there
  - If not: Mobile phone online? If yes: Call there
  - If not: Desktop computer active? If yes: Call there
  - If not: Call office phone with time limit
  - If time limit exceeded: Send email to office email address
- Mon–Fri 18–8 and Sat/Sun:
  - Send email to private email address and send SMS to mobile phone number

- *Service creation:* Developing service logic programs like above
  - In traditional telephone networks: “Intelligent network” (IN)

Example: Mobile User Registration



SIP and UMTS

- UMTS = Third generation of cellular mobile network
  - (1st: Analog, 2nd: GSM)
  - UMTS provides unique standard for Europe, USA (IMT-2000) and Japan "3rd Generation Partnership Project" 3GPP
- UMTS covers pico cells, urban cells, suburban cells, global cells
- UMTS Phase 1: New radio access to GSM core network
- UMTS Phase 2 ("Release 4/5"): Mobile multimedia system with new core network
  - IP based core network
  - Separation between call control and bearer control in Release 4
  - "Internet Multimedia Subsystem" (IMS) in Release 5:
    Call control over SIP only