13 Signaling Protocols for Multimedia Communication

Literature:

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

Outline

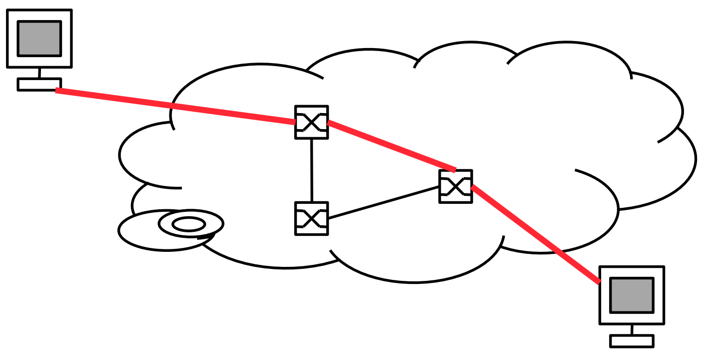
1. Introduction and Motivation

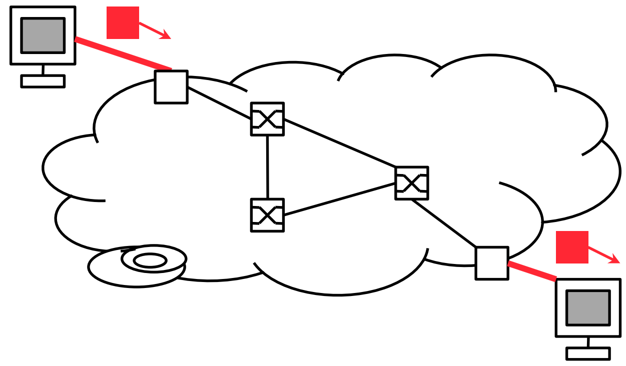
2. Media on the Web 3. Interactive Web Applications 4. Communities, the Web, and Multimedia	Part I: Web Technologies for Interactive MM
5. Digital Rights Management 6. Cryptographic Techniques 7. Multimedia Content Description	Part II: Content-Oriented Base Technologies
8. Streaming Architectures 9. Web Radio, Web TV and IPTV 10. Electronic Books and Magazines 11. Multimedia Content Production and Management	Part III: Multimedia Distribution Services
12. Multimedia Conferencing 13. Signaling Protocols for Multimedia Communication	Part IV: Conversational Multimedia Services

14. Visions and Outlook

Communication networks

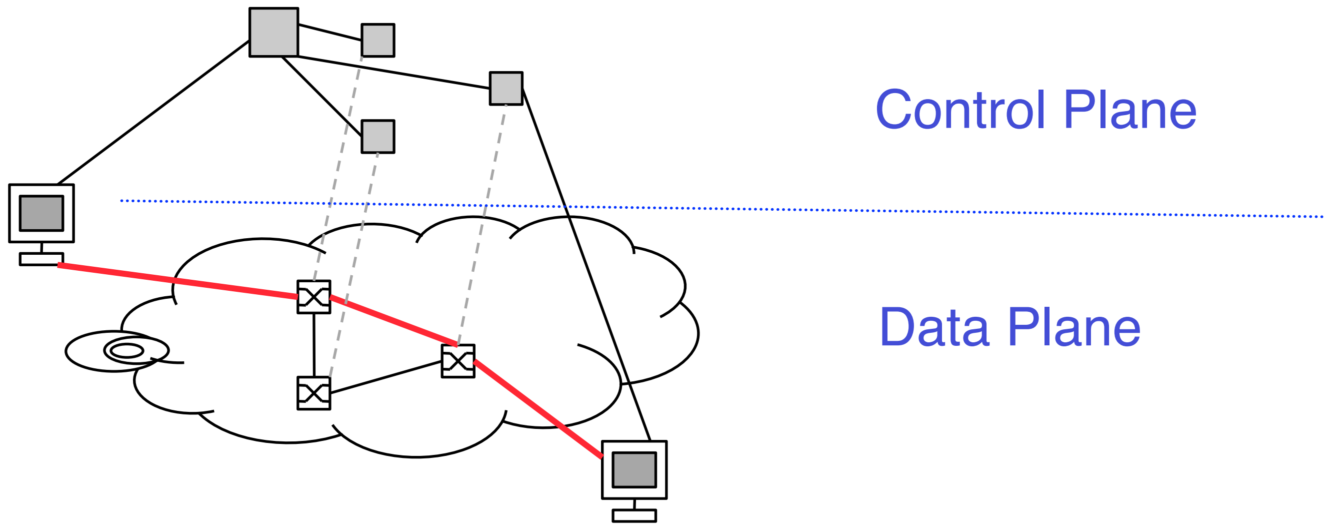
- Classification of communication networks:
  - Circuit-switched *(Leitungsvermittlung)*: Physical connection between communicating end systems (for limited duration)
    - » Traditional telephone networks
    - » *Virtual connections* in advanced digital networks (e.g. ATM)
  - Packet-switched *(Paketvermittlung)*: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





Control Plane and Data Plane

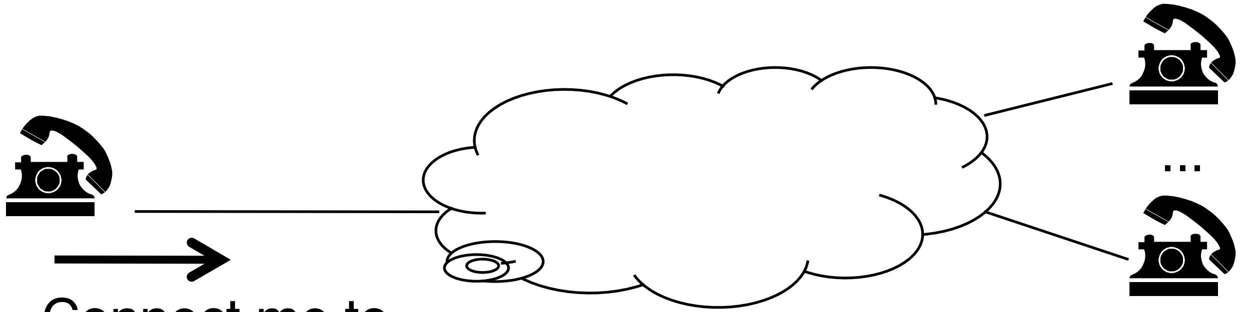
- Classification of network mechanisms:
  - *Control Plane:* Mechanisms of the network to establish, modify and remove connections
  - *Data Plane:* Mechanisms of the network to transmit data over established connections
  - Strict separation of Control and Data planes in traditional telephone networks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



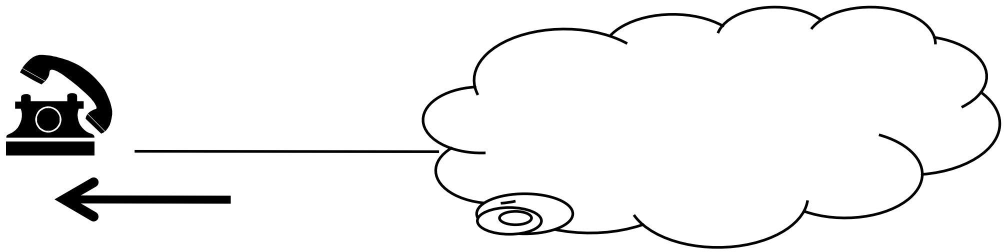
Signaling

- *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



Connect me to
number +49 2180 4650



Call has been
terminated

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
    - » Gateways between Internet and telephone networks:
      - calling a phone from a PC, using an iPod over WLAN like a phone, …
  - 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
  - Adadtion to network traffic situation
  - Advanced features (like multiple calls, intelligent forwarding)

Network Architecture Option 1: Skype Based

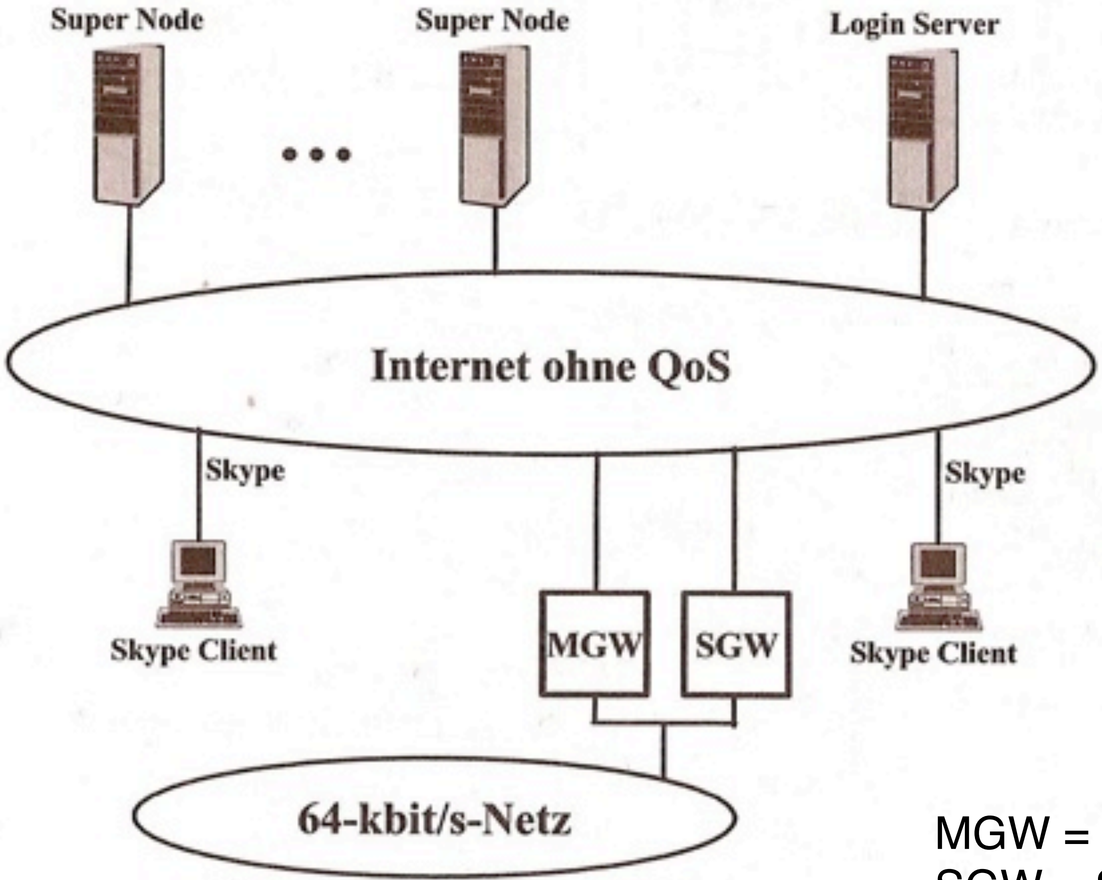


Bild 3.6: Skype für die Session-Steuerung

MGW = Media Gateway
SGW = Signaling Gateway

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 (suppplementary services), …,
      but also RTP, RTCP, …
  - Originally developed for videoconferencing, see previous 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 signaling for IP telephony (and multimedia conferencing…)

Network Architecture Option 2: H.32X Based

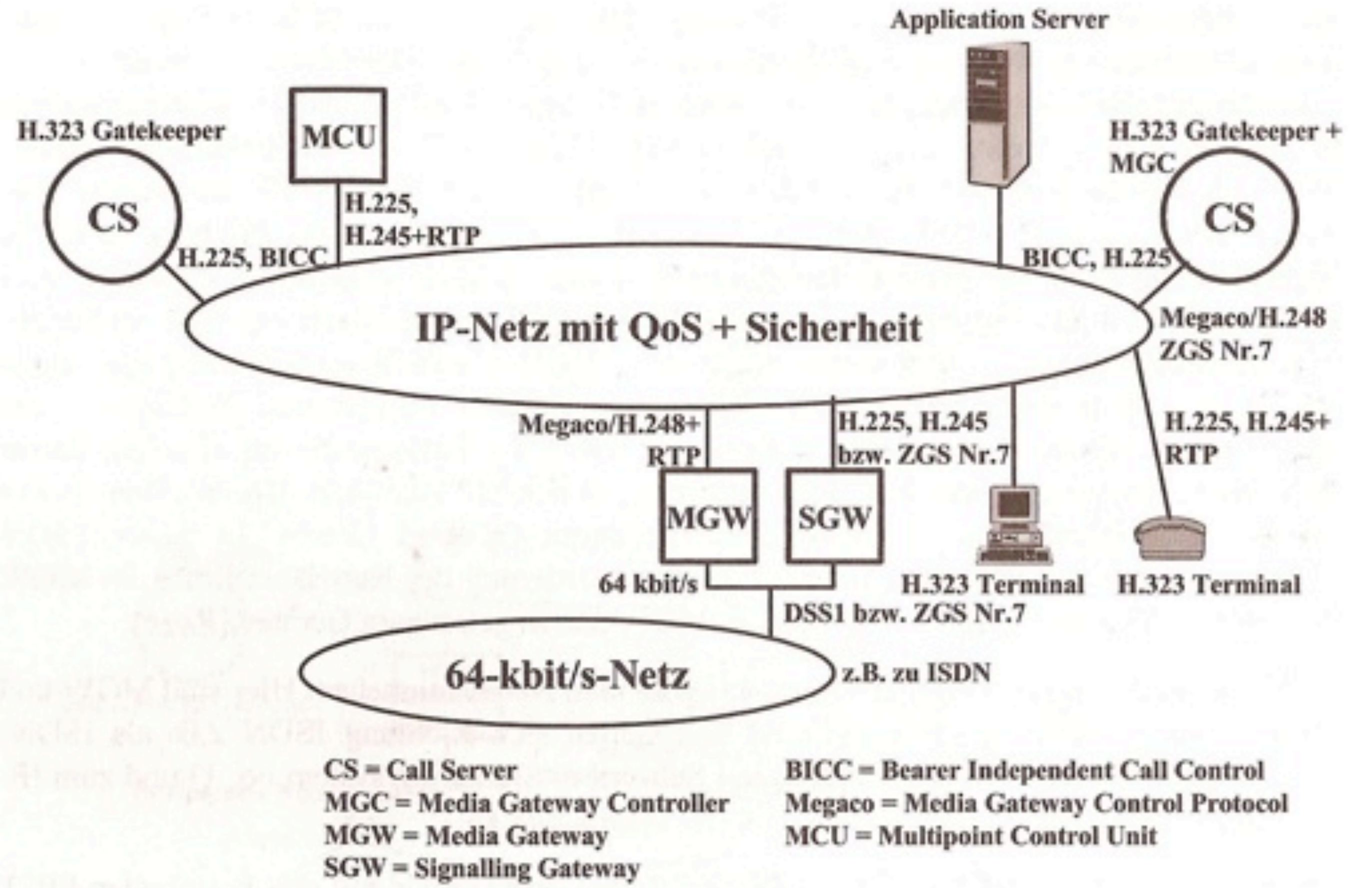


Bild 3.4: Protokolle und Netzarchitektur für Next Generation Networks mit H.323 für die Session-Steuerung

Trick/Weber

Network Architecture Option 3: SIP Based

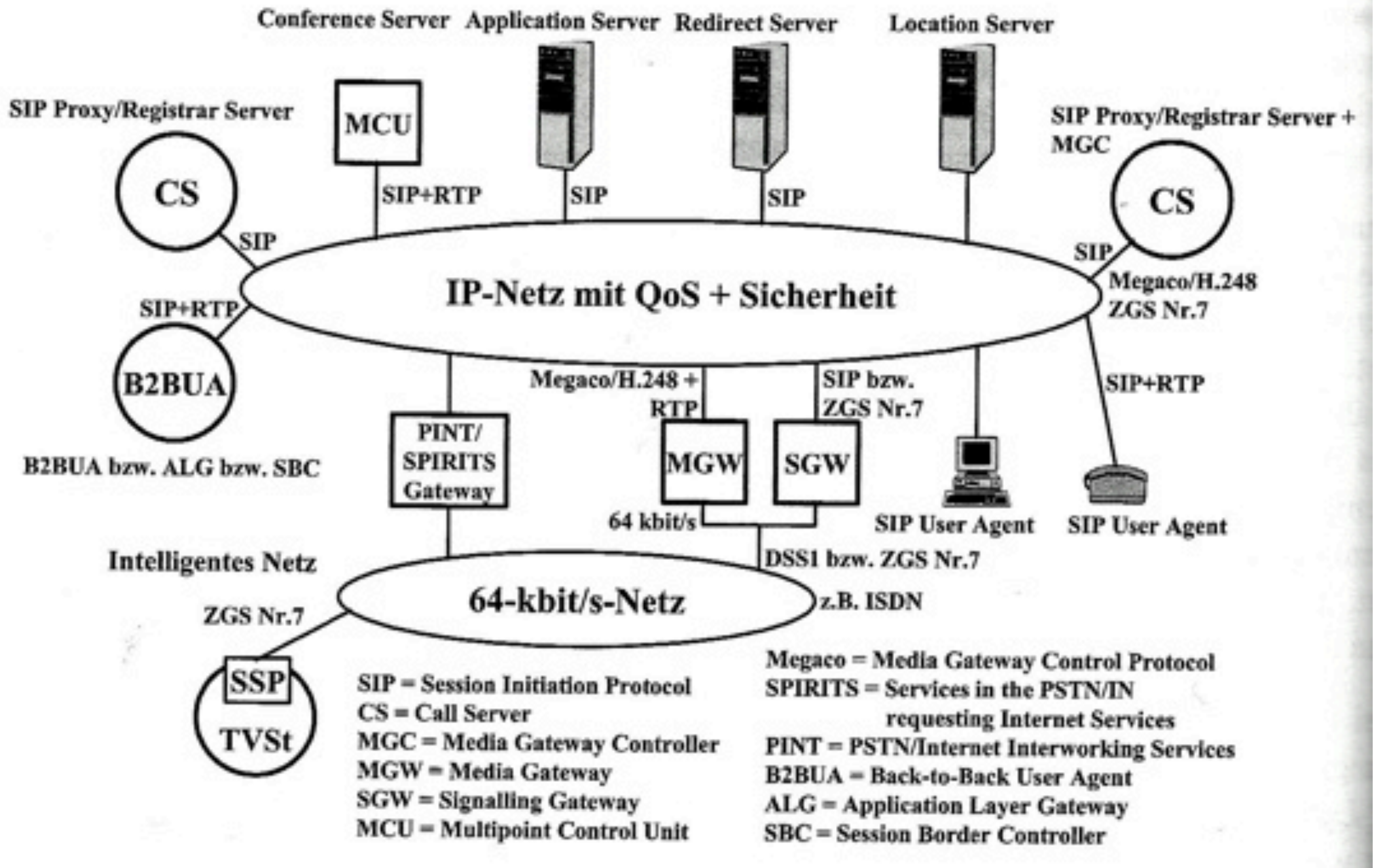


Bild 3.3: Protokolle und Netzarchitektur für Next Generation Networks mit SIP für die Session-Steuerung

Trick/Weber

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

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*

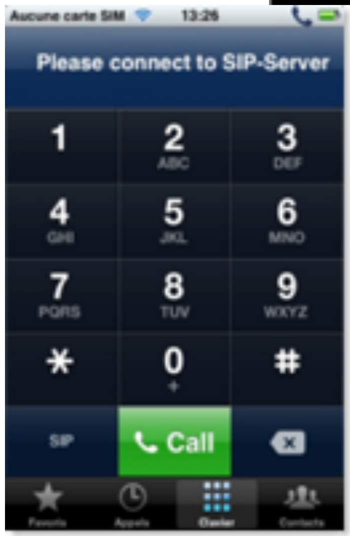
Main Features & Components of SIP

- 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”

SIP Terminals

- PCs, laptops, tablets, mobile phones, music players, ...
  – with SIP-enabled applications
  – with Internet access (e.g. WLAN)



- SIP version 2 (RFC 3261, 3262, 3263, 3264)
- SPCP with the Cisco Unified Communications 500 Series
- SIP proxy redundancy: dynamic via DNS SRV, A records
- Reregistration with primary SIP proxy server
- SIP support in NAT networks (including STUN)
- SIPFrag (RFC 3420)
- Secure (encrypted) calling via SRTP
- Codec name assignment
- Voice algorithms:
  - G.711 (A-law and μ-law)
  - G.726 (16/24/32/40 kbps)
  - G.729 A
  - G.722

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 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, …

Call Setup by Three-Way Handshake

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

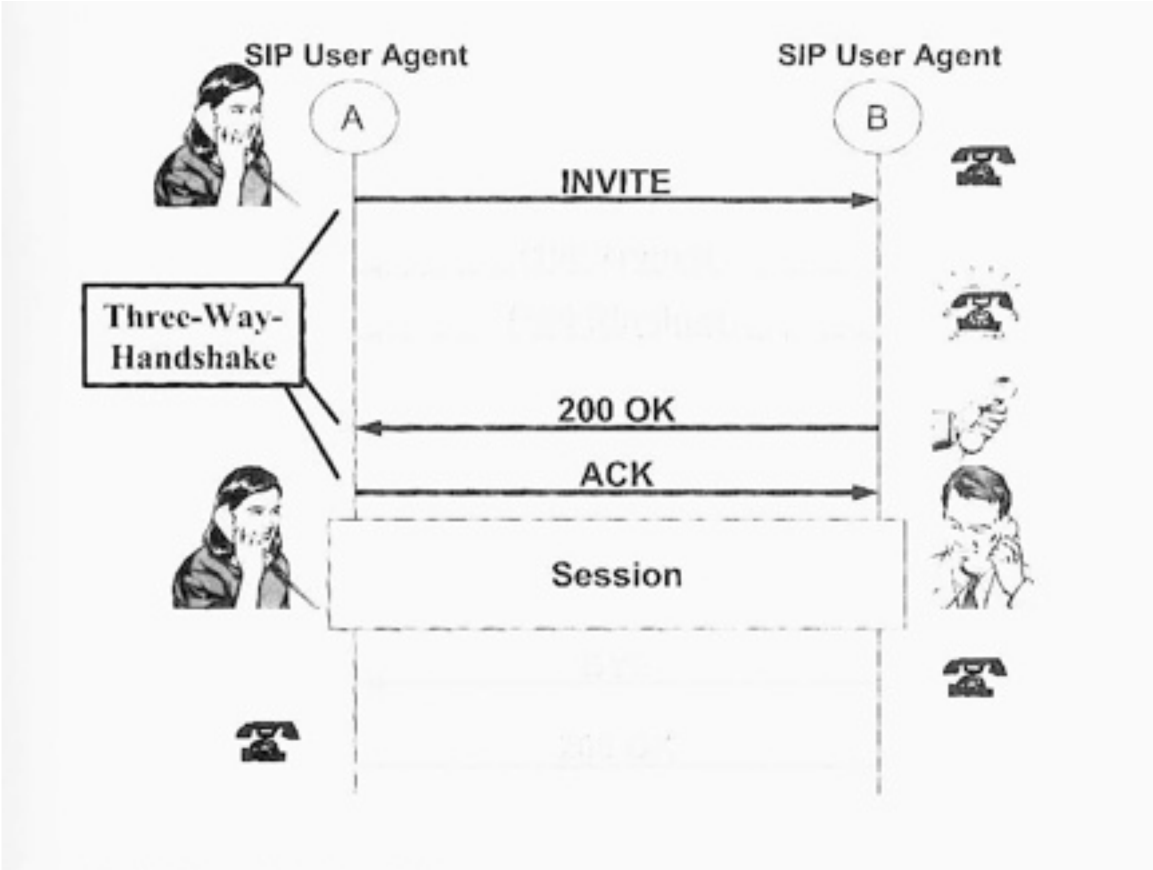


Bild 5.2: SIP Three Way Handshake

Trick/Weber

Example: SIP Message

```
INVITE sip:john@domain.com SIP/2.0
```
Start Line

```
VIA:SIP/2.0/UDP 169.130.12.5
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
```
General Header

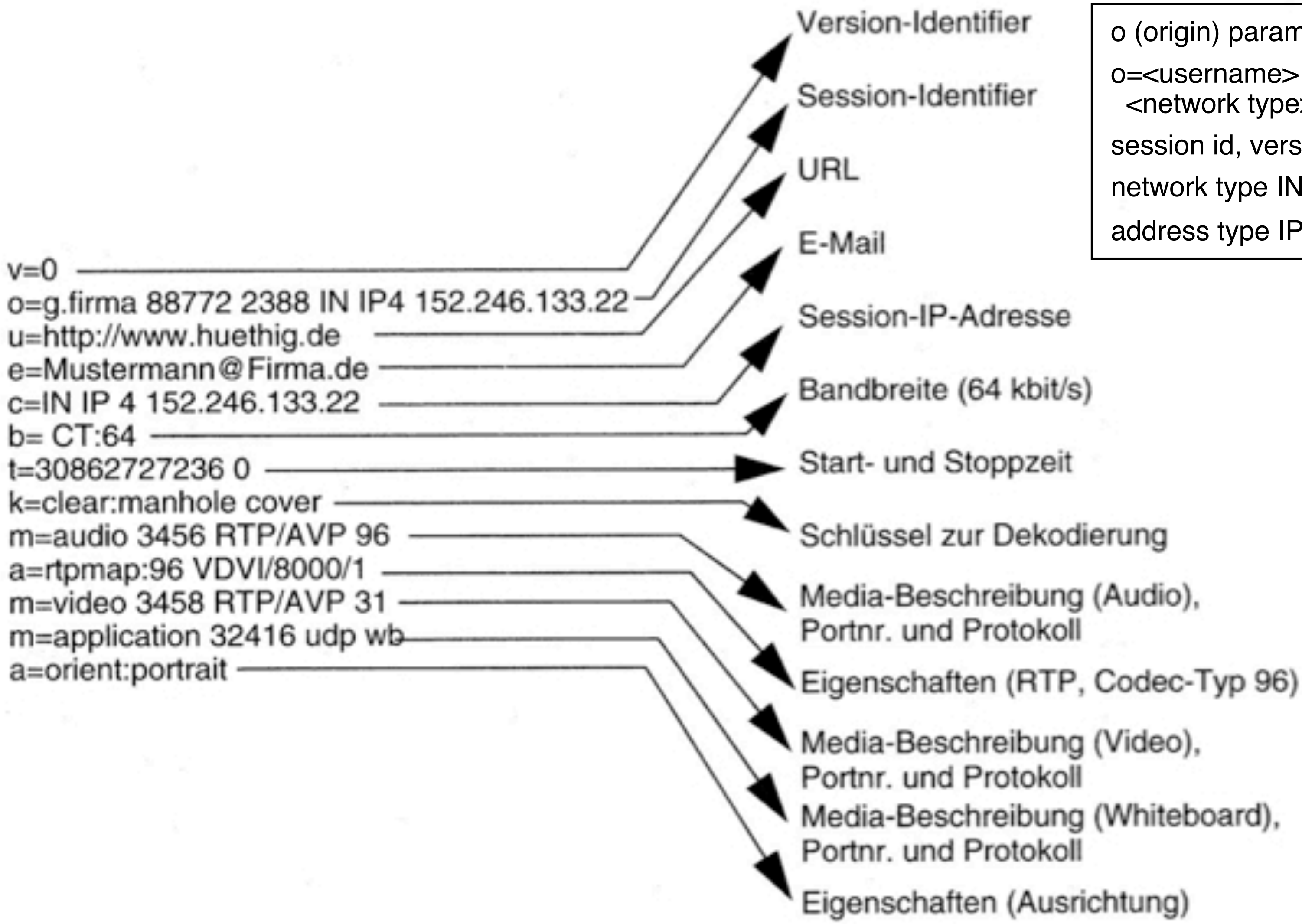
Sequence Number

```
Subject:Mr. Watson, come here
```
Request Header

```
Content-Type:application/sdp
Content-Length:885
```
Entity Header

```
v=0
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
```
Body: SDP Data

SDP Information



o (origin) parameter:

o=<username> <session id> <version> <network type> <address type> <address>

session id, version: NTP timestamp

network type IN = Internet

address type IP4 or IP6

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:

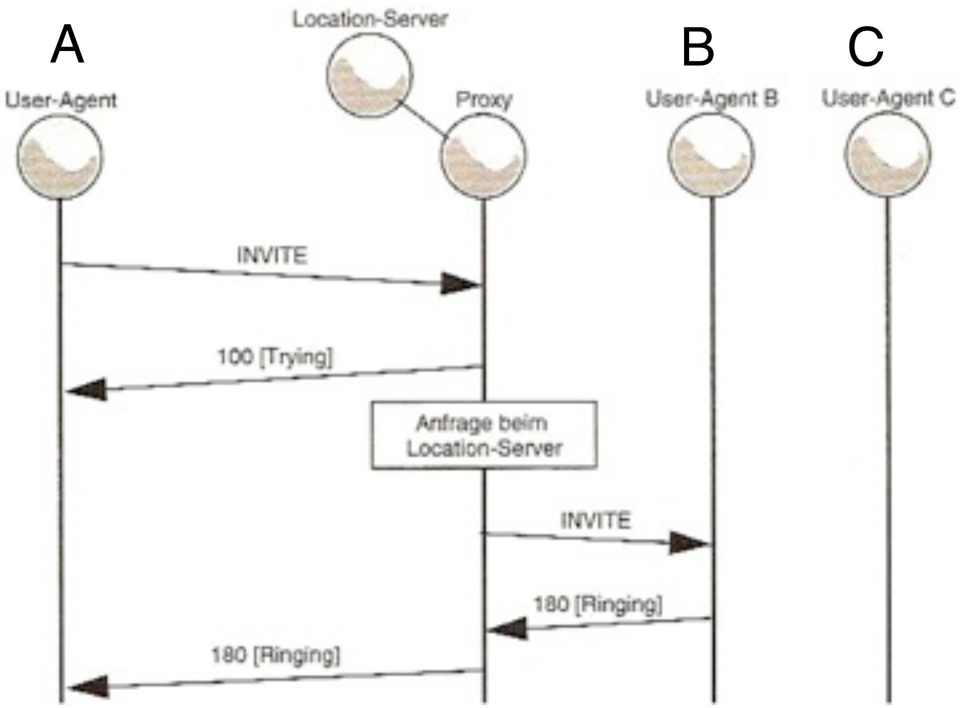
    `m=audio 2410 RTP/AVP 0 8 3 4`
  - Answer:

    `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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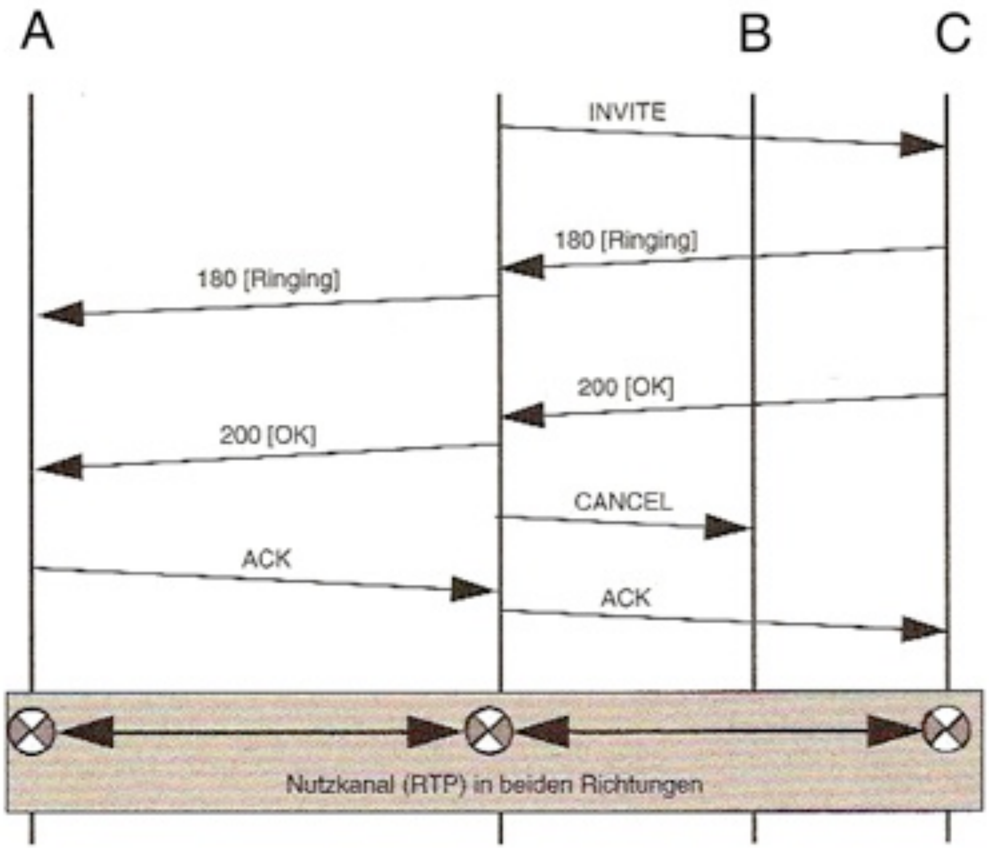
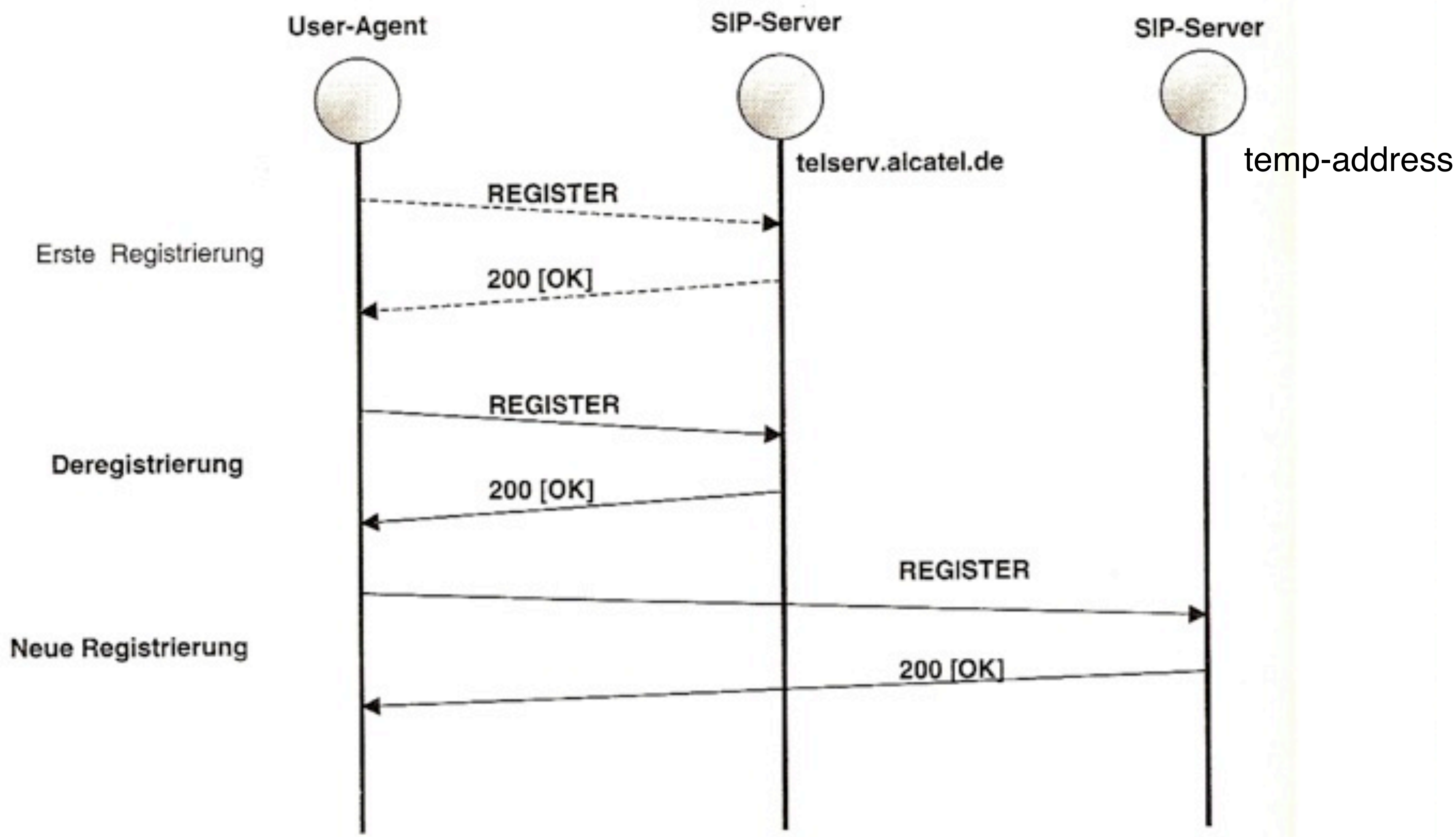


Example: Personal Number

- 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 Specification Releases (currently 8 releases)
  - Since release 4 and 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 a standard, rather a community-based effort
    - Possible alternative to SIP
  - Advantages of IAX over SIP
    - Better efficiency (not character-coded)
    - Better interworking with NAT
    - Easier to administer regarding port numbers (one port number only)

13 Signaling Protocols for Multimedia Communication

13.1 Signaling and Sessions

13.2 SIP Basics

13.3 Signaling for Instant Messaging

Literature:

www.ietf.org/impp
www.xmpp.org

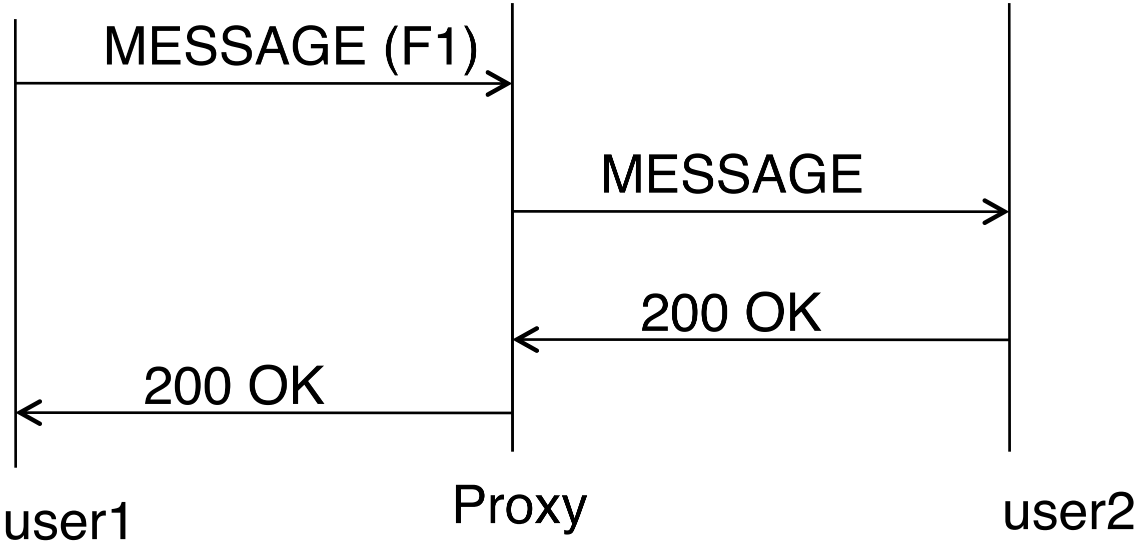
Instant Messaging (IM)

- 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, 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, 3863, 4479, ... and many drafts
  - Messaging as extensions of the SIP protocol
  - Currently no multimedia support, just text messages
- 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, Facebook Chat XMPP Interface (2010)

SIMPLE Example (1): Message

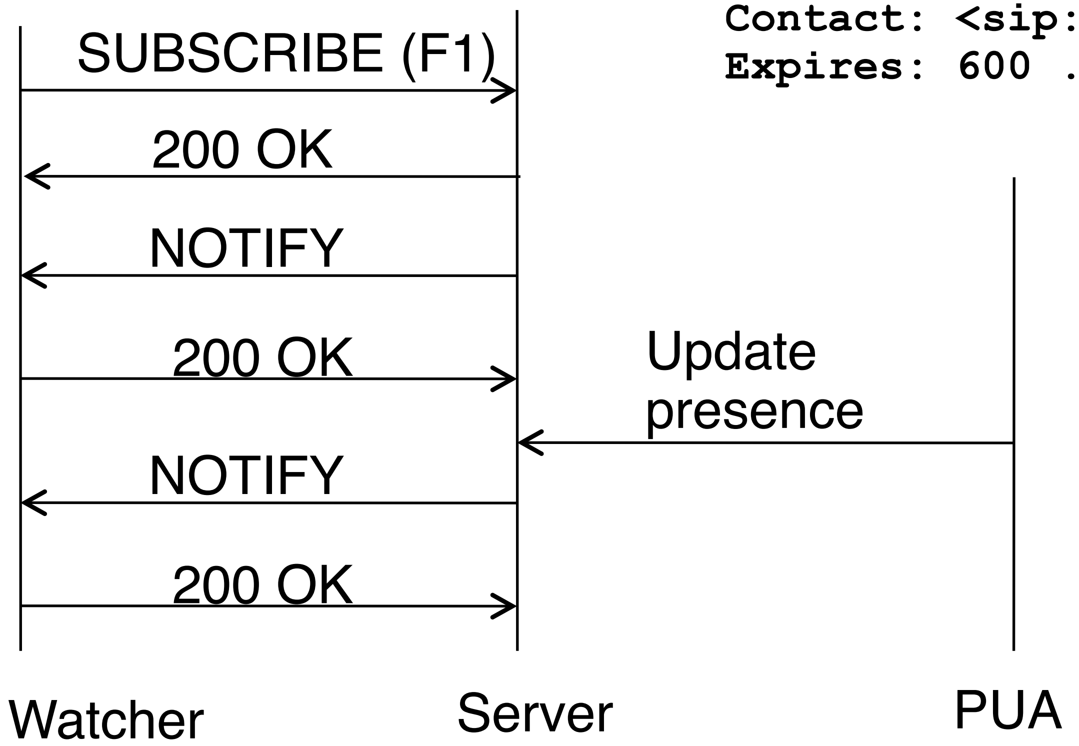


F1:

```
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=xfg9
...
Event: presence
Accept: application/pidf+xml
Contact: <sip:user@watcherhost.example.com>
Expires: 600 ...
```



PIDF (RFC 3863):
Presence Indication
Data Format

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>
```

iq = info/query

XMPP Example

C ⟷ 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>
```

Source: RFC 3920

XMPP Extensions

- Structured process for continuous extension and clarification of the protocol suite, "XMPP Extension Protocol XEP"
  – Maintained by XEP Editor
- Categories of extensions:
  – Standards Track: Additional definitions
  – Informational: Further descriptions and examples
  – Historical: Proposed but never made official
  – Humorous (often April pranks)
    » E.g. *XEP-0132 Presence Obtained via Kinesthetic Excitation (POKE)*
  – Procedural: Internal organization of XMPP Standards Foundation