

13 Signaling Protocols for Multimedia Communication

13.1 Signaling and Sessions

13.2 SIP Basics

13.3 Signaling for Instant Messaging

Literature:

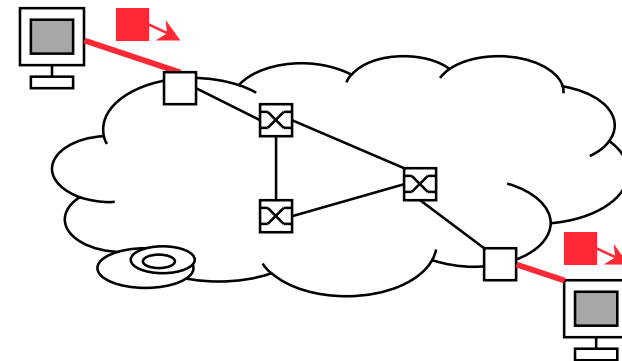
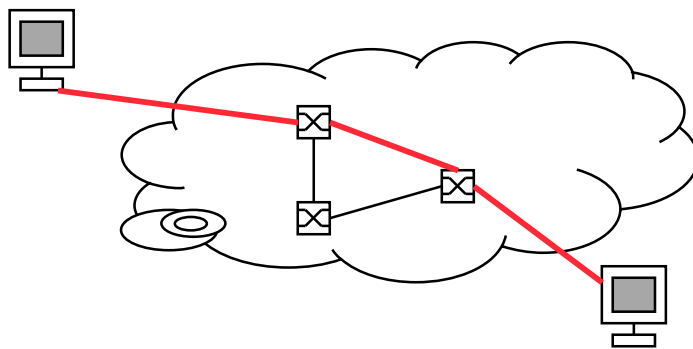
Stephan Rupp, Gerd Siegmund, Wolfgang Lautenschlager:
SIP – Multimediale Dienste im Internet, dpunkt.Verlag 2002

Outline

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- Part III:
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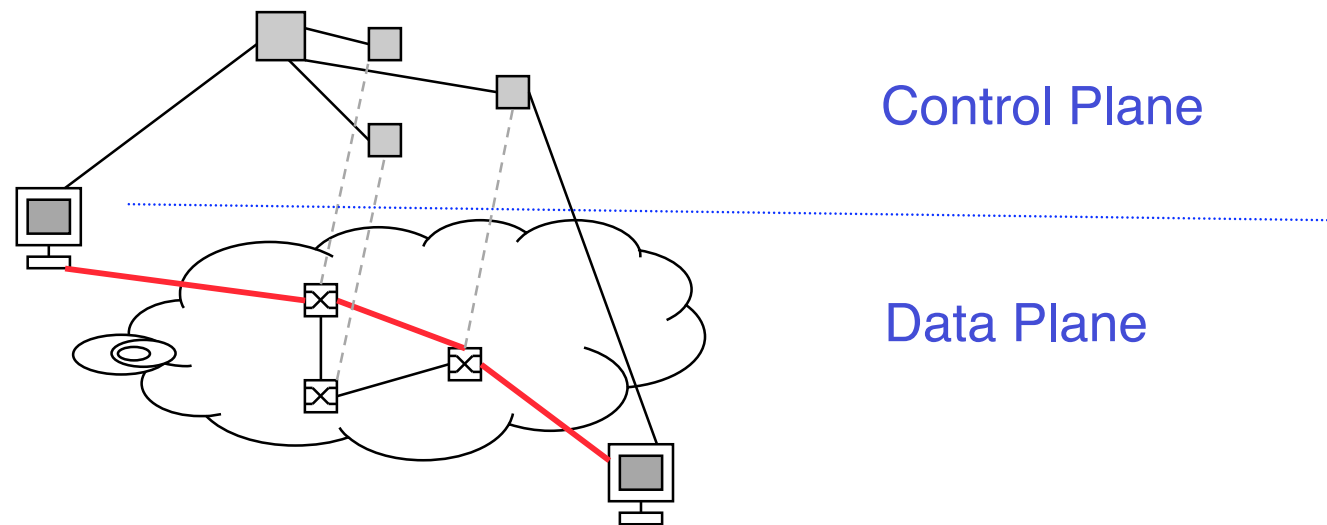
Communication networks

- 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

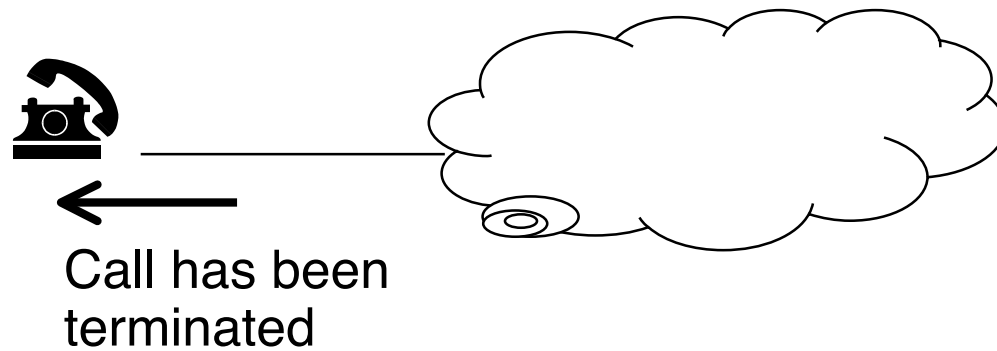
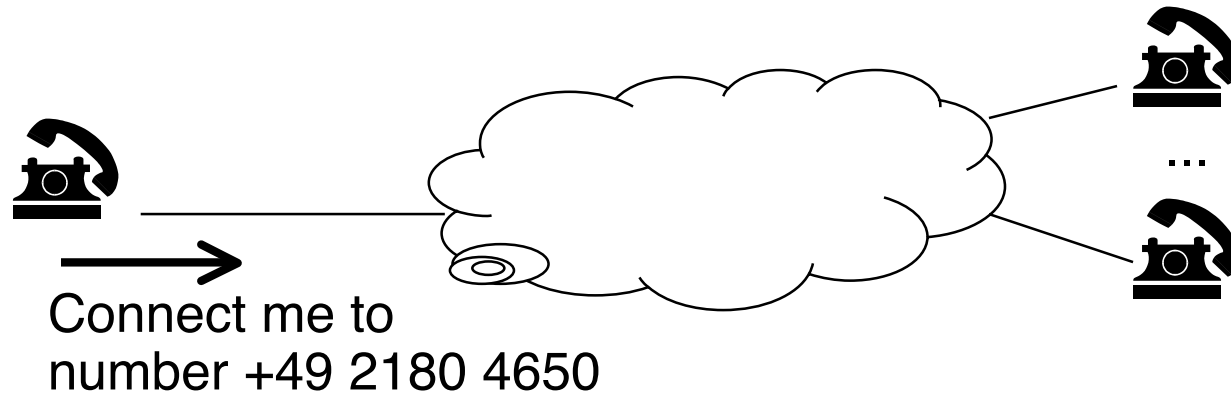
- Classification of network mechanisms:
 - *Control Plane*: Mechanisms of the network to establish, modify and remove connections
 - *Data Plane*: Mechanisms of the network to transmit data over established connections
 - Strict separation of Control and Data planes in traditional telephone networks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

Network Architecture for Multimedia Conferences

- Session control:
 - Unit managing participants of a (conference) session
 - Management of involved connections
 - Monitoring of quality
- Signalling:
 - In particular call control:
 - » How does a participant set up/join/tear down a session?
 - Negotiation of capabilities among clients
 - Addition to network traffic situation
 - Advanced features (like multiple calls, intelligent forwarding)

Network Architecture Option 1: Skype Based

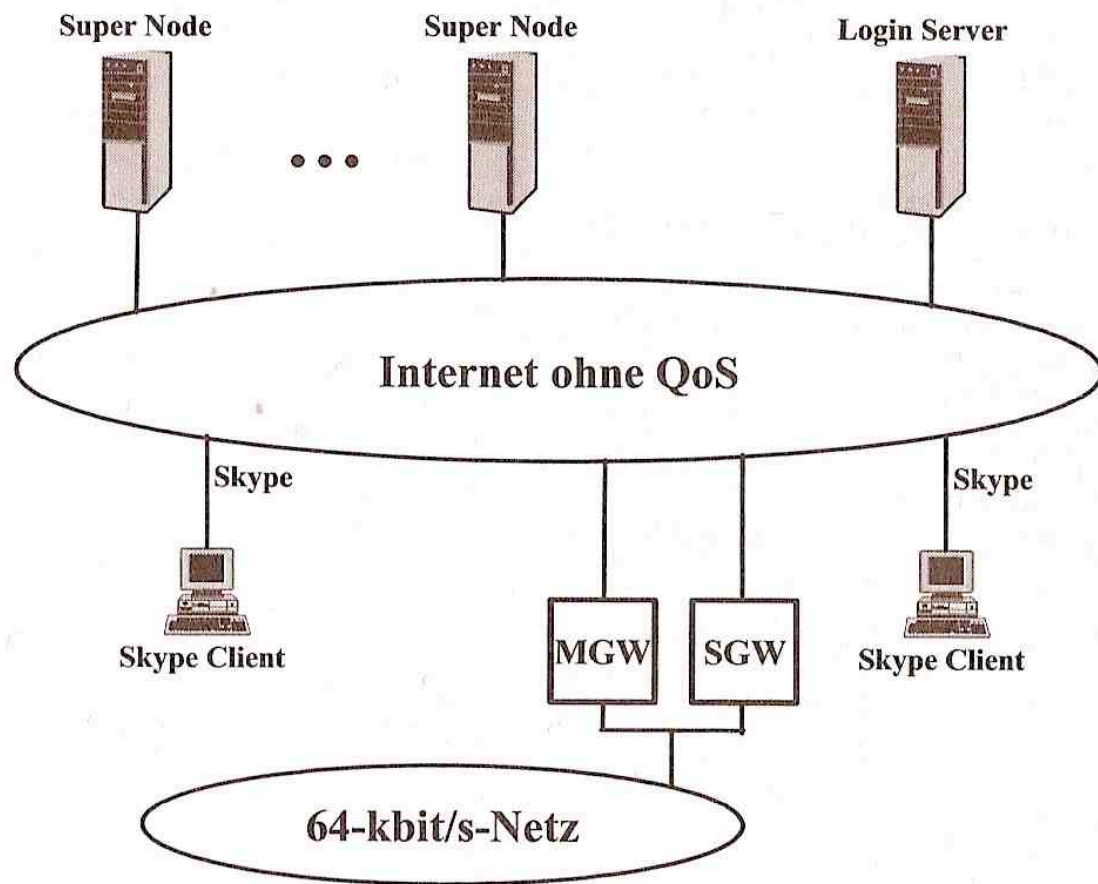


Bild 3.6: Skype für die Session-Steuerung

Trick/Weber

Skype Based Architecture

- Based on KaZaA peer-to-peer file sharing architecture
- Central *Login Server* for authentication
- Many *Super Nodes* form distributed database for user profiles
 - Powerful client computers with fixed address
- Steps in a Skype session:
 - User logs in (Login Server)
 - Client searches for Super Nodes and connects to a Super Node
 - Client gets address of communication partner from Super Node and establishes direct (peer-to-peer) communication link
 - Voice transmission: via UDP, adaptive between 24 and 128 kbit/s
 - » Predictive codecs: iSAC (LPC based), SILK (hybrid predictive/synth.)
 - Encryption of transmitted data
 - » Using AES 256 bit, key exchange through RSA
- Signalling and detailed architecture fully proprietary

IP Telephony and H.323

- ITU-T H.323 series of recommendations (standards)
 - Used as a synonym for a large group of ITU-T standards
 - » H.235 (security), H.450 (supplementary services), ..., but also RTP, RTCP, ...
 - Originally developed for videoconferencing, see next chapter
 - Works also with IP networks
 - Recent versions stress telephony applications (IP telephony)
- Definition of various gateways:
 - To PSTN/voice
 - To PSTN/fax
 - To PSTN/H.324 videophone
 - To GSM mobile phone
 - To private phone exchanges (“PBX”)
- Competition between H.323 and SIP about signalling for IP telephony (and multimedia conferencing, see later...)

Network Architecture Option 2: H.32X Based

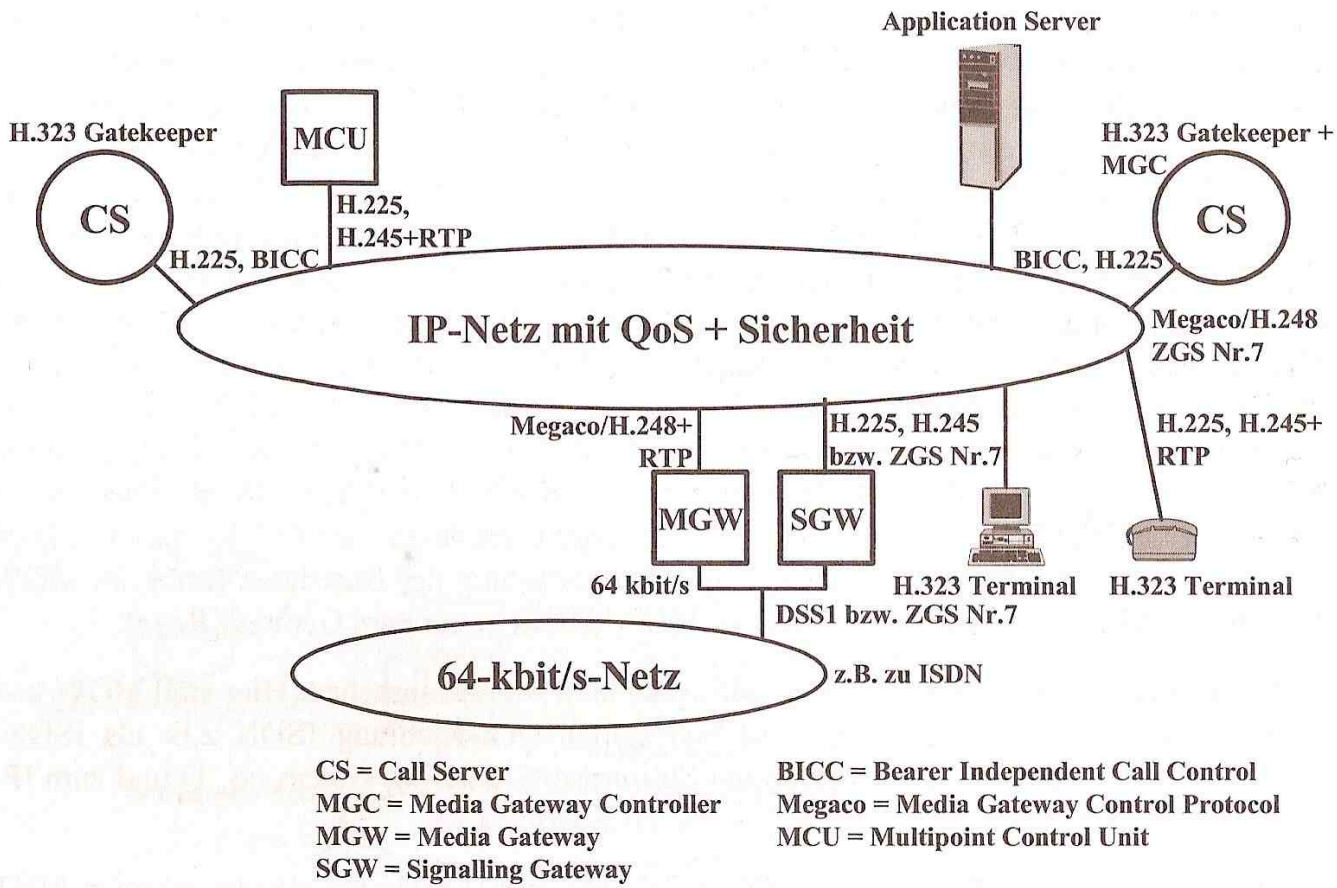


Bild 3.4: Protokolle und Netzarchitektur für Next Generation Networks mit H.323 für die Session-Steuerung

Trick/Weber

Network Architecture Option 3: SIP Based

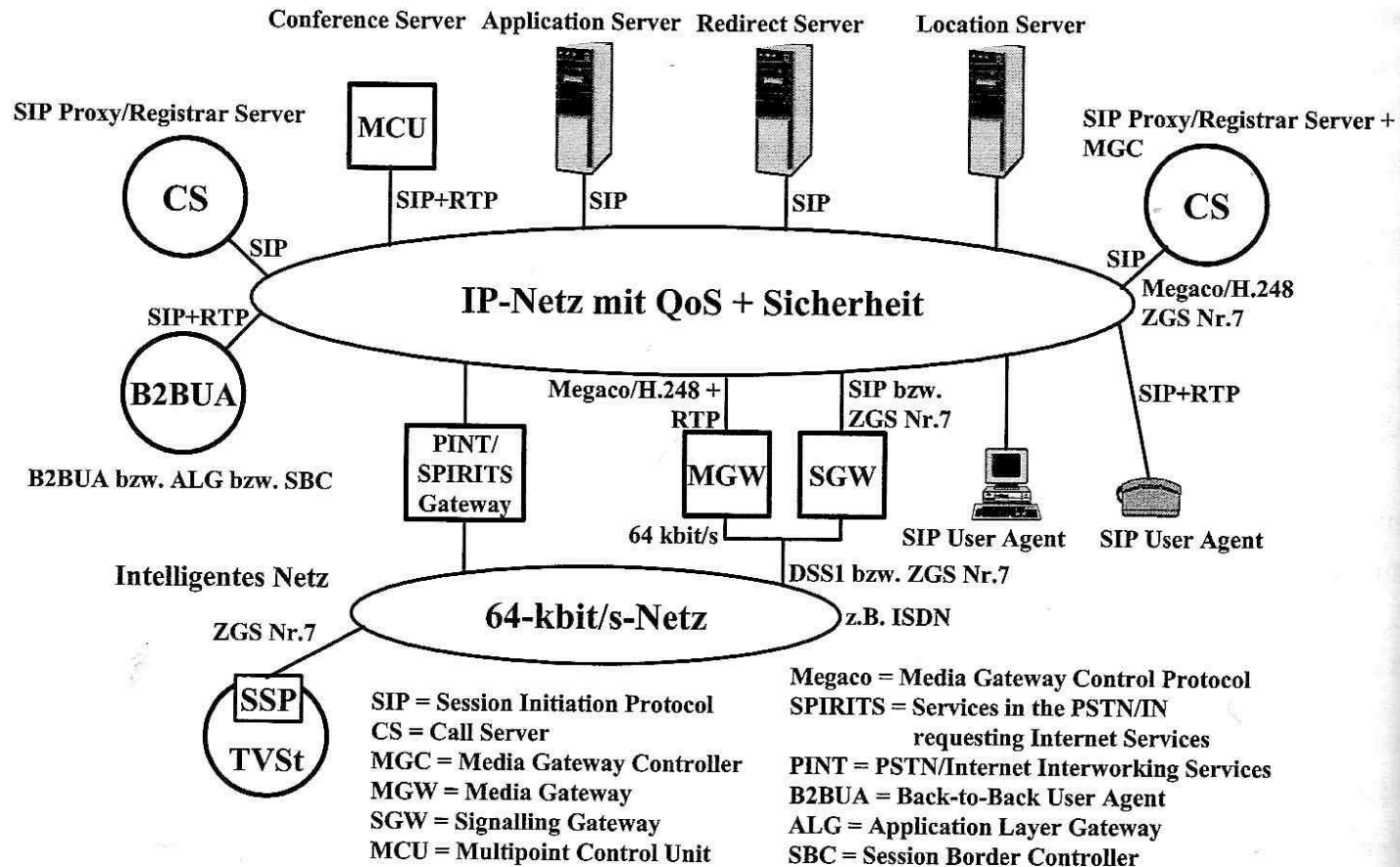


Bild 3.3: Protokolle und Netzarchitektur für Next Generation Networks mit SIP für die Session-Steuerung

Trick/Weber

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Ulrich Trick, Frank Weber: SIP, TCP/IP und
Telekommunikationsnetze, Oldenbourg, 4. Auflage 2009

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”

Addressing in SIP

- SIP supports various address formats including addresses based on phone numbers
 - ITU standard for international phone number format: E.164
- Email style addresses:
`sip:Heinrich.Hussmann@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
- IP-based addressing of terminals is a potential problem
 - Many large sites use NAT (network address translation)

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

SIP Messages

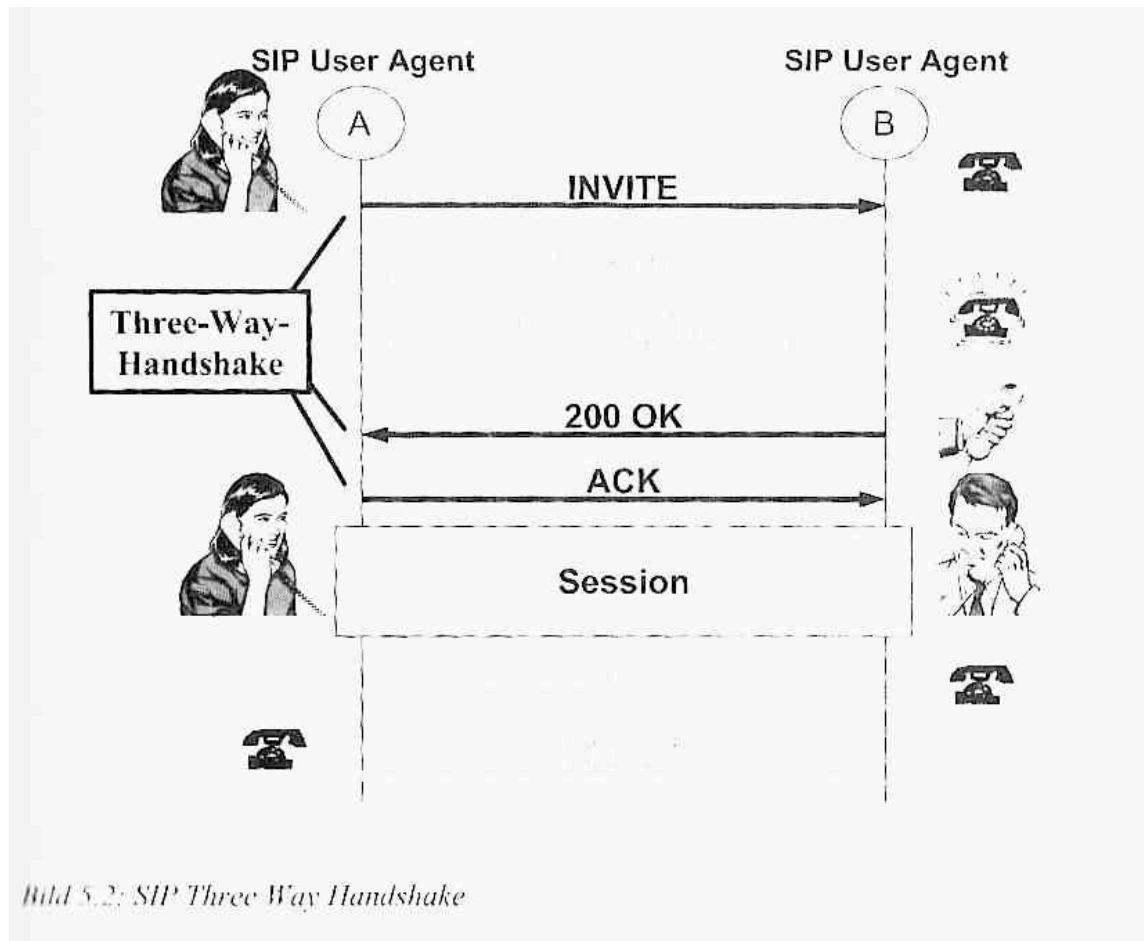
- Text-based peer-to-peer protocol
- Modelled after HTTP
 - *Header* contains connection parameters and service information
 - *Body* contains description of connection (using *Session Description Protocol* SDP)
- Requests:
 - From client (agent) to server (agent)
 - INVITE, BYE, OPTIONS, STATUS, CANCEL, 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	<i>Start Line</i>
VIA:SIP/2.0/UDP 169.130.12.5	<i>General Header</i>
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	<i>Sequence Number</i>
Subject:Mr. Watson, come here	<i>Request Header</i>
Content-Type:application/sdp	<i>Entity Header</i>
Content-Length:885	
v=0	<i>Body: SDP Data</i>
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	

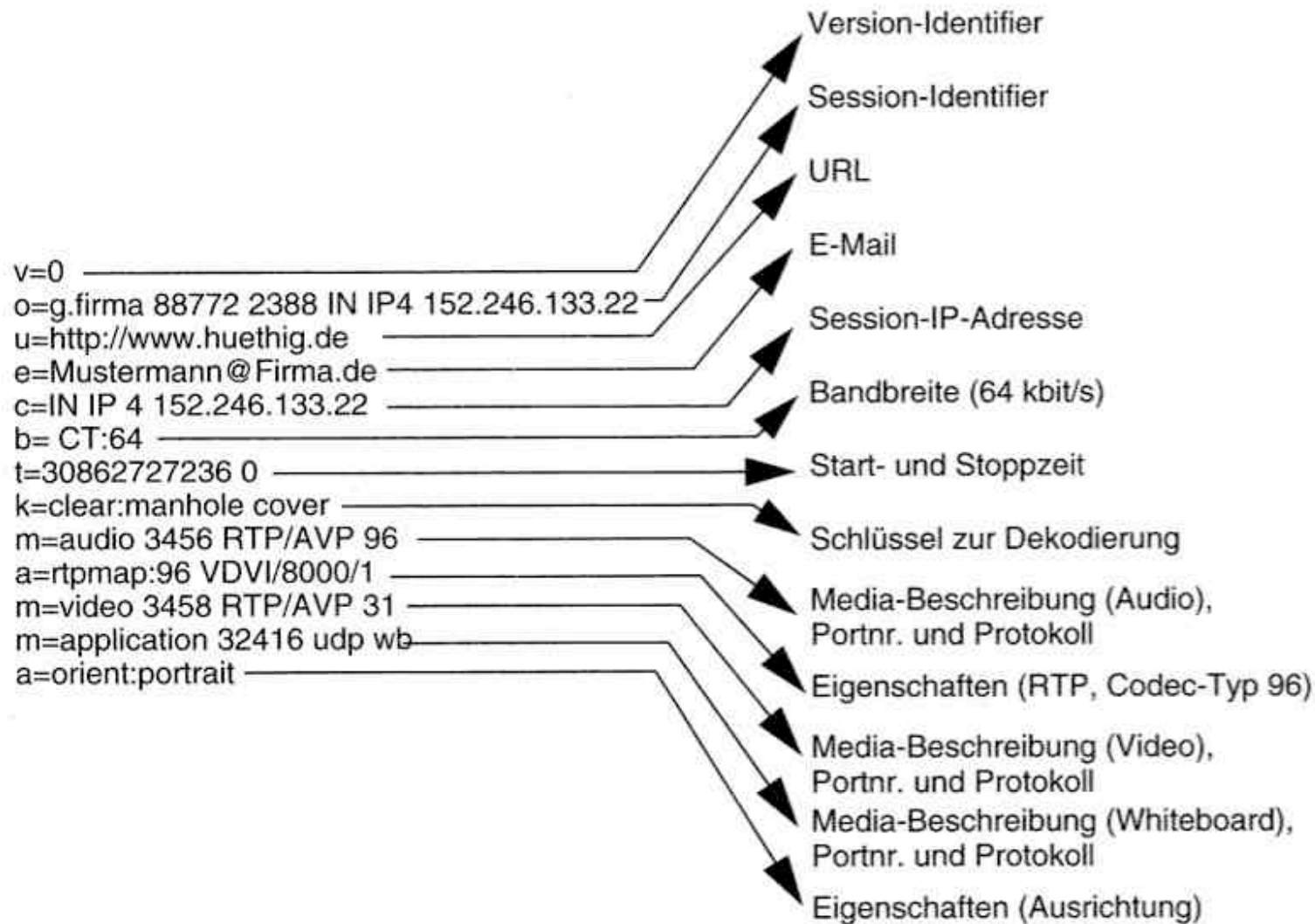
Call Setup by Three-Way Handshake

- Direct connection establishment between two SIP terminals (user agents)



Trick/Weber

SDP Information



SDP Media Description and Attributes

- Media description (*m*)
 - Media type (e.g. *audio*)
 - Used port number
 - User data transport protocol
 - » e.g. RTP/AVP = Real-Time Transport Protocol, Audio/Video Profile
 - List of available formats/codecs
 - » "96" in previous example, may be a list of options
- Attribute description (*a*)
 - Codec details for all mentioned media formats
 - E.g. from "rtpmap" in RTP/AVP standard (IETF RFC 3551)

Example for Multiple Media Formats

```
m=audio 2410 RTP/AVP 0 8 3 4
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:3 GSM/8000
a=rtpmap:4 G723/8000
```

- Communication partner announces the codecs/formats which are locally supported
- Standardized list of RTP-Codecs in RTP/AVP standard, excerpt:

Payload type	Encoding name	Media type	Clock rate	Channels
0	PCMU (μ -law)	A	8000	1
1	reserved	A		
2	reserved	A		
3	GSM	A	8000	1
4	G723	A	8000	1
5	DVI4	A	8000	1
6	DVI4	A	16000	1
7	LPC	A	8000	1
8	PCMA (a-law)	A	8000	1

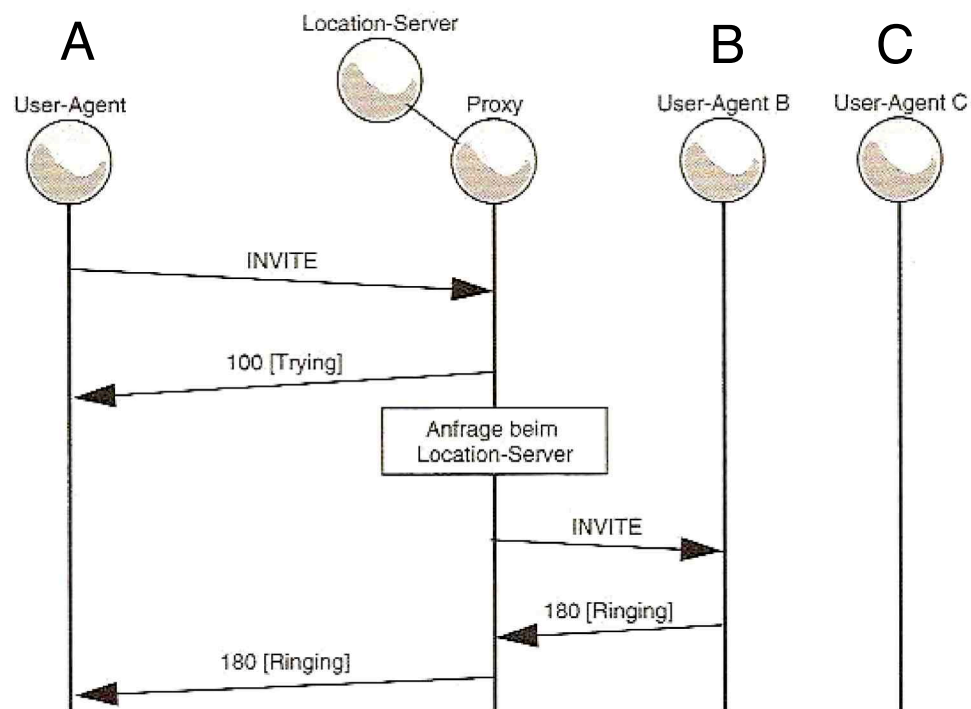
Codec Negotiation by Offer&Answer

- In connection establishment dialogue (3-way handshake):
 - Partner A sends *offer* (list of supported codecs) as SDP part of *INVITE*
 - Partner B selects appropriate options and specifies them as SDP part of *OK*
- Example:
 - Offer:
`m=audio 2410 RTP/AVP 0 8 3 4`
 - Answer:
`m=audio 2468 RTP/AVP 0 3`
- Analogous negotiation for multiple media channels
 - E.g. audio + video
 - E.g. chat, possibly encrypted
 - E.g. file transfer

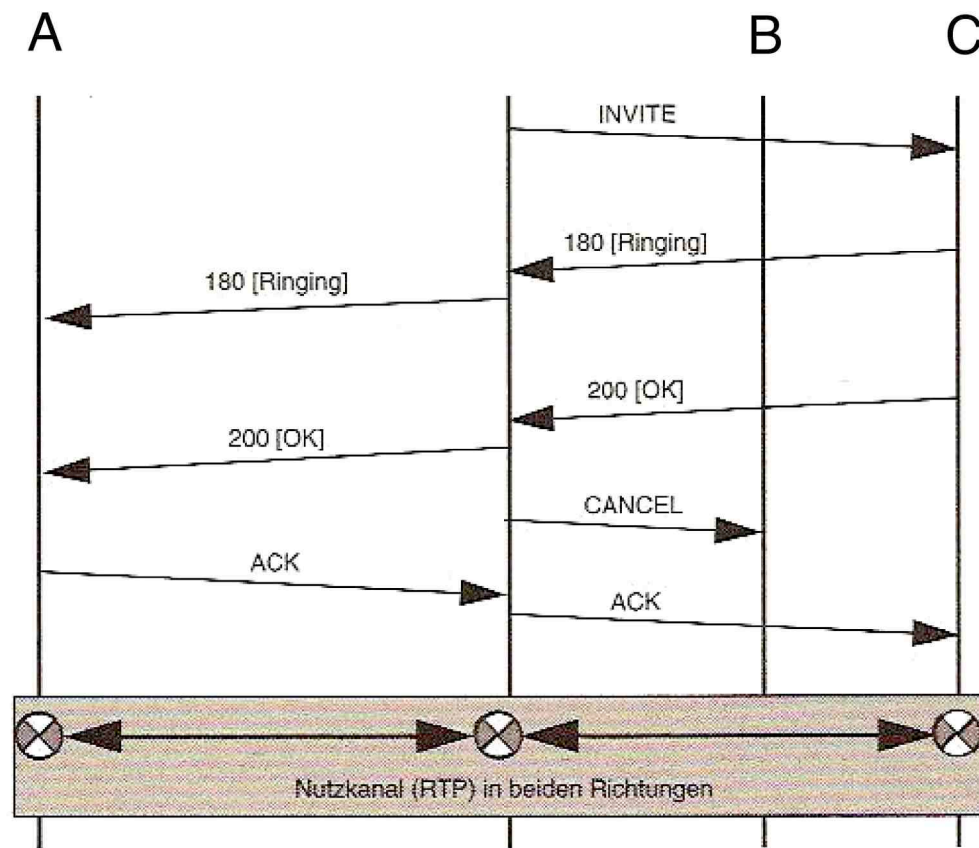
SIP Proxy Servers

- *Proxy Servers* realize message routing
 - Proxy server forwards SIP messages
 - Takes local decisions on routing
 - In some cases initiates more complex signaling sequences
- Stateless proxy server:
 - Just forwards messages, only routing decisions taken
- Stateful proxy server:
 - Active network element
 - Stores status of incoming requests
 - May create new requests on its own

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



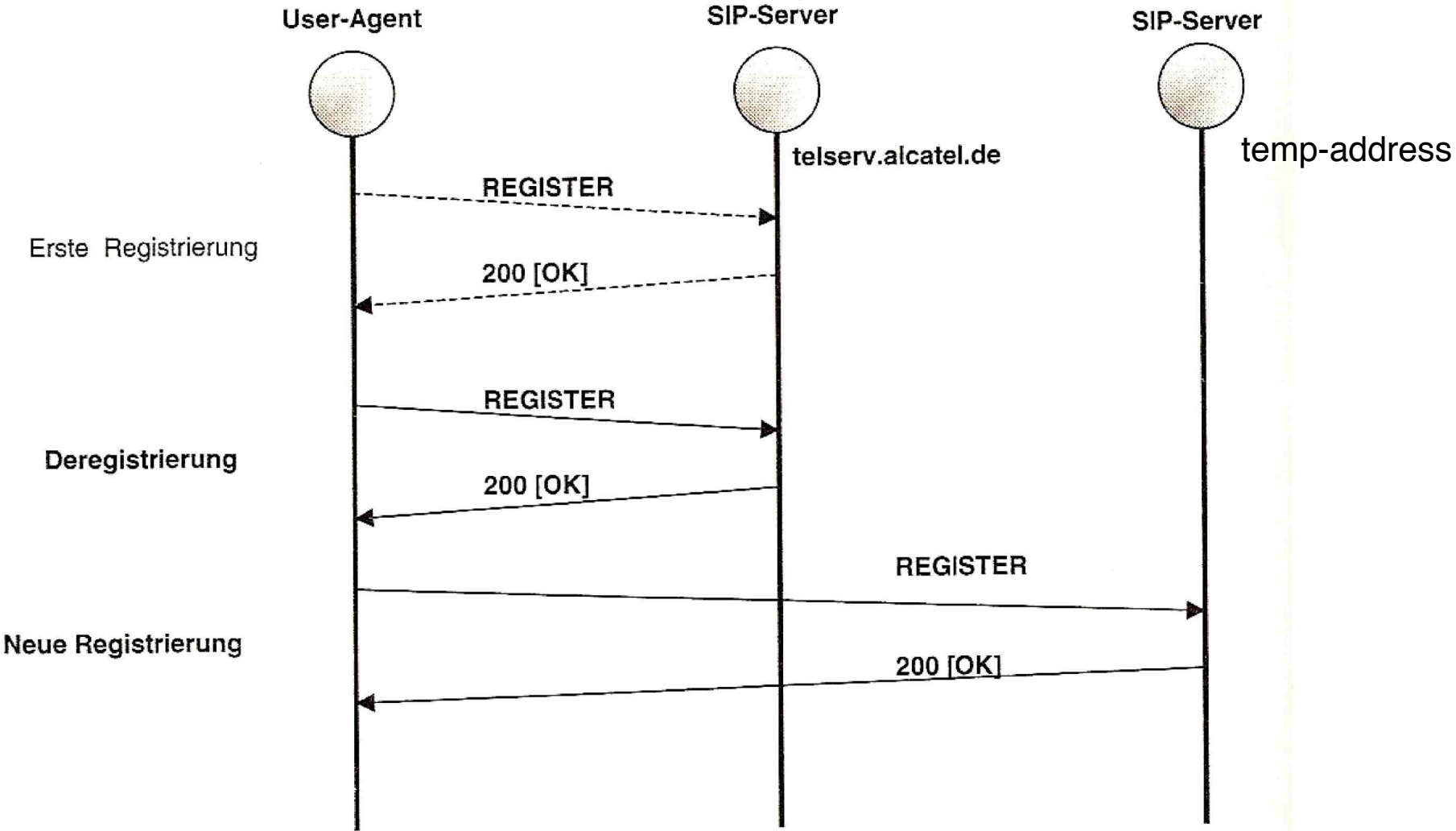
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Example: Personal Number

- 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 (IMT-2000)
 - (1st: Analog, 2nd: GSM)
 - UMTS provides unique standard for Europe, USA 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

Alternatives to SIP

- H.32X and Skype, see above
- Current open-source development:
 - Open source VoIP switching software
 - » “Asterisk”, see www.asterisk.org
 - » Developed by company (Digium) selling gateway hardware
 - Asterisk Inter-Exchange Protocol (IAX)
 - » Not (yet) a standard
 - » Possible alternative to SIP
 - Advantages of IAX over SIP
 - » Better efficiency (not character-coded)
 - » Better interworking with NAT
 - » Easier to administer regarding prt numbers (one port number only)

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Literature:

www.ietf.org/impp

www.xmpp.org

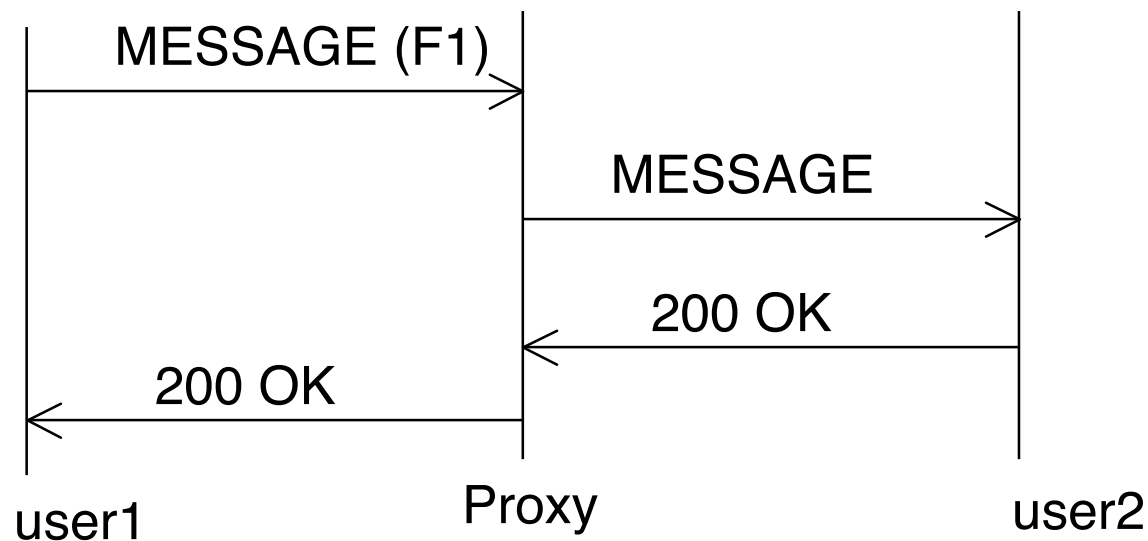
Instant Messaging (IM)

- Exchange of text information between clients in real-time
- Usually combined with *presence information*
- Traditionally computer-based, but may be used on other devices
- Modern clients often integrated with audio/video conferencing
- History:
 - 1970s: Terminal-based messaging (e.g. Unix “talk”)
 - Commercial GUI-based systems: ICQ (1996), AOL Instant Messenger (1997)
 - Many incompatible systems: Yahoo, MSN, Excite, ...
 - 2000: Open-source protocol “Jabber”, developed into XMPP
 - Current: Multi-protocol clients, e.g. Trillian, Proteus, iChat
- Architecture:
 - Many clients, few servers
 - Device-based or network-based (server-based)
 - Centralized servers (e.g. ICQ) vs. decentralized servers (e.g. Jabber)

Signaling for Instant Messaging

- Majority of used protocols is proprietary to specific service!
- Several efforts for standardization, two important examples:
- SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions)
 - RFCs 3428, 3856 plus drafts
 - Messaging as extensions of the SIP protocol
 - Many features still under construction, e.g. security, multimedia
- XMPP (Extensible Messaging and Presence Protocol)
 - Standardized form of XML-based streaming and presence protocols developed by the “Jabber” community (since 1999)
 - IETF standardization 2002–2004: RFCs 3920-23
 - Quite complete, covers e.g. authentication and encryption, multi-user chat, privacy blocking
 - Increasing support from commercial IM applications
 - » e.g. Google Talk, Apple iChat

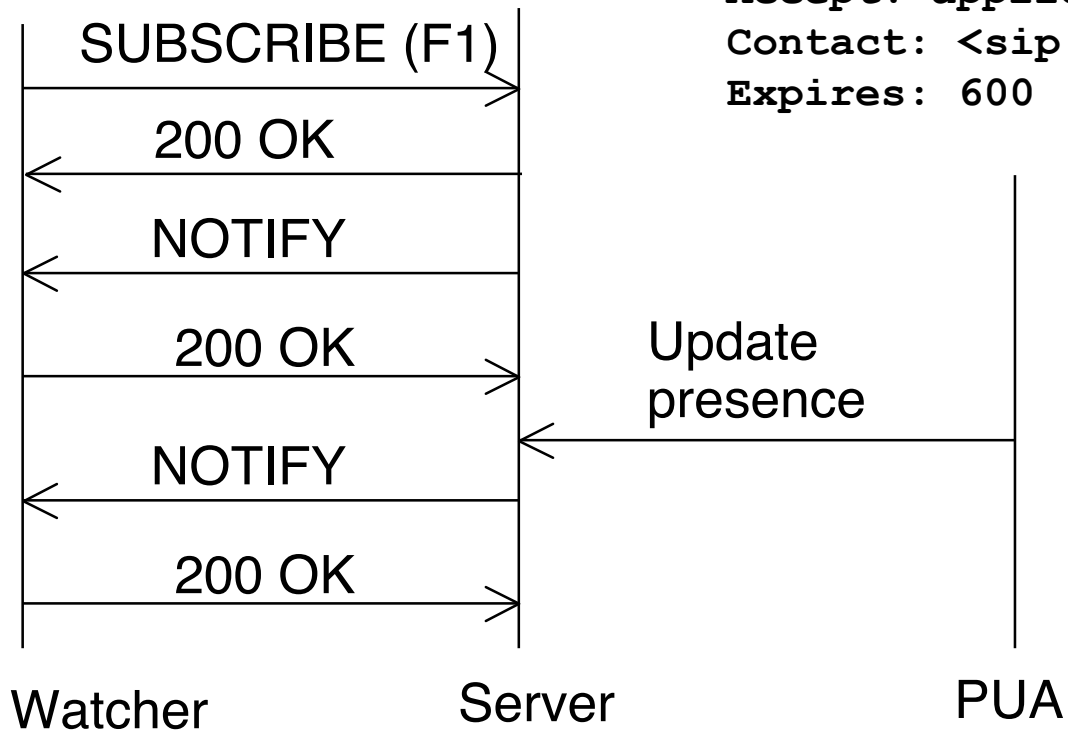
SIMPLE Example (1): Message



F1: MESSAGE sip:user2@domain.com SIP/2.0
Via: SIP/2.0/TCP user1pc.domain.com;branch=z9hG4bK776sgdkse
Max-Forwards: 70
From: sip:user1@domain.com;tag=49583
To: sip:user2@domain.com
Call-ID: asd88asd77a@1.2.3.4
CSeq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 18
Watson, come here.

SIMPLE Example (2): Presence

```
SUBSCRIBE sip:resource@example.com SIP/2.0
Via: SIP/2.0/TCP watcherhost.example.com;...
To: <sip:resource@example.com>
From: <sip:user@example.com>;tag=xf9
...
Event: presence
Accept: application/pidf+xml
Contact: <sip:user@watcherhost.example.com>
Expires: 600 ...
```



XMPP

- Based on generic transport protocol for XML streams over the Internet
- Idea:
 - Two-way exchange of XML files of potentially infinite length
 - Transmission of discrete semantic units (*XML stanzas*)

```
<stream>  
  <presence>  
    <show/>  
  </presence>  
  <message to='foo'>  
    <body/>  
  </message>  
  <iq to='bar'>  
    <query/>  
  </iq>  
  ...  
</stream>
```



XMPP Example

C \longleftrightarrow S

```
C: <?xml version='1.0'?>
  <stream:stream
    to='example.com'
    xmlns='jabber:client'
    xmlns:stream='http://etherx.jabber.org/streams'
    version='1.0'>
S: <?xml version='1.0'?>
  <stream:stream
    from='example.com'
    id='someid'
    xmlns='jabber:client'
    xmlns:stream='http://etherx.jabber.org/streams'
    version='1.0'>
... encryption, authentication, and resource binding ...
C: <message from='juliet@example.com'
  to='romeo@example.net'
  xml:lang='en'>
C:   <body>Art thou not Romeo, and a Montague?</body>
C: </message>
S: <message from='romeo@example.net'
  to='juliet@example.com'
  xml:lang='en'>
S:   <body>Neither, fair saint, if either thee dislike.</body>
S: </message>
C: </stream:stream>
S: </stream:stream>
```