Chair for Future Communication
Prof. Dr. K. Tutschku
Institute for Distributed and Multimedia Systems
Faculty for ComputerScience

050069 VO Netzwerktechnologie für Multimedia Anwendungen

Lecture 5: Multimedia-Networking

Prof. K. Tutschku (kurt.tutschku@univie.ac.at)

BachelorInformatik (Medieninformatik)
WS 2010/11

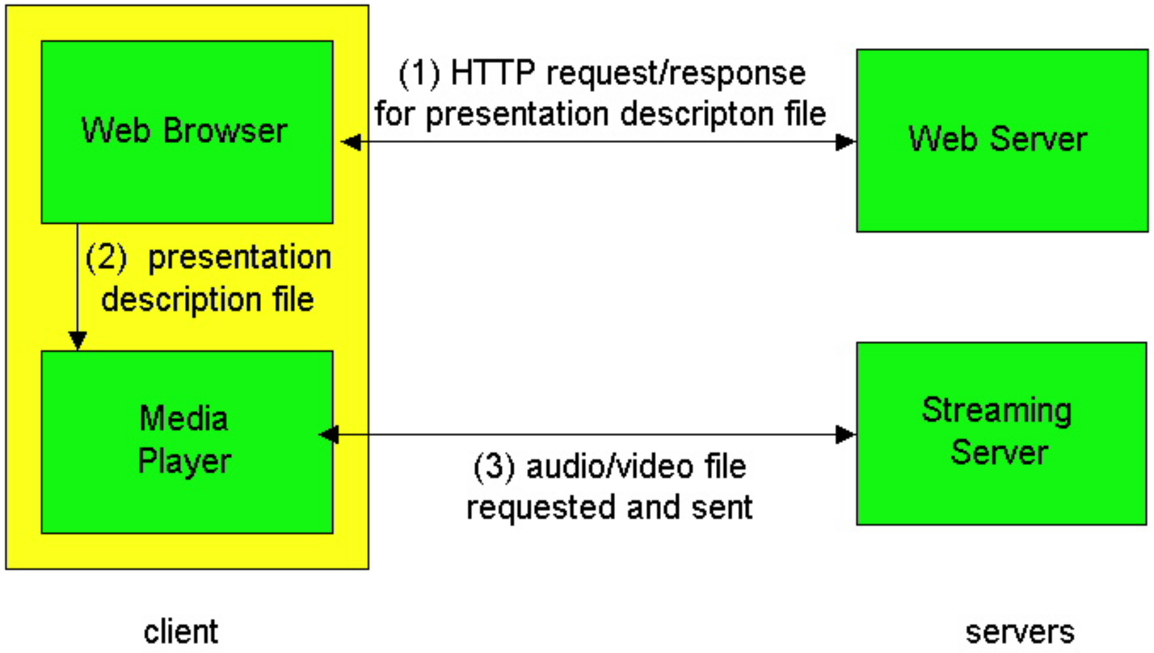
Endowed by

universität wien

Streaming from a streaming server

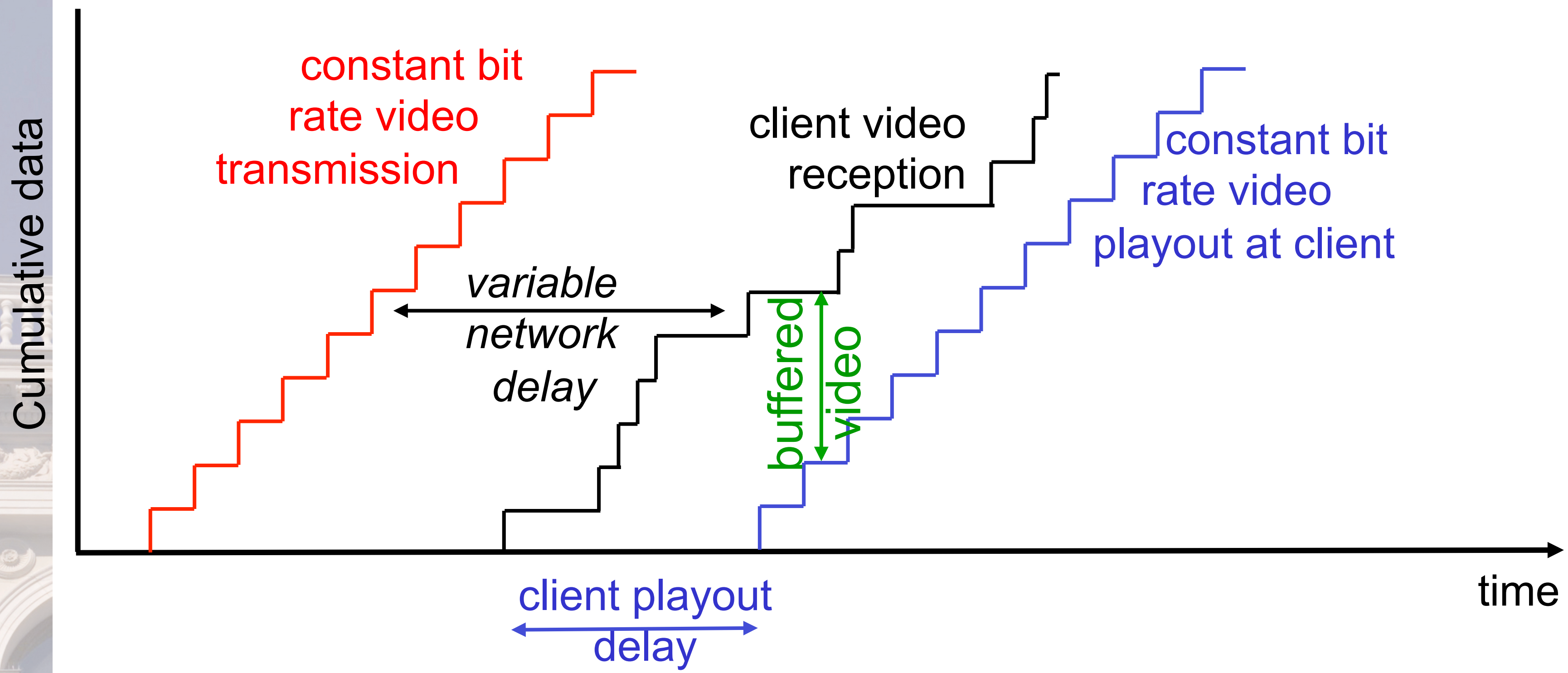
Chair of **Future Communication**

endowed by



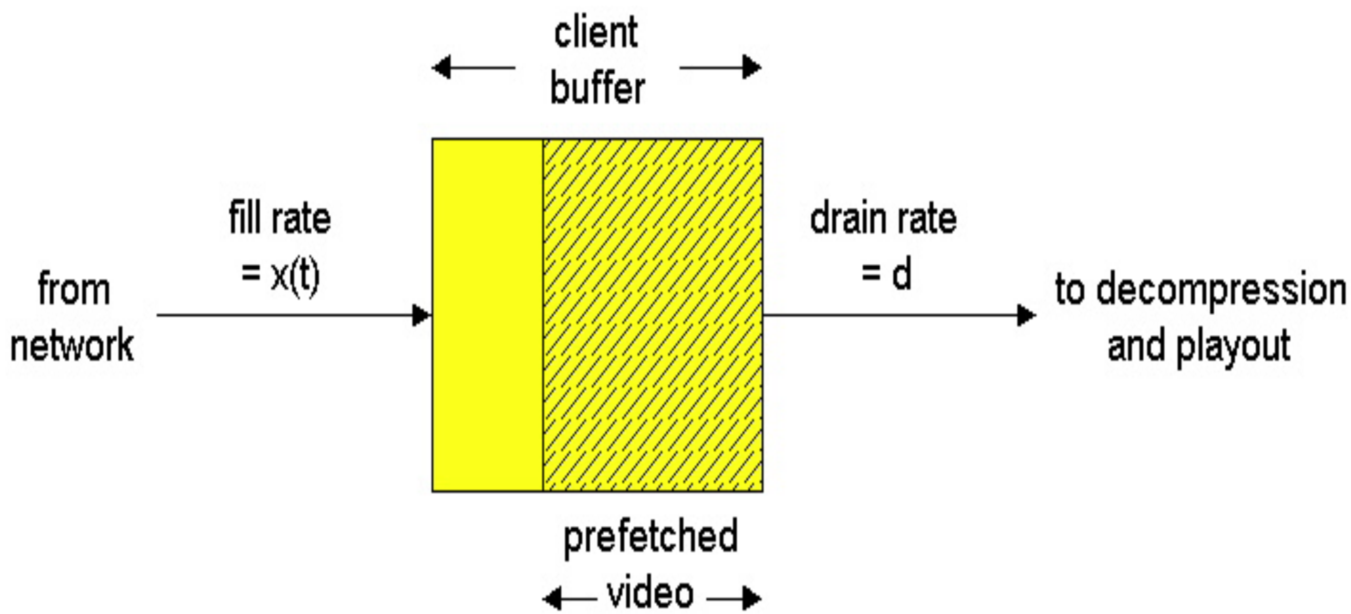
- **This architecture allows for non-HTTP protocol between server and media player**
- **Can also use UDP instead of TCP.**

Streaming Multimedia: Client Buffering



- **Client-side buffering, playout delay compensate for network-added delay, delay jitter**

Streaming Multimedia: Client Buffering



- **Client-side buffering, playout delay compensate for network-added delay, delay jitter**

Chapter 3: Multimedia Networking

Chair of
Future Communication

endowed by

Overview:

- 2.1 Multimedia Networking Applications
- 2.2 Streaming stored audio and video
- 2.3 Real-time Multimedia: Internet Phone study
- 2.4 Protocols for Real-Time Interactive Applications
  - RTP,RTCP
- 2.5 IP Telefony, SIP, and H.323
- 2.6 Distributing Multimedia: content distribution networks

Real-time interactive applications

- **PC-2-PC phone**
  - instant messaging services are providing this
- **PC-2-phone**
  - Dialpad
  - Net2phone
- **videoconference with Webcams**

Going to now look at a PC-2-PC Internet phone example in detail

Chair of
Future Communication

endowed by

A1 TELEKOM AUSTRIA

Introduce Internet Phone by way of an example

- **speaker's audio: alternating talk spurts, silent periods.**
  - 64 kbps during talk spurt
- **pkts generated only during talk spurts**
  - 20 msec chunks at 8 Kbytes/sec: 160 bytes data
- **application-layer header added to each chunk.**
- **Chunk+header encapsulated into UDP segment.**
- **application sends UDP segment into socket every 20 msec during talkspurt.**

Internet Phone: Packet Loss and Delay

universität wien

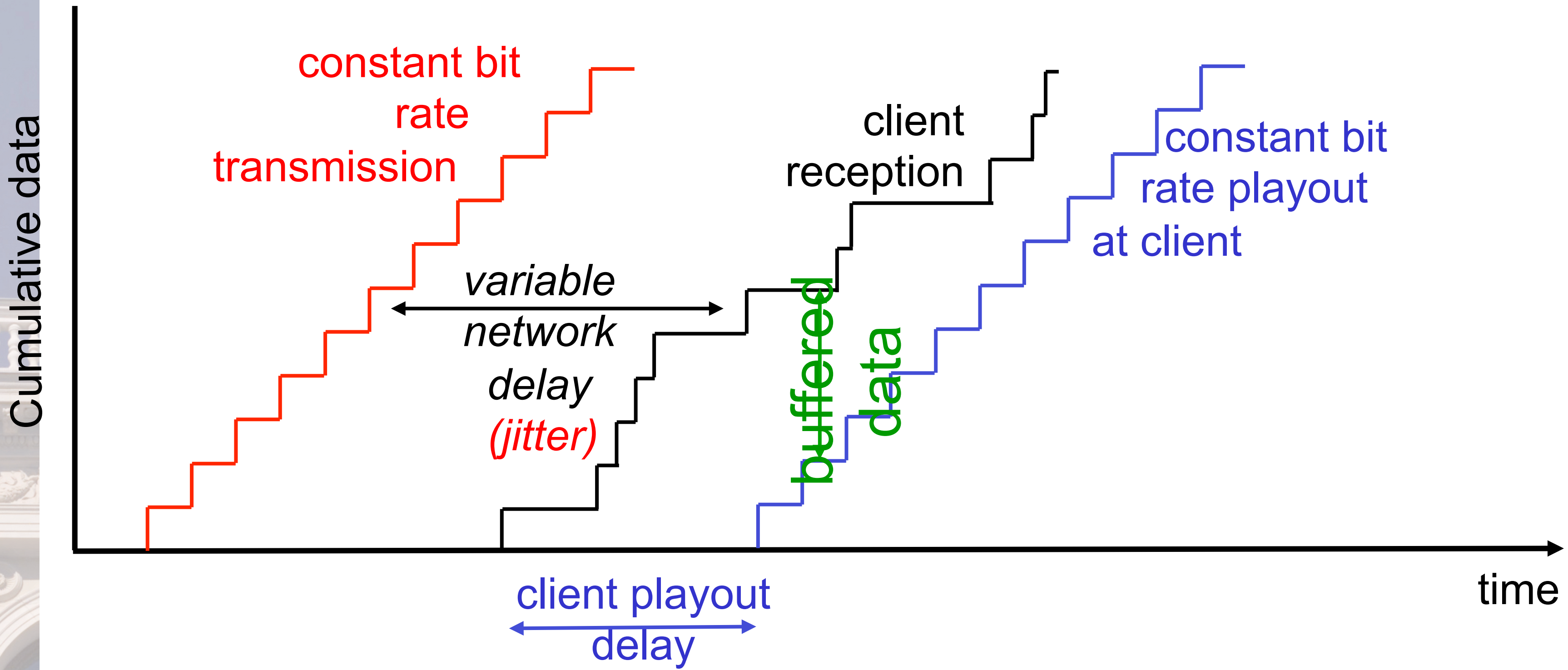
Chair of
Future Communication

endowed by

A1 TELEKOM AUSTRIA

- **network loss:** **IP datagram lost due to network congestion (router buffer overflow)**
- **delay loss:** **IP datagram arrives too late for playout at receiver**
  - delays: processing, queueing in network; end-system (sender, receiver) delays
  - typical maximum tolerable delay: 400 ms
- **loss tolerance: depending on voice encoding, losses concealed, packet loss rates between 1% and 10% can be tolerated.**

Delay Jitter



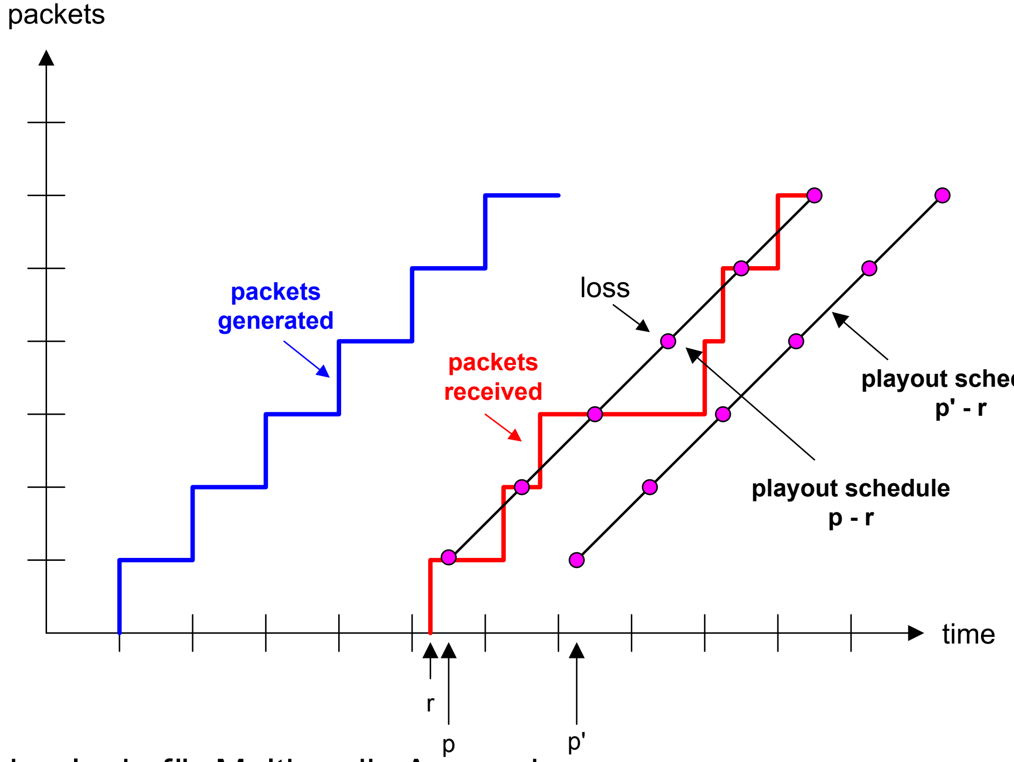
- **Consider the end-to-end delays of two consecutive packets: difference can be more or less than 20 msec**

Internet Phone: Fixed Playout Delay

- **Receiver attempts to playout each chunk exactly q msecs after chunk was generated.**
  - chunk has time stamp t: play out chunk at t+q .
  - chunk arrives after t+q: data arrives too late for playout, data "lost"
- **Tradeoff for q:**
  - large q: less packet loss
  - small q: better interactive experience

Fixed Playout Delay

- Sender generates packets every 20 msec during talk spurt.
- First packet received at time r
- First playout schedule: begins at p
- Second playout schedule: begins at p'



Adaptive Playout Delay, I

- **Goal: minimize playout delay, keeping late loss rate low**
- **Approach: adaptive playout delay adjustment:**
  - Estimate network delay, adjust playout delay at beginning of each talk spurt.
  - Silent periods compressed and elongated.
  - Chunks still played out every 20 msec during talk spurt.

t_i=timestamp of the ith packet

r_i=the time packet i is received by receiver

p_i=the time packet i is played at receiver

$r_i - t_i$=network delay for ith packet

d_i=estimate of average network delay after receiving ith packet

Dynamic estimate of average delay at receiver:

$$d_i = (1 - u)d_{i-1} + u(r_i - t_i)$$

where *u* is a fixed constant (e.g., *u* = .01).

Adaptive playout delay II

Chair of
Future Communication

endowed by

Also useful to estimate the average deviation of the delay, *vi* :

$$\boxed{v_i = (1-u)v_{i-1} + u\,|\,r_i - t_i - d_i\,|}$$

The estimates *di* and *vi* are calculated for every received packet, although

they are only used at the beginning of a talk spurt.

For first packet in talk spurt, playout time is:

$$\boxed{p_i = t_i + d_i + Kv_i}$$

where K is a positive constant.

Remaining packets in talkspurt are played out periodically

Adaptive Playout, III

Q: How does receiver determine whether packet is first in a talkspurt?

- **If no loss, receiver looks at successive timestamps.**
  – difference of successive stamps > 20 msec -->talk spurt begins.
- **With loss possible, receiver must look at both time stamps and sequence numbers.**
  – difference of successive stamps > 20 msec and sequence numbers without gaps --> talk spurt begins.

Chair of **Future Communication**

endowed by

Recovery from packet loss (1)

forward error correction (FEC): simple scheme

- **for every group of n chunks create a redundant chunk by exclusive OR-ing the n original chunks**
- **send out n+1 chunks, increasing the bandwidth by factor 1/n.**
- **can reconstruct the original n chunks if there is at most one lost chunk from the n+1 chunks**

- **Playout delay needs to be fixed to the time to receive all n+1 packets**
- **Tradeoff:**
  - increase n, less bandwidth waste
  - increase n, longer playout delay
  - increase n, higher probability that 2 or more chunks will be lost

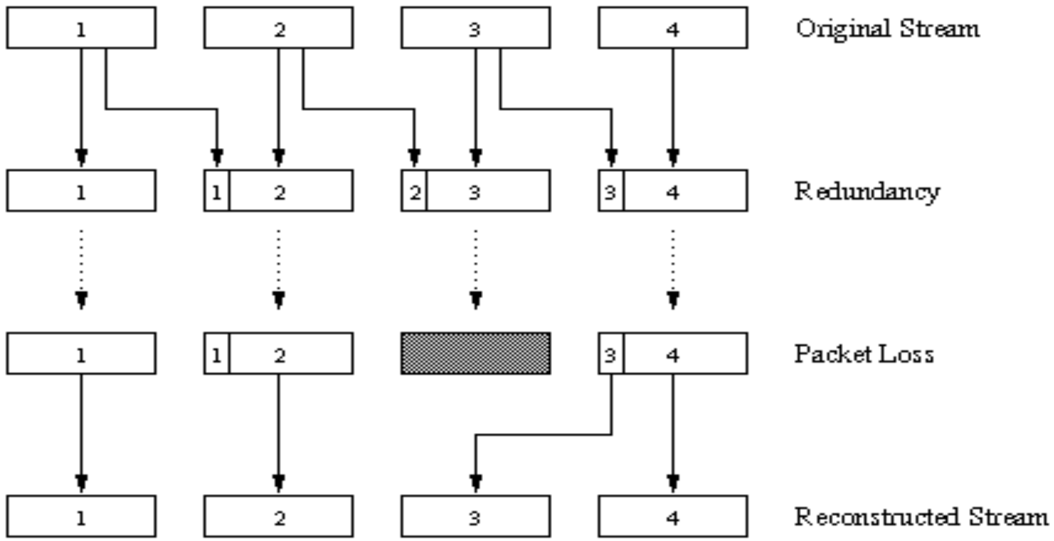
Recovery from packet loss (2)

Chair of
Future Communication

endowed by

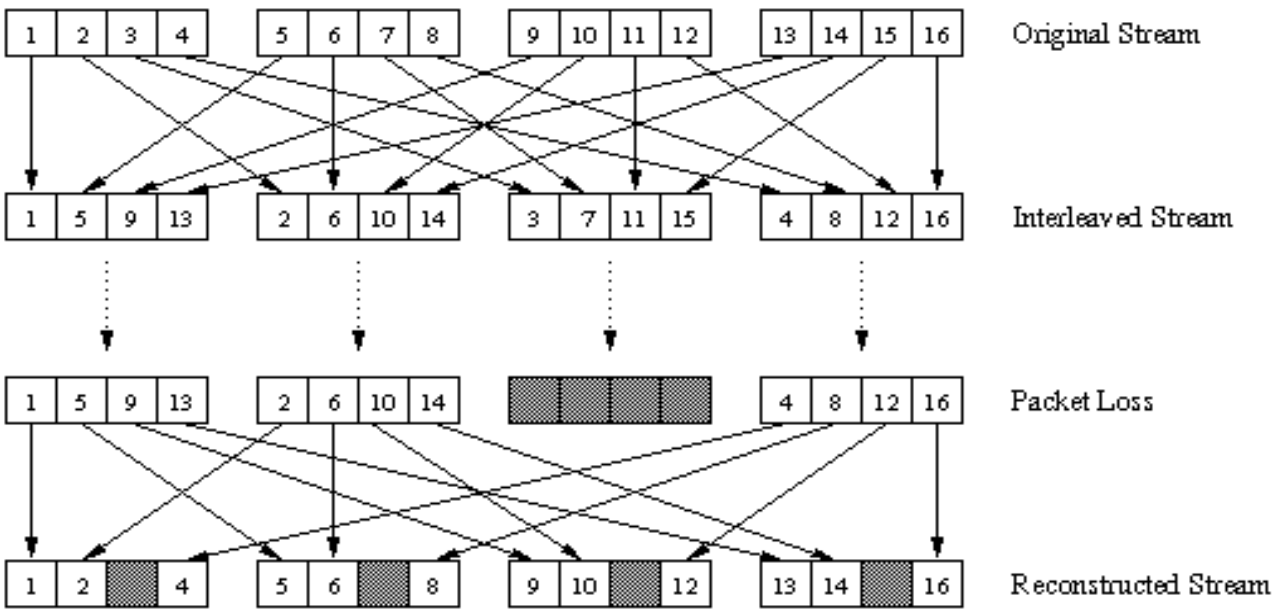
2nd FEC scheme

- "piggyback lower quality stream"
- send lower resolution audio stream as the redundant information
- for example, nominal stream PCM at 64 kbps
  and redundant stream GSM at 13 kbps.



- Whenever there is non-consecutive loss, the receiver can conceal the loss.
- Can also append (n-1)st and (n-2)nd low-bit rate chunk

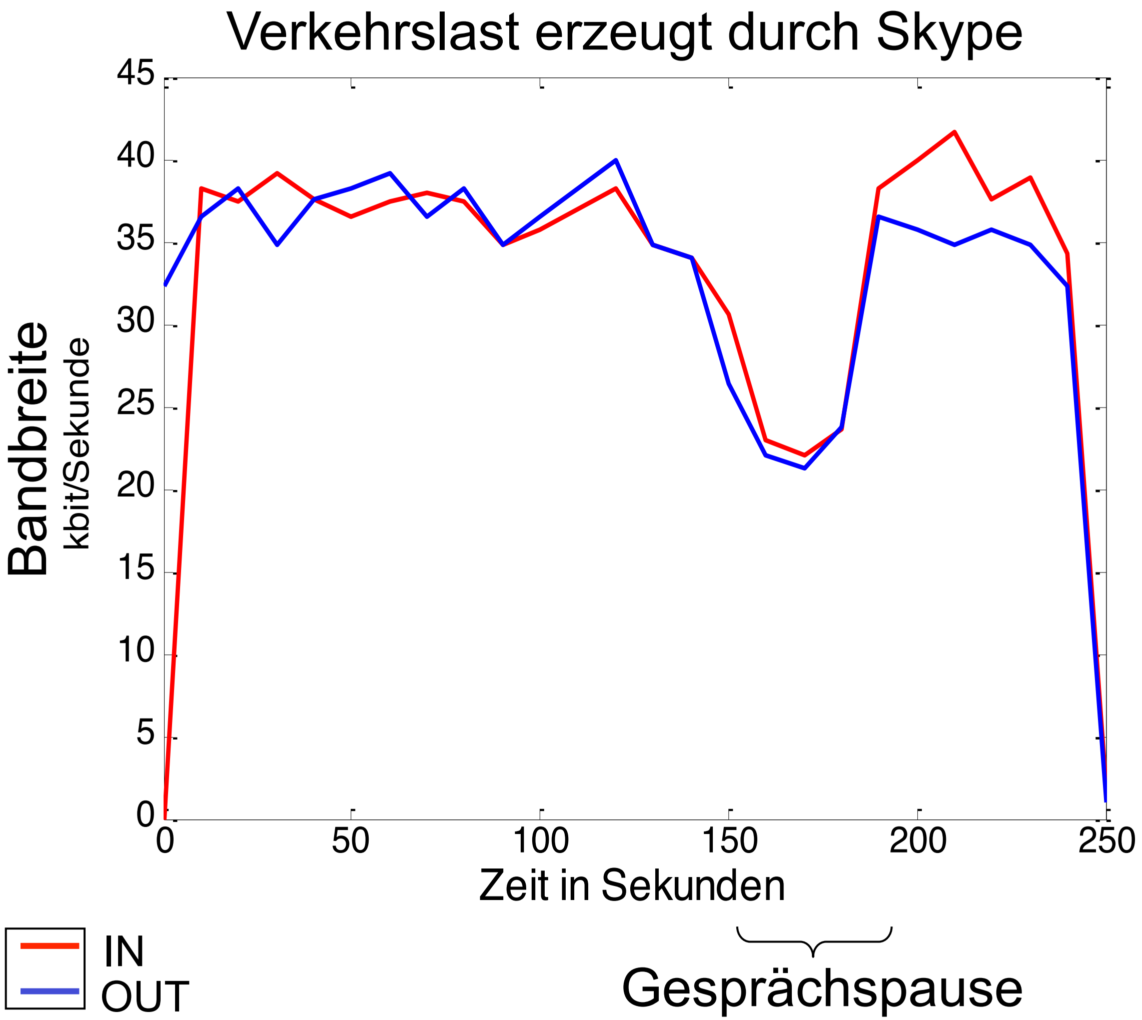
Recovery from packet loss (3)



Interleaving

- chunks are broken up into smaller units
- for example, 4 5 msec units per chunk
- Packet contains small units from different chunks
- if packet is lost, still have most of every chunk
- has no redundancy overhead
- but adds to playout delay

Bandbreitenmessung VoIP (Skype)



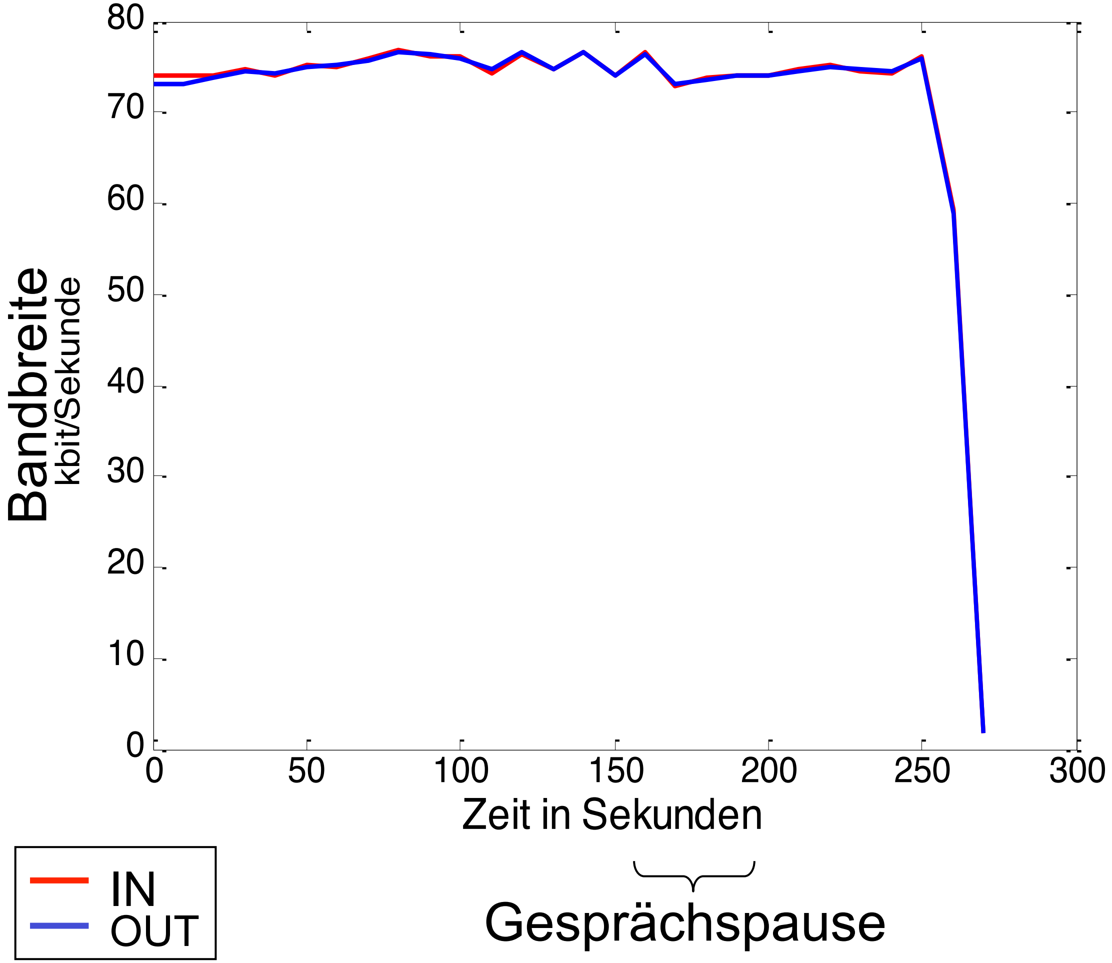
Mittelwert:
IN: 32,4 kbit/s
OUT: 32,6 kbit/s

Bandbreitenmessung VoIP (SIP)

Chair of
Future Communication

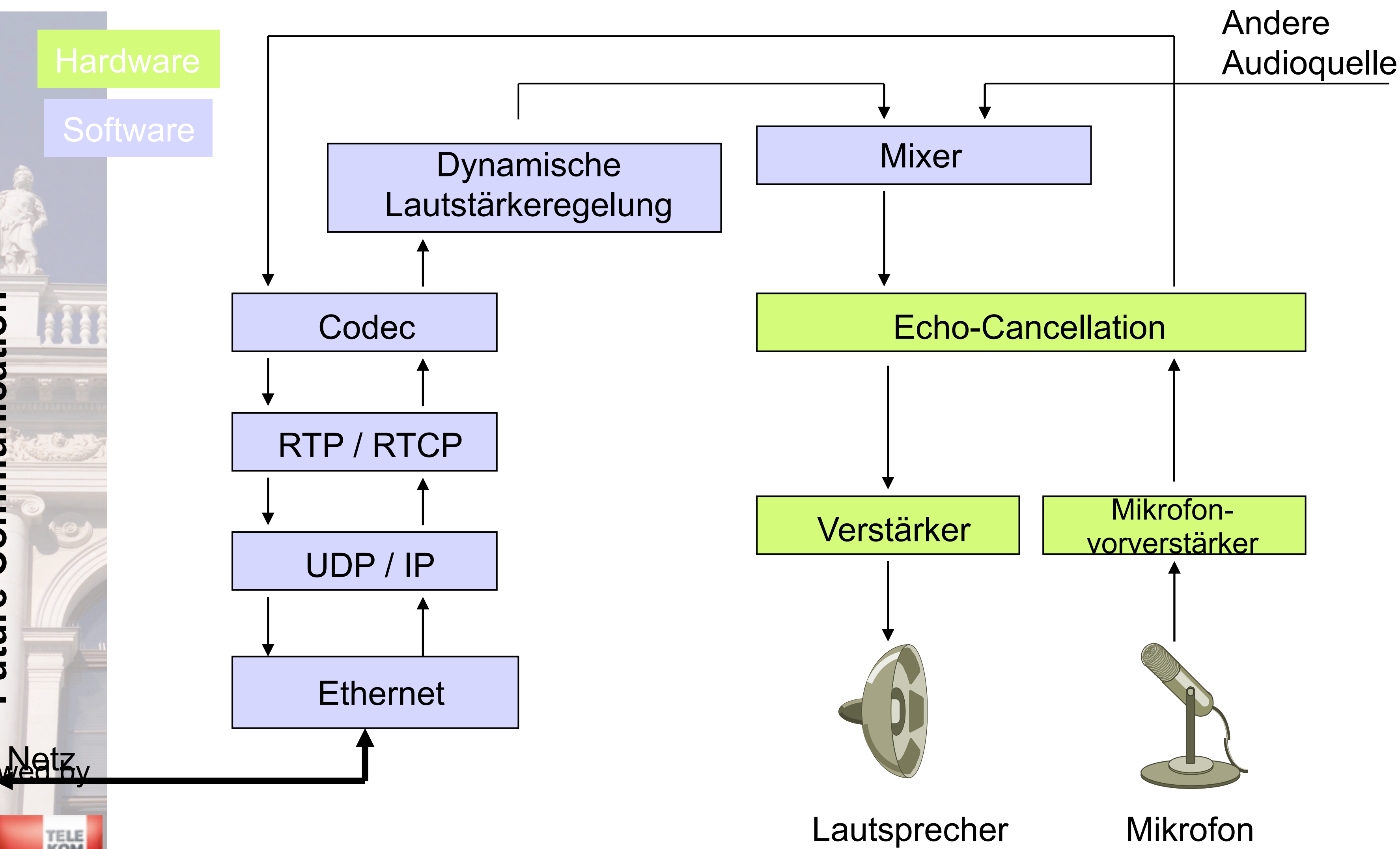
endowed by

Verkehrslast erzeugt durch SIP (Codec: G711u)



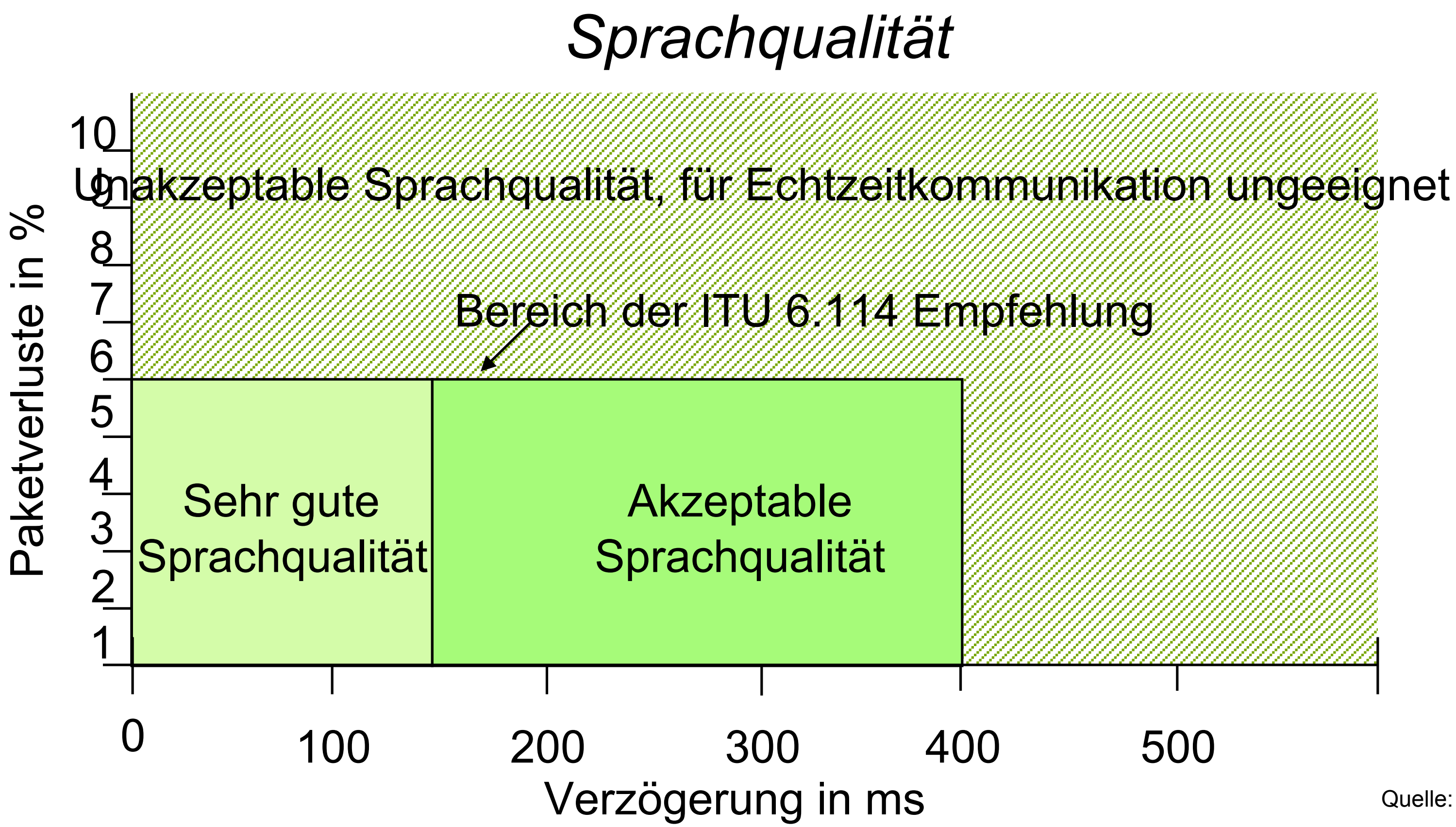
Mittelwert:
IN: 71,8 kbit/s
OUT: 71,6 kbit/s

Lokale Enflussfaktoren - Beispiel PC



Anforderungen Sprachqualität

- **Die Qualität der Sprache in IP-Netzen hängt wesentlich von den Paketverlusten und von der Verzögerung ab.**



Quelle: swyx

Bewertung der Übertragungsgüte

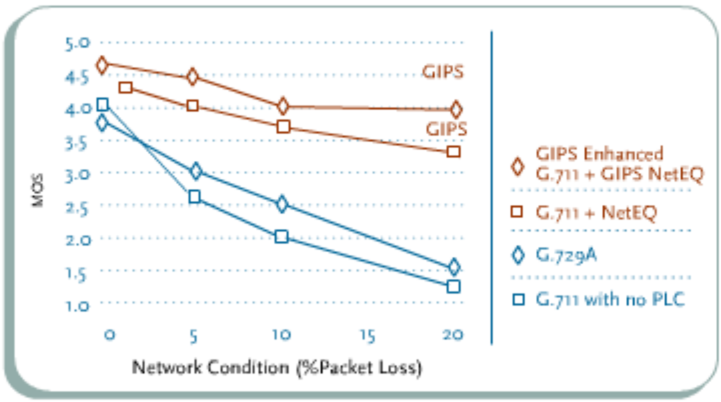
- **Bewertung der subjektiven Übertragungsqualität (Quality-of-Experience, QoE)**
  - Übertragung einer vorgegebenen Audiodatei
  - Aufzeichnung am Empfänger
  - Vergleich mittels eines standardisierten Algorithmus ergibt:
    - **Mean Opinion Score (MOS)**

Sehr gute Sprachqualität in leiser Umgebung	Excellent	5.0
Natürliche Sprachqualität wie digitales Telefon	Good	4.0
Akzeptabel, erfordert aber teilweise Konzentration	Fair	3.0
Schwer zu verstehende Sprache	Poor	2.0
Kaum zu verstehen, Unterbrechungen	Bad	1.0

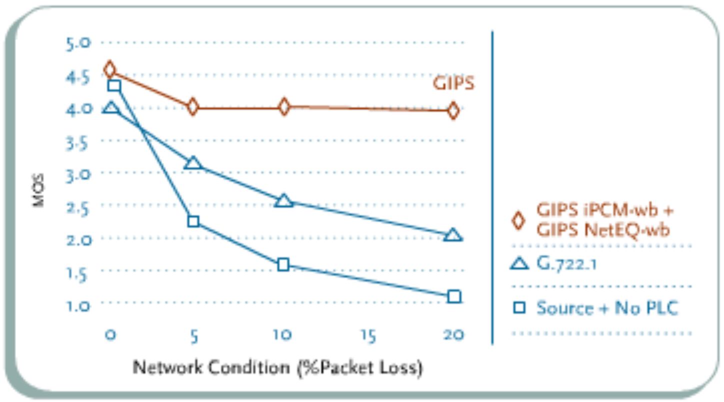
Skype - Sprachqualität

- **Einsatz eines speziellen Voicecodecs**
  - Global IP Sound

Telephony Bandwidth



Wideband Speech



  - Reagiert adaptiv auf Bandbreitenveränderungen
  - Kann das Verhältnis Prozessor-/Bandbreitenlast optimieren

Quelle: www.globalipsound.com

Our Research: What do I have to do when I want to apply QoE?

Chair of **Future Communication**

1. How can QoE be measured in general?

2. What metric can be applied and how can this metric be matched with the providers view of the networks?

3. How can a scalable, adaptive, inter-domain measurement concept be implemented?

However:

- Applications require different metrics!
- Is a generic measurement concept feasible at all?
- What happens if domains do not collaborate?

endowed by

universität wien

Our Research: The IQX Hypothesis

Chair of
Future Communication

endowed by

Example:

If we dined in a five-star restaurant, a single spot on the clean white table cloth strongly disturbs the experience. The same incident appears much less severe in a beer pub.

The IQX Hypothesis [Hossfeld, Fiedler, Tutschku et al. ’07/08]

- QoE is function QoS, i.e. $QoE = f(QoS)$
- The subjective sensibility on QoE is more sensitive, the higher the experienced quality is.
- If the QoE is very high, a small disruption (i.e. decrease of QoS) will decrease strongly the QoE.
- Mathematical description: $\frac{\partial QoE}{\partial QoS} = -\tilde{\beta} \cdot (QoE - \gamma).$

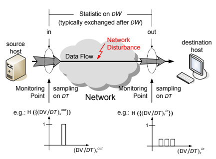
→ Only possible solution: **exponential relationship (IQX Hypothesis)**

$$\boxed{QoE = \alpha \cdot \exp(-\beta \cdot QoS) + \gamma.}$$

End-to-End Comparative Monitoring

Chair of **Future Communication**

endowed by



Possible application : audio/video quality

MOS (Mean Opinion Score; ITU-T. rec. P.800.1): numerical indication of the perceived quality of received media after compression and/or transmission

MOS range:

MOS	quality	Impairment
5	excellent	imperceptible
4	good	perceptible but not annoying
3	fair	slightly annoying
2	poor	annoying
1	bad	very annoying

Multiplicative relationship model

$$(MOS_{Out} - 1) = f_{Utility}(MOS_{In}) \approx U_{Netw,MOS} \cdot (MOS_{In} - 1)$$

Test of the IQX Hypothesis

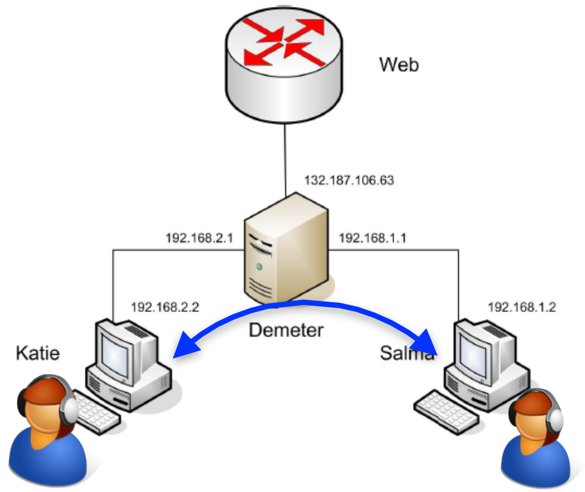
Chair of **Future Communication**

Case Study: **Audio Codecs**

- iLBC
- G.711

Setup:

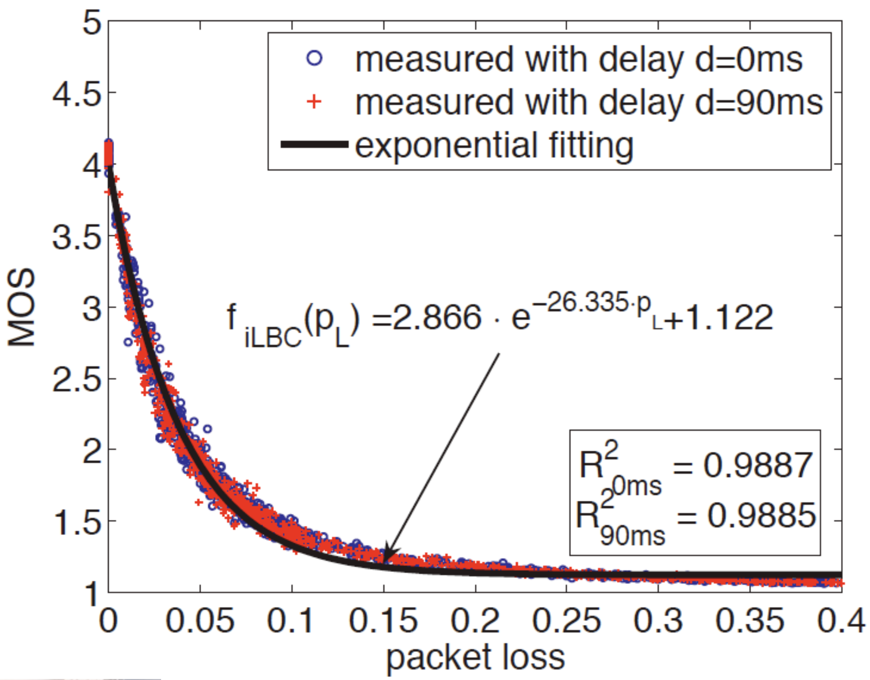
- use of comparative measurements
- computation of PESQ values mapped to MOS at input and output
- automatic initiation of test calls
- variation of QoS, i.e., of packet delay/packet loss, at inter-connecting router (Demeter)
- non-linear regression for fitting QoS onto the QoE mapping function:



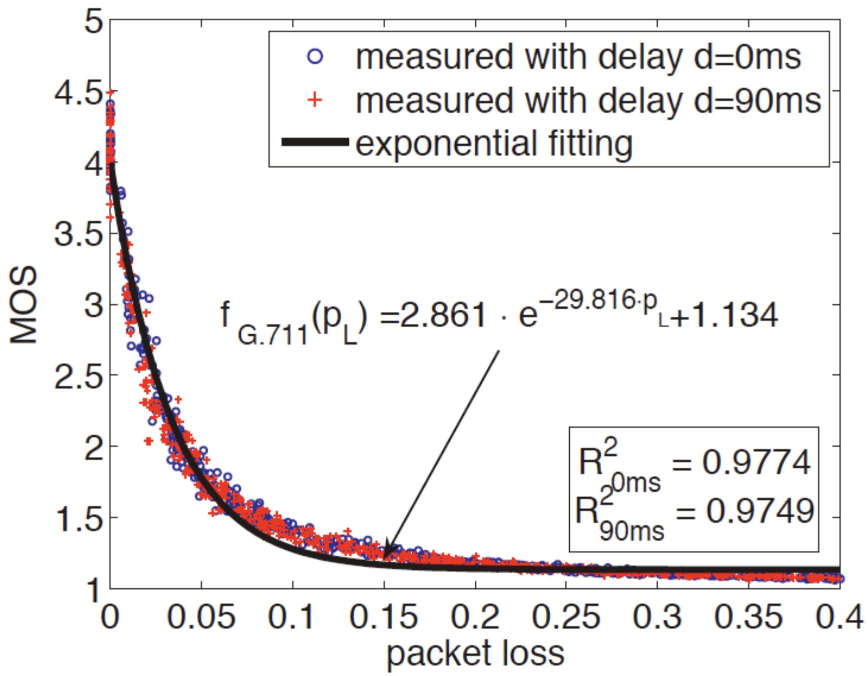
$$QoE = \alpha \cdot \exp(-\beta \cdot QoS) + \gamma.$$

endowed by

Test of the IQX Hypothesis: Results for Packet Loss

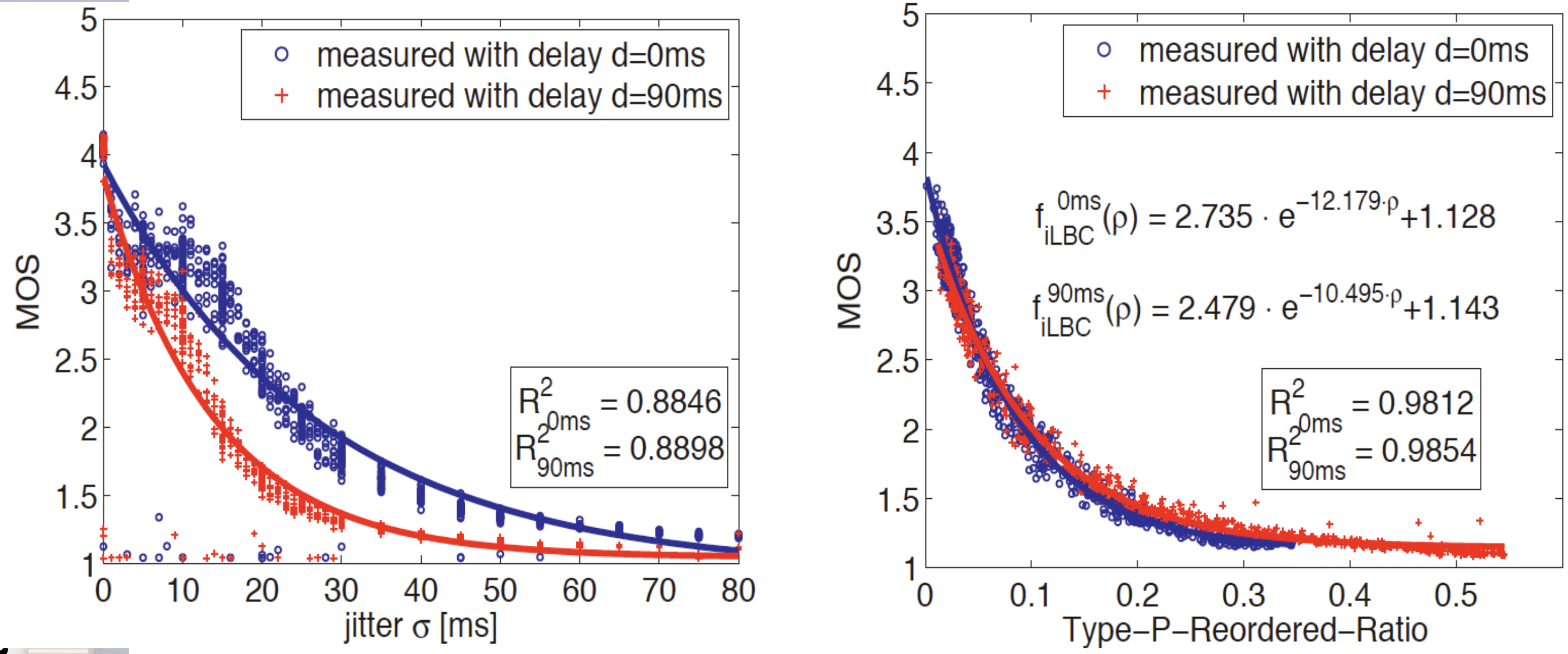


iLBC Codec



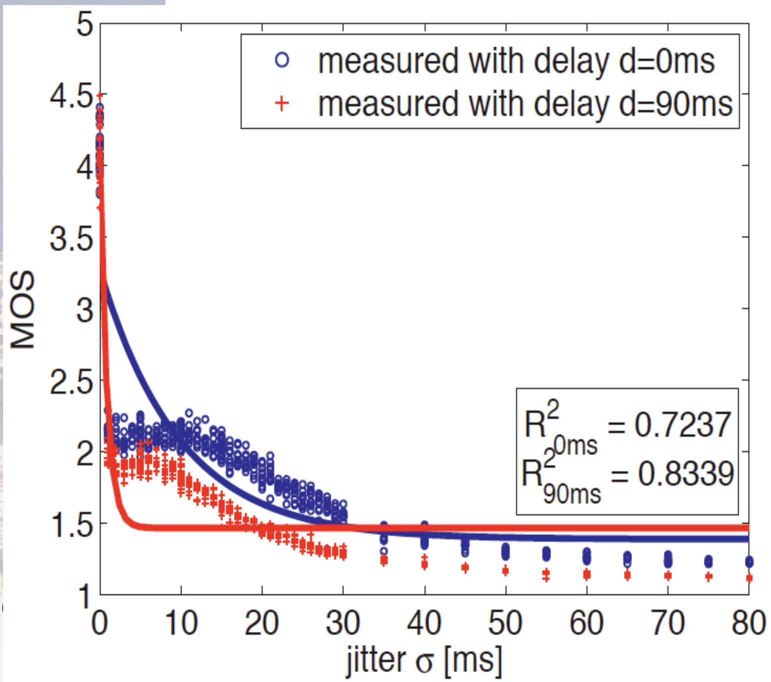
G.711 Codec

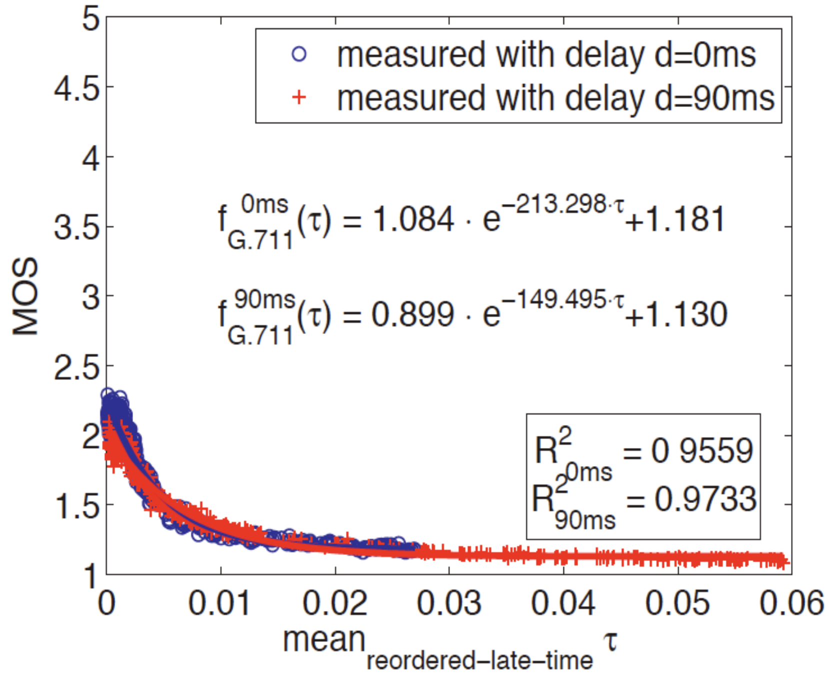
Test of the IQX Hypothesis: Results For Jitter I



iLBC Codec

Test of the IQX Hypothesis: Results For Jitter II



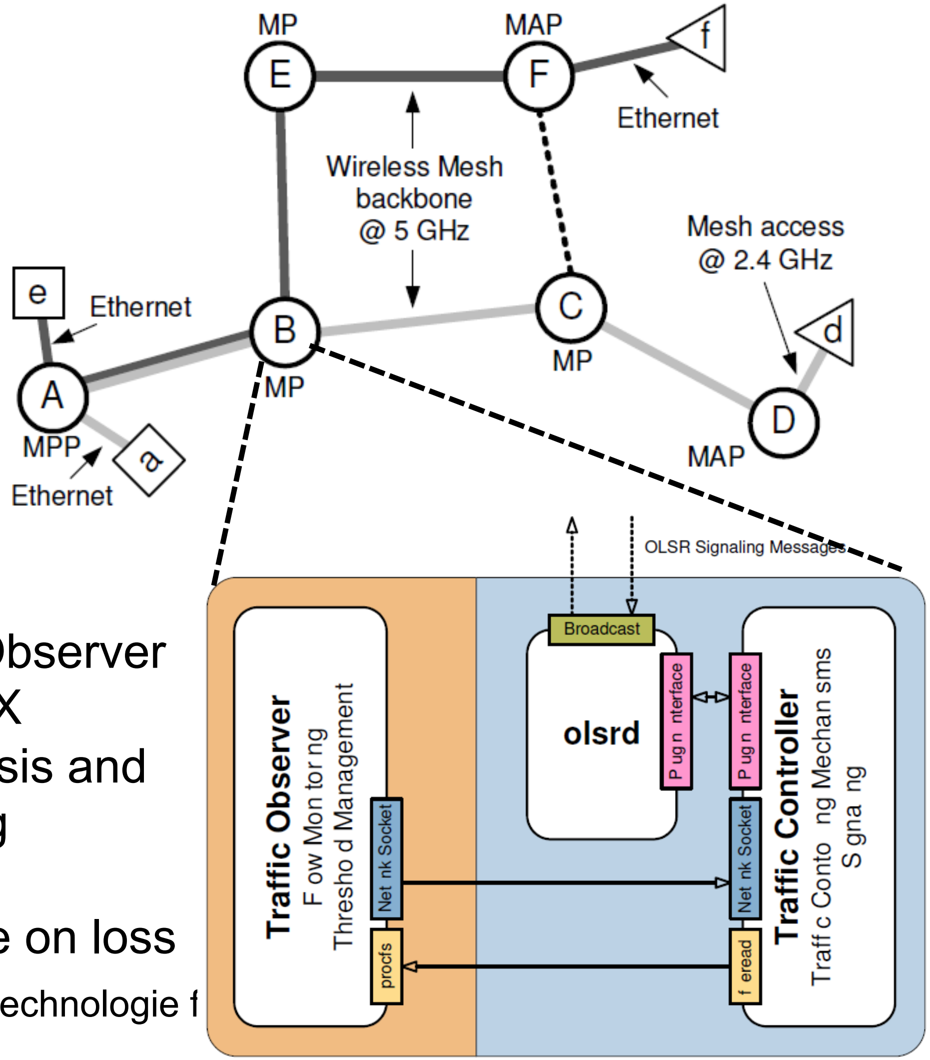


G.711 Codec

universität wien

Adaptive Traffic Management Based on QoE and the IQX Hypothesis

Dynamic Bandwidth Control in Wireless Mesh Networks
[Pries et al. ‘08]



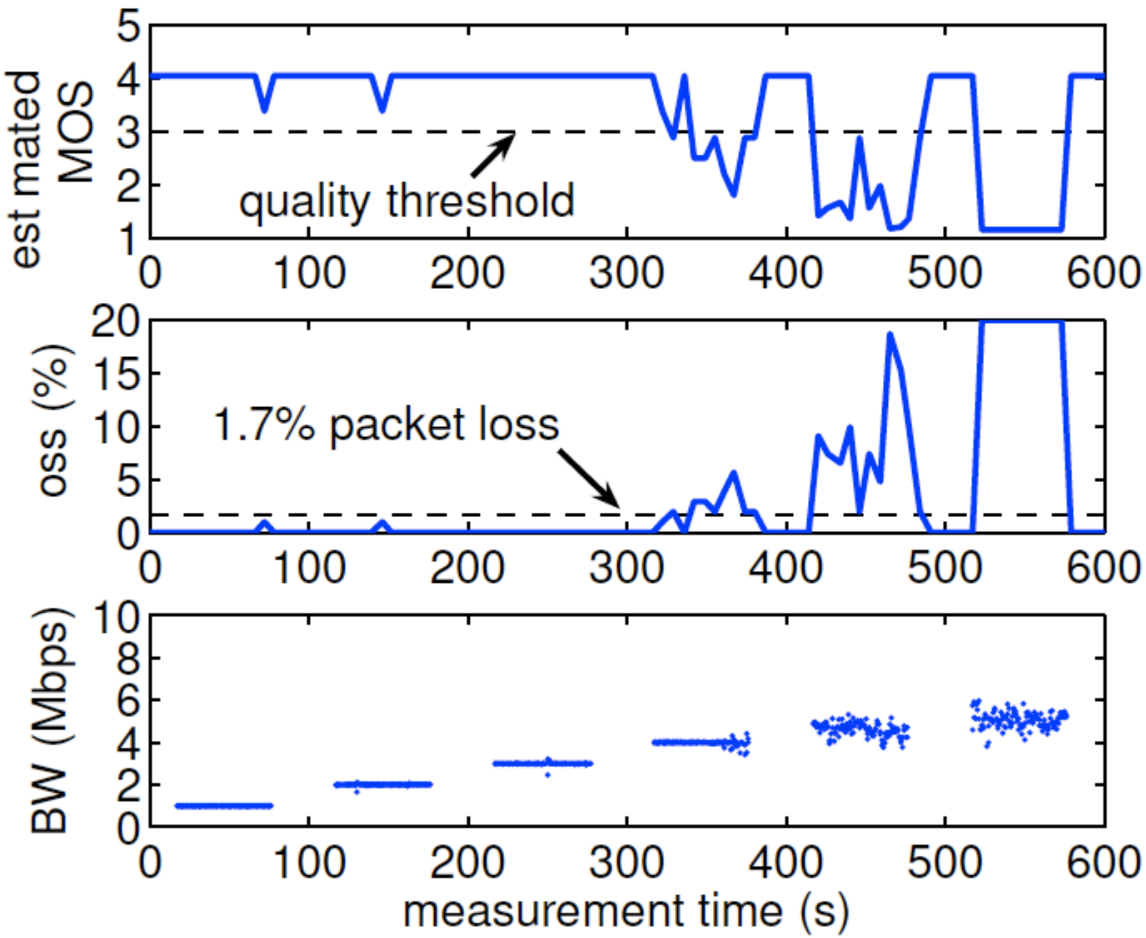
Traffic Observer uses IQX hypothesis and mapping function sensitive on loss

Routing Controller

Chair of **Future Communication**

endowed by

Adaptive Traffic Management Based on QoE and the IQX Hypothesis -- Results



Summary: Internet Multimedia: bag of tricks

- **use UDP to avoid TCP congestion control (delays) for time-sensitive traffic**
- **client-side adaptive playout delay: to compensate for delay**
- **server side matches stream bandwidth to available client-to-server path bandwidth**
  - chose among pre-encoded stream rates
  - dynamic server encoding rate
- **error recovery (on top of UDP)**
  - FEC, interleaving
  - retransmissions, time permitting
  - conceal errors: repeat nearby data

Chapter 3: Multimedia Networking

Overview:

- 2.1 Multimedia Networking Applications
- 2.2 Streaming stored audio and video
- 2.3 Real-time Multimedia: Internet Phone study
- 2.4 Protocols for Real-Time Interactive Applications
  - RTP,RTCP
- 2.5 IP Telefony, SIP, and H.323
- 2.6 Distributing Multimedia: content distribution networks

Chair of **Future Communication**

endowed by

Real-Time Protocol (RTP)

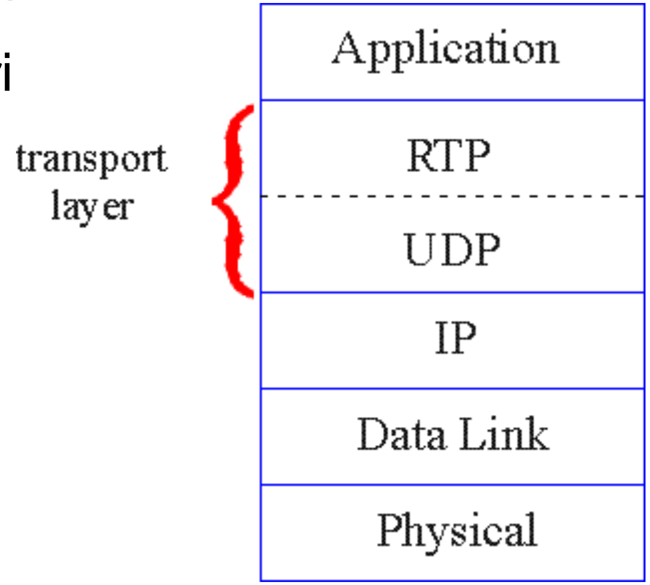
- **RTP specifies a packet structure for packets carrying audio and video data**
- **RFC 1889.**
- **RTP packet provides**
  - payload type identification
  - packet sequence numbering
  - timestamping
- **RTP runs in the end systems.**
- **RTP packets are encapsulated in UDP segments**
- **Interoperability: If two Internet phone applications run RTP, then they may be able to work together**

RTP and QoS

- **RTP does not provide any mechanism to ensure timely delivery of data or provide other quality of service guarantees.**
- **RTP encapsulation is only seen at the end systems: it is not seen by intermediate routers.**
  - Routers providing best-effort service do not make any special effort to ensure that RTP packets arrive at the destination in a timely matter.

RTP runs on top of UDP

- **RTP libraries provide a transport-layer interface**
- **that extend UDP:**
  - port numbers, IP addresses
  - payload type identification
  - packet sequence numberi
  - time-stamping



universität wien

RTP Example

Chair of
Future Communication

endowed by

- **Consider sending 64 kbps PCM-encoded voice over RTP.**
- **Application collects the encoded data in chunks, e.g., every 20 msec = 160 bytes in a chunk.**
- **The audio chunk along with the RTP header form the RTP packet, which is encapsulated into a UDP segment.**
- **RTP header indicates type of audio encoding in each packet**
  - sender can change encoding during a conference.
- **RTP header also contains sequence numbers and timestamps**

RTP Header

RTP Header

- **Payload Type (7 bits)**
  - Indicates type of encoding currently being used.
  - If sender changes encoding in middle of conference, sender informs the receiver through this payload type field.
    - Payload type 0: PCM mu-law, 64 kbps
    - Payload type 3, GSM, 13 kbps
    - Payload type 7, LPC, 2.4 kbps
    - Payload type 26, Motion JPEG
    - Payload type 31. H.261
    - Payload type 33, MPEG2 video
- **Sequence Number (16 bits)**
  - Increments by one for each RTP packet sent
  - May be used to detect packet loss and to restore packet sequence.

RTP Header (2)

- **Timestamp field (32 bits long)**
  - Reflects the sampling instant of the first byte in the RTP data packet.
  - For audio, timestamp clock typically increments by one for each sampling period (for example, each 125 usecs for a 8 KHz sampling clock)
  - If application generates chunks of 160 encoded samples, then timestamp increases by 160 for each RTP packet when source is active. Timestamp clock continues to increase at constant rate when source is inactive.

- **SSRC field (32 bits long)**
  - Identifies the source of the RTP stream.
  - Each stream in a RTP session should have a distinct SSRC.

universität wien

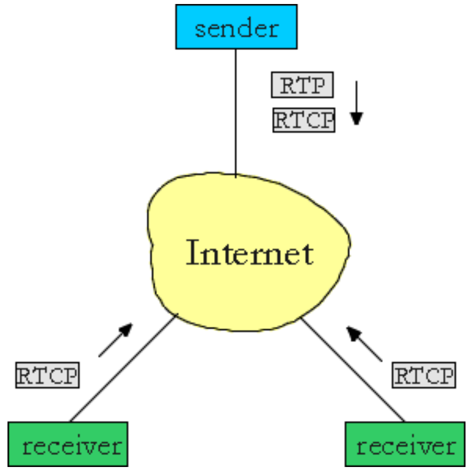
Real-Time Control Protocol (RTCP)

Chair of **Future Communication**

endowed by

- **Works in conjunction with RTP.**
- **Each participant in RTP session periodically transmits RTCP control packets to all other participants.**
- **RTCP packets contain sender and/or receiver reports**
  - report statistics useful to application
- **Statistics include**
  - number of packets sent,
  - number of packets lost,
  - interarrival jitter, etc.
- **Feedback can be used to control performance**
  - Sender may modify its transmissions based on feedback

RTCP - Continued



- **For an RTP session there is typically a single multicast address; all RTP and RTCP packets belonging to the session use the multicast address.**
  - RTP and RTCP packets are distinguished from each other through the use of distinct port numbers.
  - To limit traffic, each participant reduces his RTCP traffic as the number of conference participants increases.

RTCP Packets

Receiver report packets:

- Fraction of lost packets,
- last sequence number,
- average interarrival jitter.

Sender report packets:

- SSRC of the RTP stream,
- the current time,
- the last time stamp
- the number of sent packets,
- and the number of sent bytes.

Source description packets:

- e-mail address of sender, sender's name, SSRC of associated RTP stream.
- Provide mapping between the SSRC and the user/host name

Synchronization of Streams

- **RTCP can synchronize different media streams within a RTP session.**
- **Consider videoconferencing app for which each sender generates one RTP stream for video and one for audio.**
- **Timestamps in RTP packets tied to the video and audio sampling clocks**
  - not tied to the wall-clock time
- **Each RTCP sender-report packet contains (for the most recently generated packet in the associated RTP stream):**
  - timestamp of the RTP packet
  - wall-clock time for when packet was created.
- **Receivers can use this association to synchronize the playout of audio and video.**

RTCP Bandwidth Scaling

- **RTCP attempts to limit its traffic to 5% of the session bandwidth.**

Example

- **Suppose one sender, sending video at a rate of 2 Mbps. Then RTCP attempts to limit its traffic to 100 Kbit/s.**
- **RTCP gives 75% of this rate to the receivers; remaining 25% to the sender**

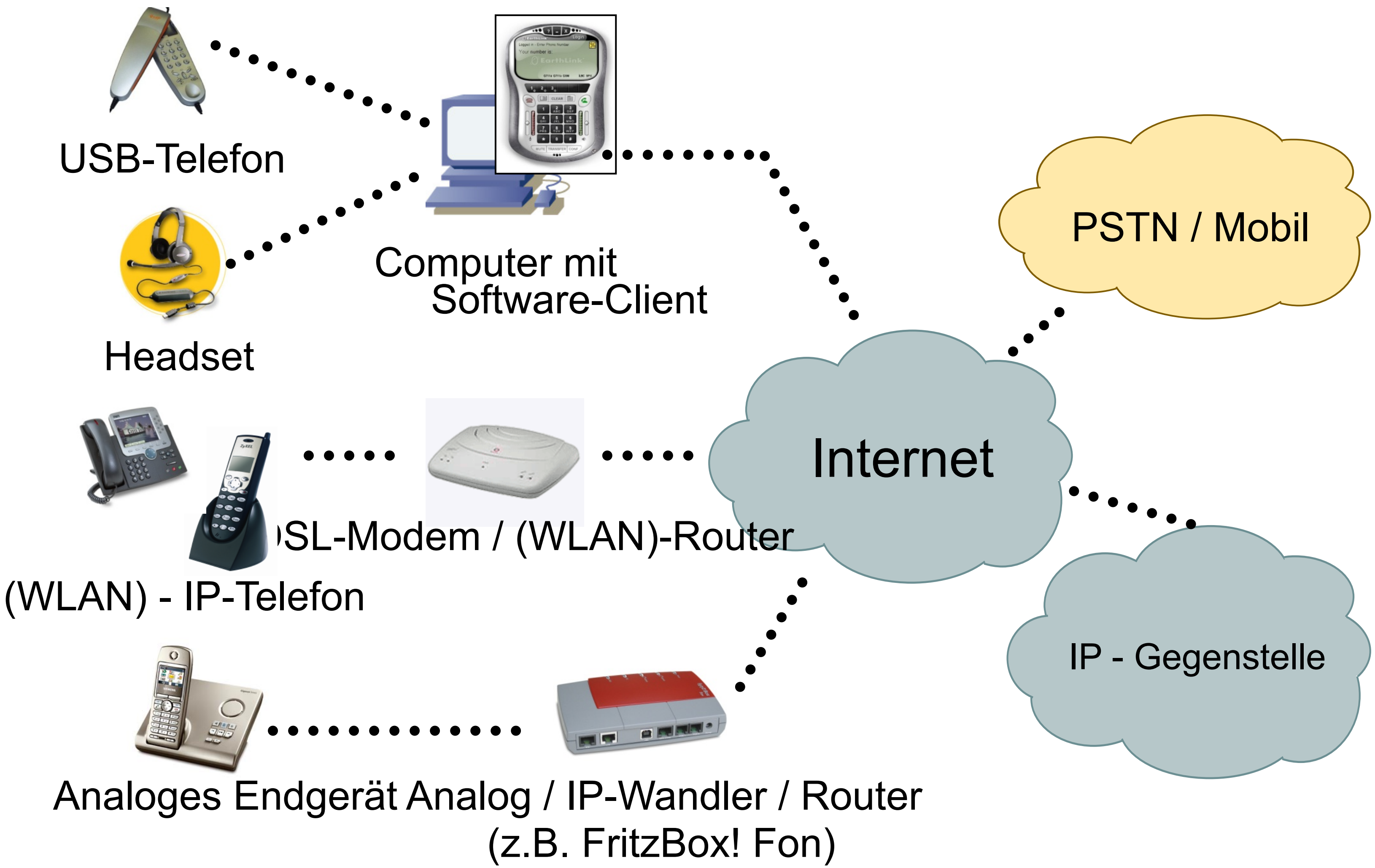
- **The 75 kbps is equally shared among receivers:**
  - With R receivers, each receiver gets to send RTCP traffic at 75/R kbps.
- **Sender gets to send RTCP traffic at 25 kbps.**
- **Participant determines RTCP packet transmission period by calculating avg RTCP packet size (across the entire session) and dividing by allocated rate.**

Chapter 3: Multimedia Networking

Overview:

- 2.1 Multimedia Networking Applications
- 2.2 Streaming stored audio and video
- 2.3 Real-time Multimedia: Internet Phone study
- 2.4 Protocols for Real-Time Interactive Applications
  - RTP,RTCP
- 2.5 IP Telefony, SIP, and H.323
- 2.6 Distributing Multimedia: content distribution networks

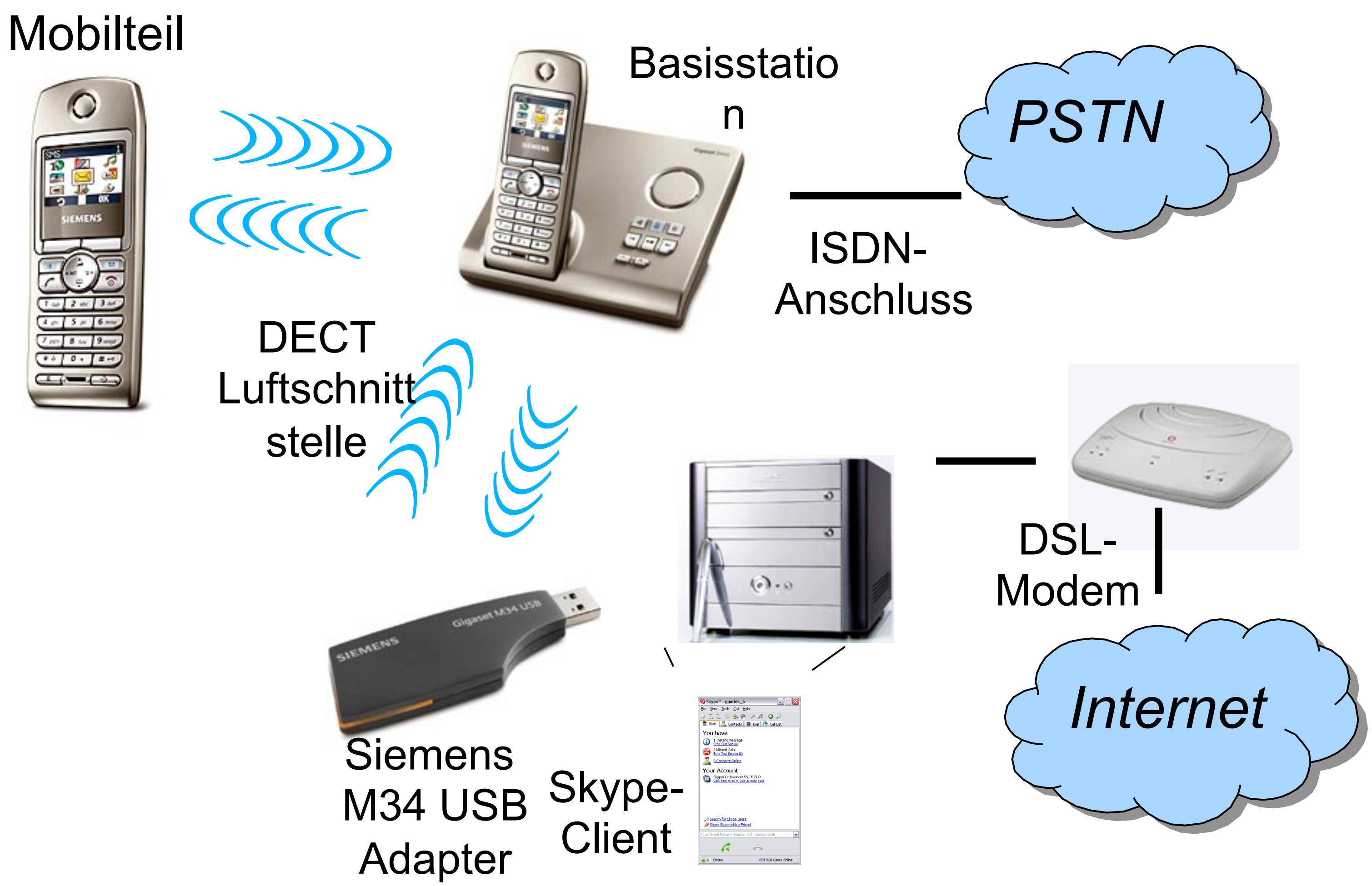
Zugangsmöglichkeiten zu VoIP



Chair of
Future Communication

endowed by

Skype / DECT Beispiel-Installation

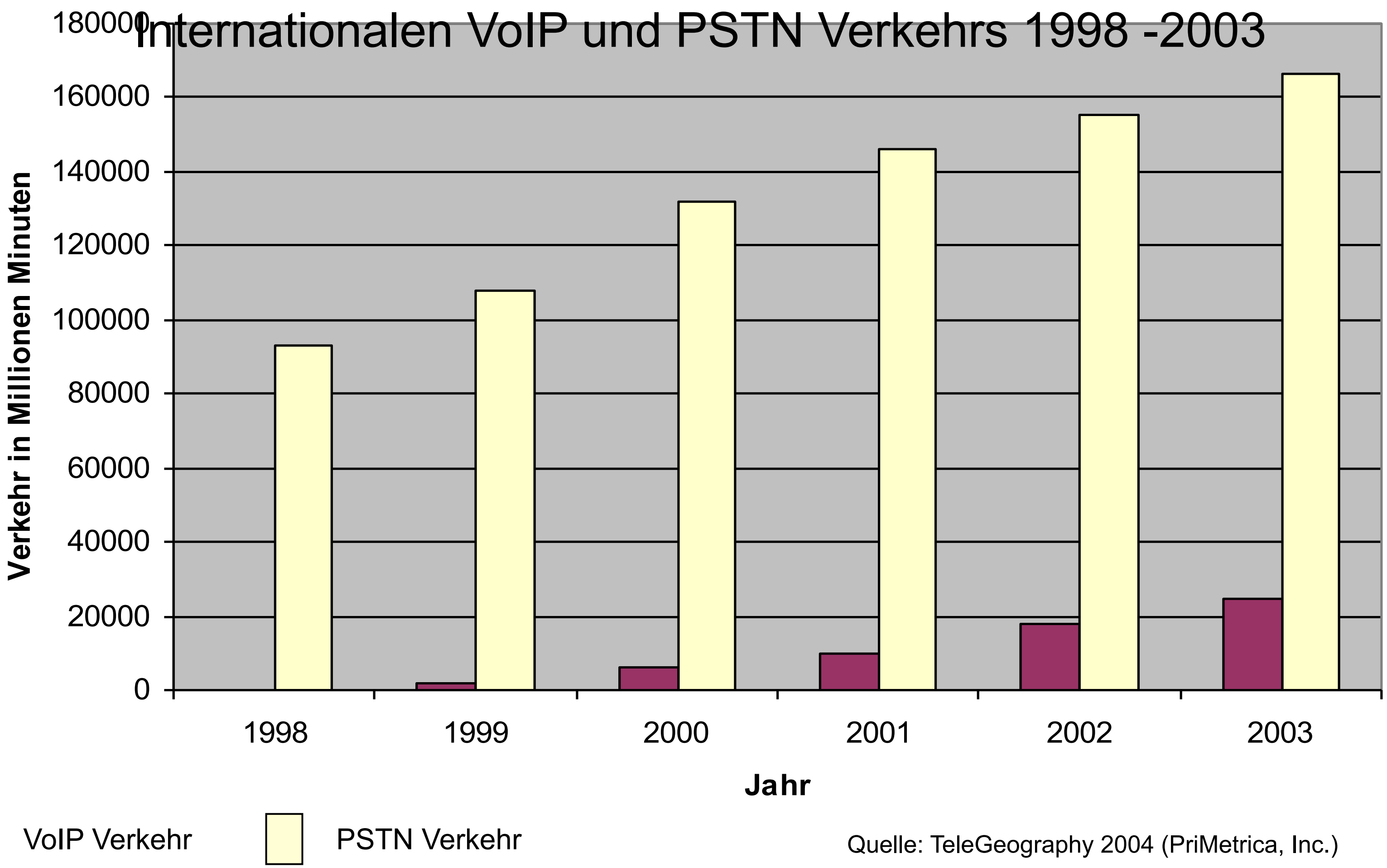


Chair of
Future Communication

endowed by

VoIP-Entwicklung in Zahlen

Chair of
Future Communication



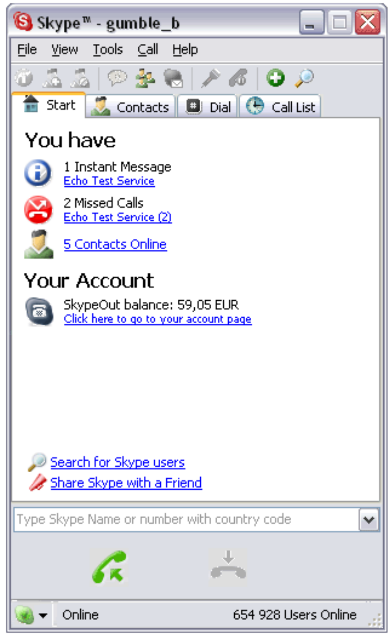
Quelle: TeleGeography 2004 (PriMetrica, Inc.)

endowed by

Freie Softwarekomponenten

- **Softwareunterstützung für VoIP**

SIP-fähige Clients



Skype

ICQ

Netmeeting

Chair of
Future Communication

endowed by

Voice over IP - International

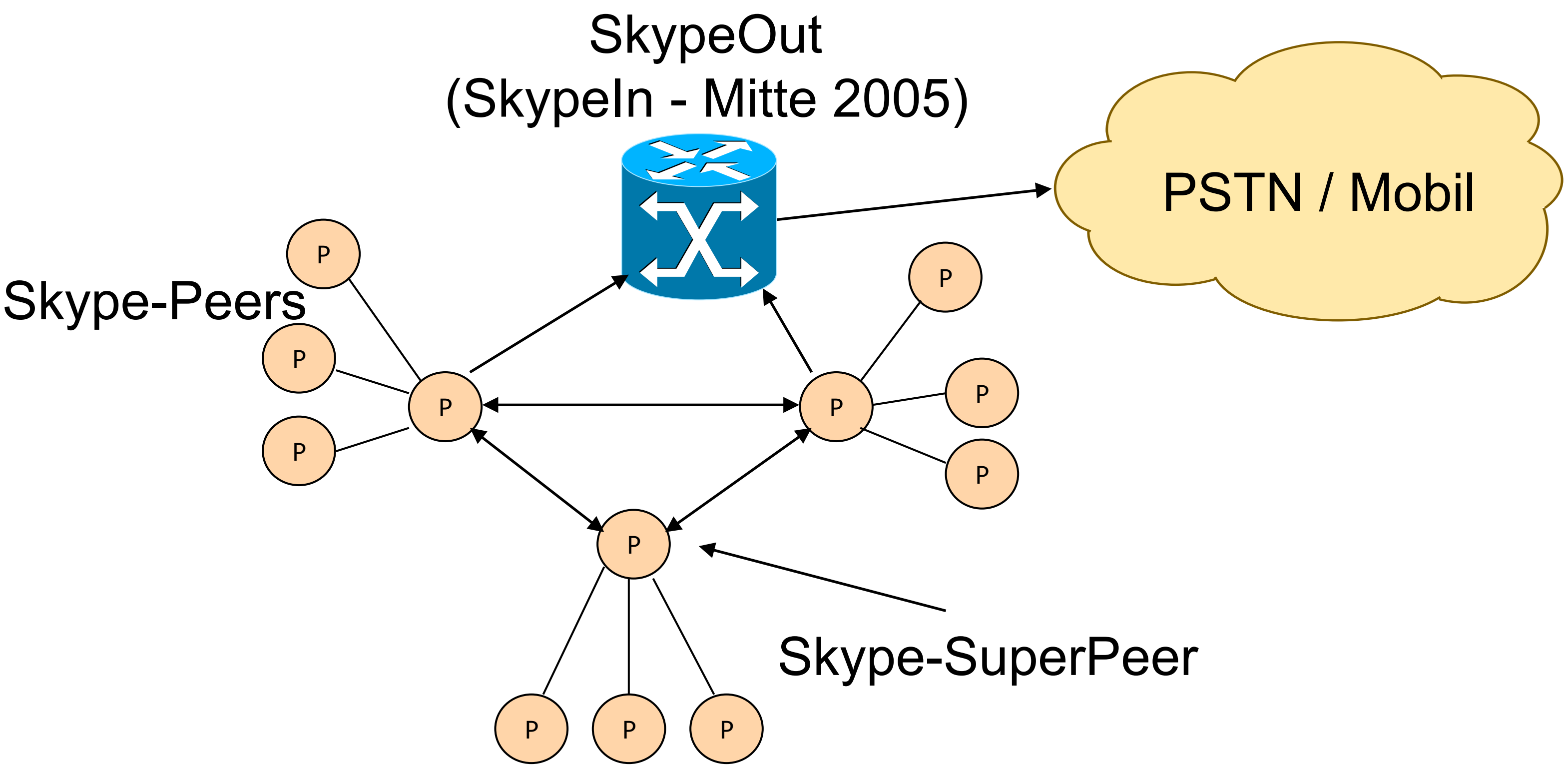
- **China**
  - 50% des Ferngesprächsverkehrs über VoIP
  - China Unicom: 29 Mrd VoIP von insgesamt 54 Mrd. Gesprächsminuten
  - Auch China Mobile setzt auf IP-Telefonie

- **Australien**
  - Mehr Internettelefonsysteme als klassische Nebenstellenanlagen verkauft
  - Anstieg der Umsätze von VoIP-Systemen um 175% gegenüber 13,2% weniger Verkauf traditioneller Anlagen

Quelle: pressetext deutschland

Skype – Ein P2P-Ansatz

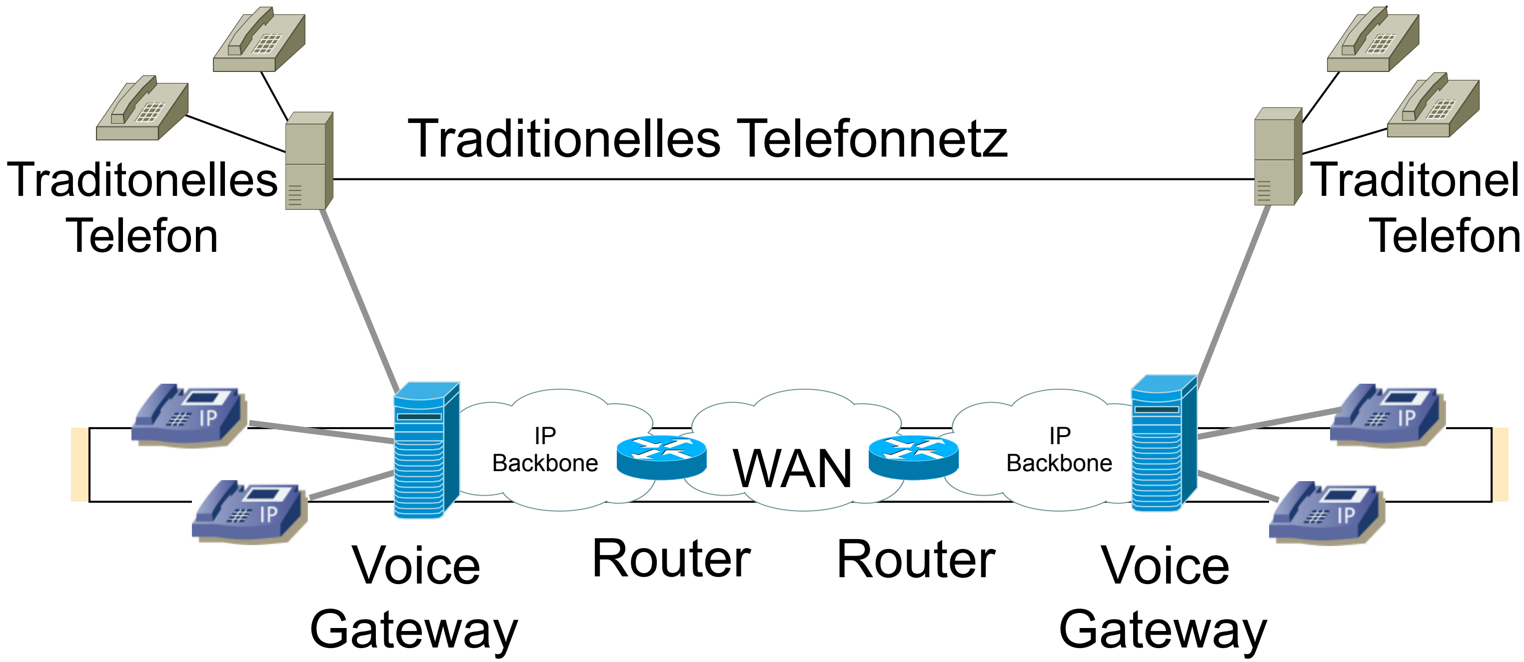
- **Skype ist aus der KaZaa-Entwicklung hervorgegangen**



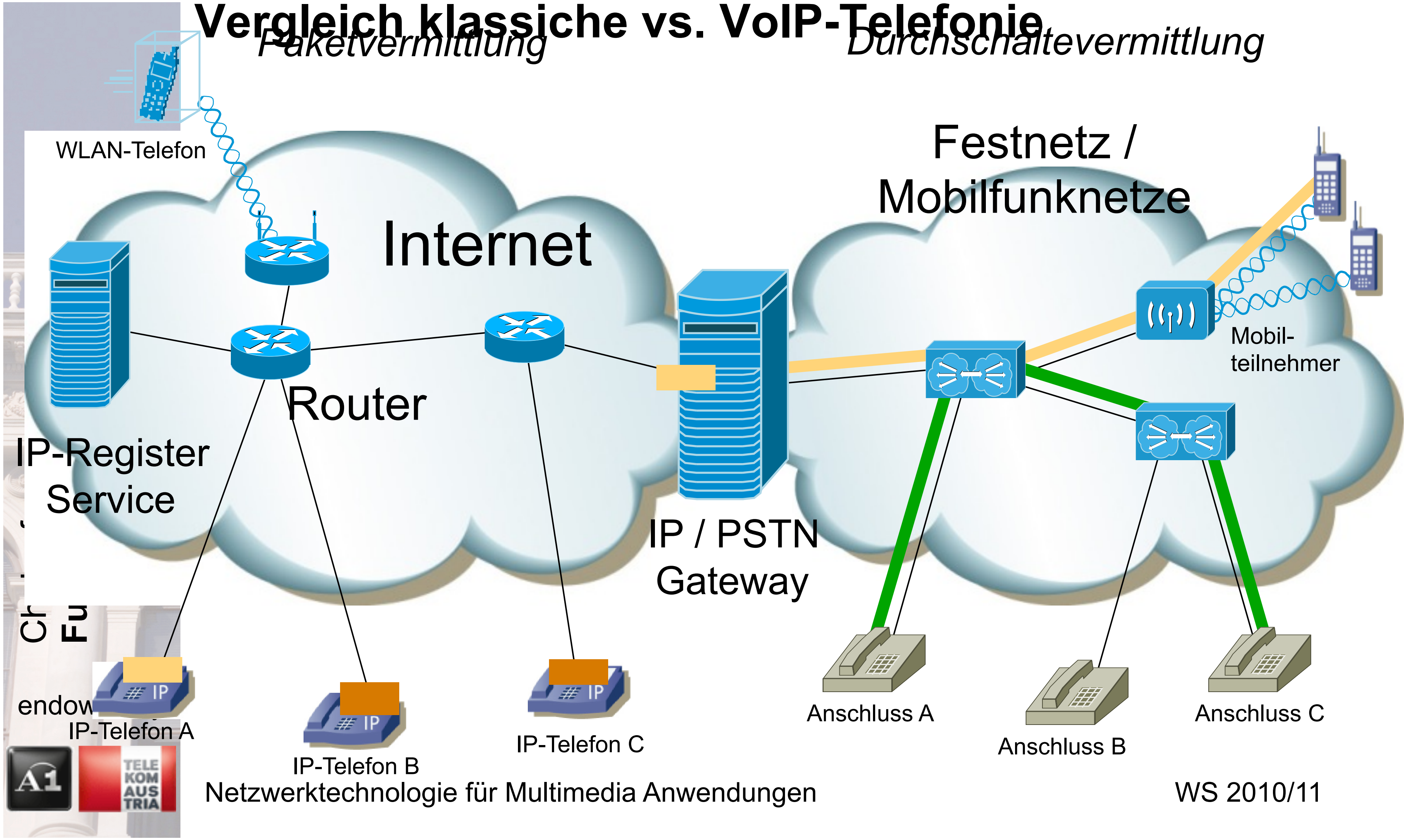
Chair of **Future Communication**

endowed by

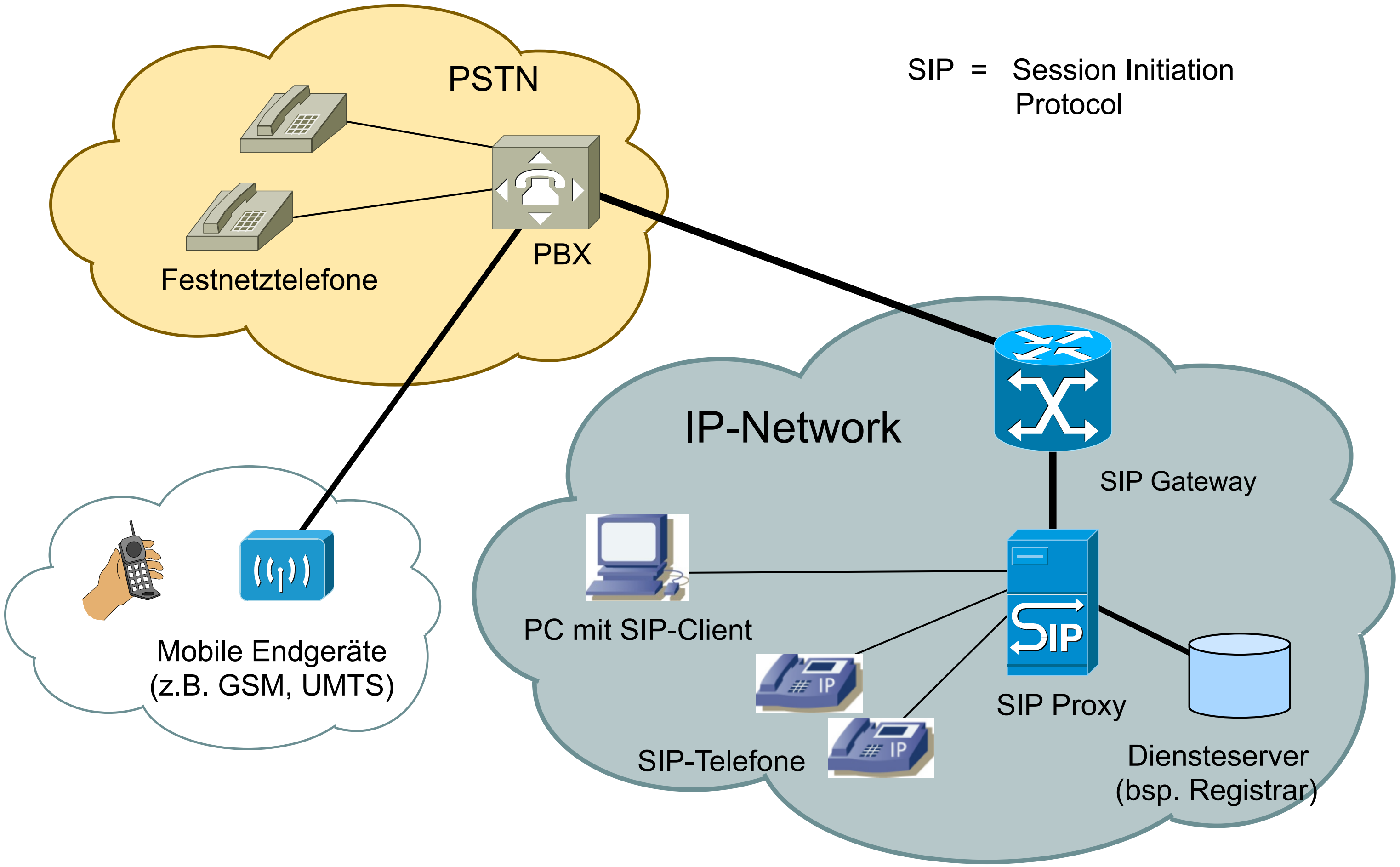
Voice over IP - Umgebung



Vergleich klassiche vs. VoIP-Telefonie



Aufbau eines SIP-basierten Netzes



Erreichbarkeitsproblematik

- **In PSTN-Netzen ist Erreichbarkeit über eindeutige Telefonnummer gewährleistet**

- **Bei VoIP eine Vielzahl an unterschiedlichen Erreichbarkeits-merkmalen:**
  - IP-Adresse
  - SIP-String / VoIP-Nummer
  - Festnetznummer (Wohnortsbereich, RegTP Nummerngasse (032), anbieterspezifisch (z.B. sipgate 01801)

- **Aufgrund der nicht garantierten Ausfallssicherheit und Zuordnung auch keine Erreichbarkeit gewährleistet (*Notrufnummernproblematik*)**

Protokolle für VoIP: H.323 vs. SIP

- **Session Initiation Protocol (SIP)**
  - Universales Initialisierungsprotokoll für den Verbindungsaufbau
  - Nicht festgelegt auf ein spezielles Medium (Video, Audio)
  - Sehr einfaches Protokoll, dadurch leicht in Hardware implementierbar
  - IETF - Standard
- **H.323**
  - Umfassender Standard für Multimediaübertragung, dadurch kompliziert zu implementieren
  - Spezialisiert auf Sprach- und Multimediadienste
  - Objektorientiert, basiert auf QSIG - Standard

Der praktische Einsatz

- **Bestehende Anbieter für SIP-Telefonie**

WEB.DE

freenet.de
normal ist das nicht!

nikotel©

1&1

sipgate

- **Serverlösungen für SIP-Telefonie**

Asterisk

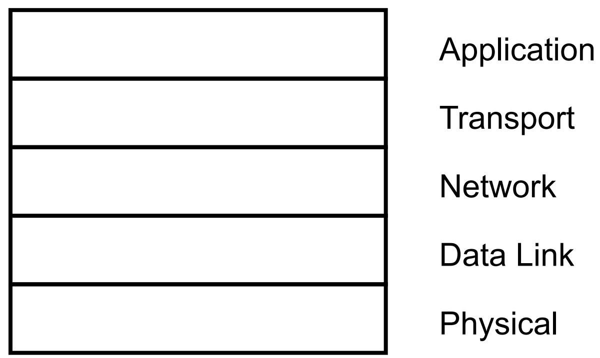
CISCO SYSTEMS

Cisco CallManager

Session Initiation Protocol (SIP)

- **Multiparty Multimedia Session Control (MMUSIC) working group of the Internet Engineering Task Force (IETF)**
  - RFC 2543 (1999)
  - RFC3261 (updated)
- **SIP long-term vision**
  - All telephone calls and video conference calls take place over the Internet
  - People are identified by names or e-mail addresses, rather than by phone numbers.
  - You can reach the callee, no matter where the callee roams, no matter what IP device the callee is currently using.

- **Works according to the client – server principle**



SIP Services

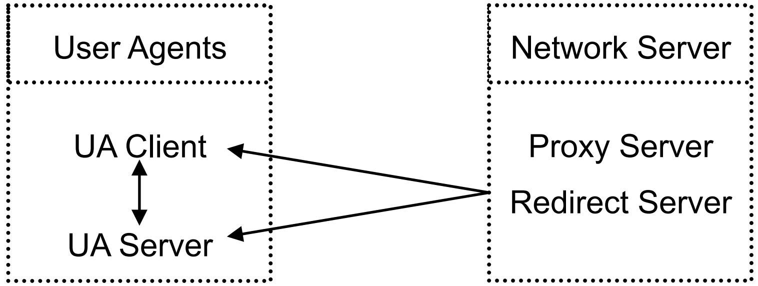
- **Determine current IP address of callee.**
  - Provides mnemonic sip addresses [sip:]<user>@(<host>|<domain>)
    - me@132.187.10.51
    - sip:0123-45-67-89@telefon.com
    - Additional parameters e.g. for transport protocol <URI>;tag1;tag2;…
  - Maps them to current IP addresses
- **Setting up a call**
  - Provides mechanisms for caller to let callee know she wants to establish a call
  - Provides mechanisms so that caller and callee can agree on media type and encoding.
  - Provides mechanisms to end call
- **Call management**
  - Add new media streams during call
  - Change encoding during call
  - Invite others (multi-party conference)
  - Transfer and hold calls

Chair of **Future Communication**

endowed by

SIP Elements

- **User agents**
  - Want to communicate with each other
  - Examples
    - Application on a user‘s computer
    - Cellular phone
    - PSTN gateway
- **SIP proxies or SIP redirect servers**
  - Help to find other users
- **Registrars**
  - Map mnemonic part of sip addresses to IP numbers
  - Usually collocated with SIP server
- **SIP gateways for interoperability with PSTN**

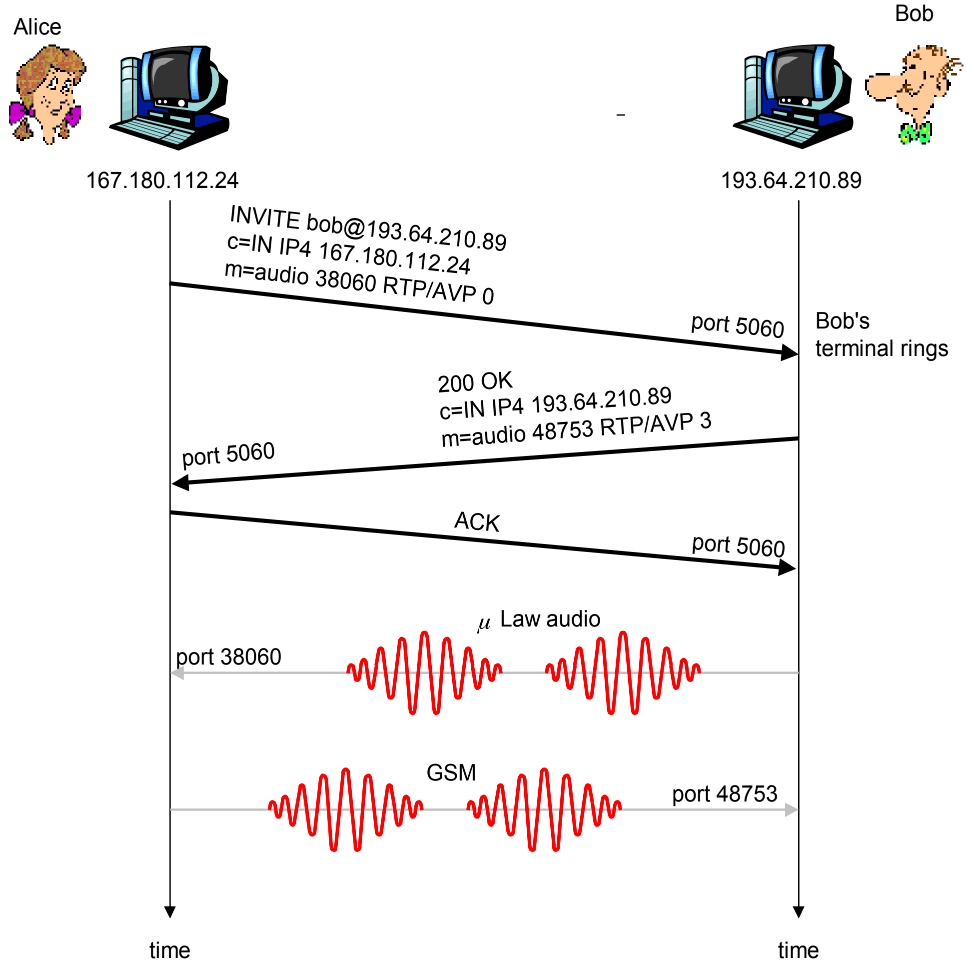


universität wien

SIP Messages

- **Client side methods**
  - INVITE: request for session setup
  - ACK: acknowledgement of a response
  - CANCEL: cancels requests
  - BYE: terminates connection
  - OPTIONS: checks capabilitties of peer
  - REGISTER: registers at SIP registrar
- **Server side status codes (6 categories, similar for other protocols)**
  - 1xx: information about progress of transaction
    - E.g. 180 „Ringing“, 181 „Call is Beeing Forwarded“, 182 „Queued“
  - 2xx: success of transaction (200 „OK“)
  - 3xx: deviation, more effort required
    - E.g. 301 „Moved Permanently“, 302 „Moved Temporarily“
  - 4xx: flaw in request
    - E.g. 404 „Not Found“, 420 „Bad Extension“, 486 „Busy Here“
  - 5xx: flaw at server side
    - E.g. 500 „Internal Server Error“, 504 „Server Time Out“
  - 6xx: general fault
    - E.g. 600 „Busy Everywhere“, 603 „Decline“, 604 „Does Not Exist Anywhere“

Setting up a call to a known IP address



- **Alice's SIP invite message indicates her port number & IP address. Indicates encoding that Alice prefers to receive (PCM ulaw)**
- **Bob's 200 OK message indicates his port number, IP address & preferred encoding (GSM)**
- **SIP messages can be sent over TCP or UDP; here sent over RTP/UDP.**
- **Default SIP port number is 5060.**

SIP Registrar

- **When Bob starts SIP client, client sends SIP REGISTER message to Bob's registrar server**

(similar function needed by Instant Messaging)

Register Message:

```
REGISTER sip:domain.com SIP/2.0
Via: SIP/2.0/UDP 193.64.210.89
From: sip:bob@domain.com
To: sip:bob@domain.com
Expires: 3600
```

Name Translation and User Location

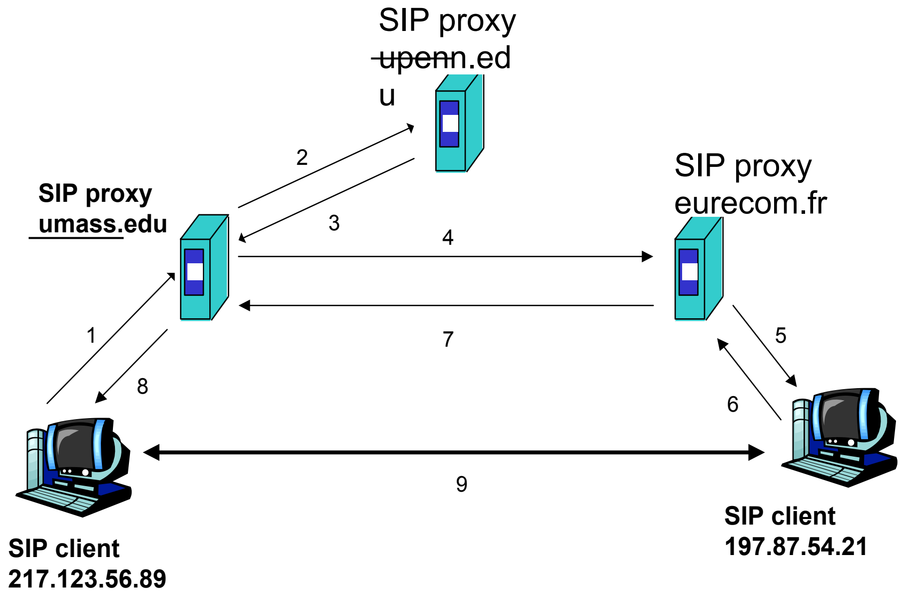
- **Caller wants to call callee, but only has callee's name or e-mail address.**
- **Need to get IP address of callee's current host:**
  - User moves around
  - DHCP protocol
  - User has different IP devices (PC, PDA, car device)
- **Caller asks SIP server (like DNS server)**
- **Result can be based on:**
  - Time of day (work, home)
  - Caller (don't want boss to call you at home)
  - Status of callee (calls sent to voicemail when callee is already talking to someone)

- **Alice sends invite message to her proxy server containing Bob's address "sip:bob@domain.com"**
- **Proxy responsible for routing SIP messages to callee possibly through multiple proxies.**
- **Callee sends response back through the same set of proxies.**
- **Proxy returns SIP response message to Alice containing Bob's IP address**

Example

Caller jim@umass.edu places a call to keith@upenn.edu

- **(1) Jim sends INVITE message to umass SIP proxy.**
- **(2) Proxy forwards request to upenn proxy/registrar server.**
- **(3) upenn server returns redirect response, indicating that it should try keith@eurecom.fr**



- **(4) umass proxy sends INVITE to eurecom SIP proxy/registrar.**
- **(5) eurecom proxy/registrar forwards INVITE to 197.87.54.21, which is running keith's SIP client.**
- **(6-8) SIP response sent back**
- **(9) media sent directly between clients.**
- **Note: also a SIP ack message, which is not shown.**

Setting up a call (more)

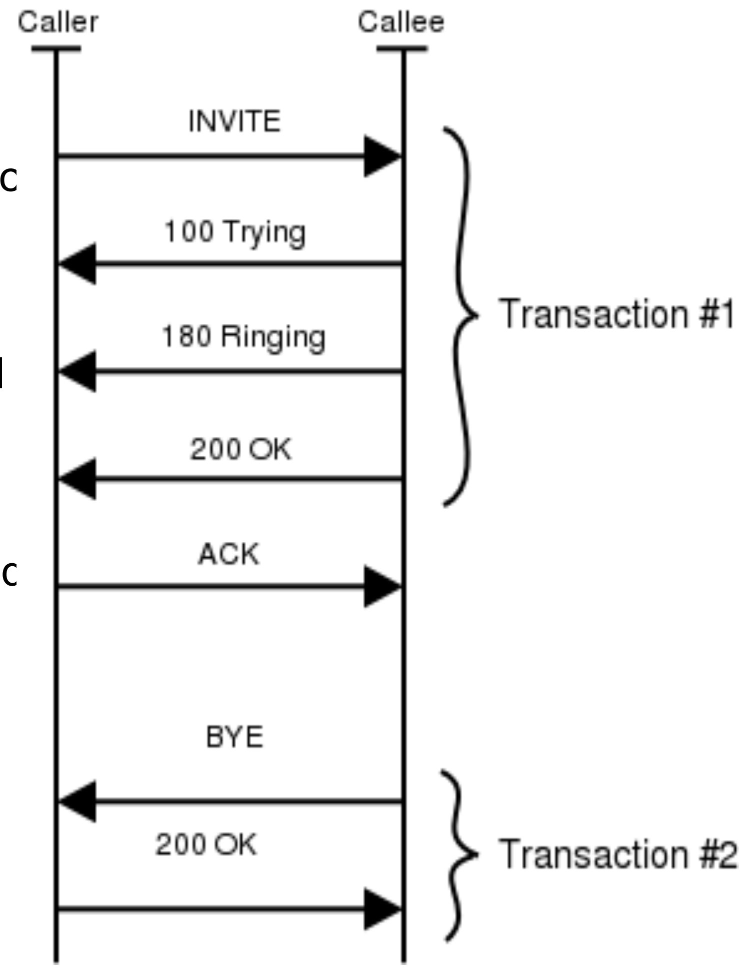
- **Codec negotiation:**
  - Suppose Bob doesn't have PCM ulaw encoder.
  - Bob will instead reply with 606 Not Acceptable Reply and list encoders he can use.
  - Alice can then send a new INVITE message, advertising an appropriate encoder.
- **Rejecting the call**
  - Bob can reject with replies "busy," "gone," "payment required," "forbidden".
- **Media can be sent over RTP or some other protocol.**

Chair of **Future Communication**

endowed by

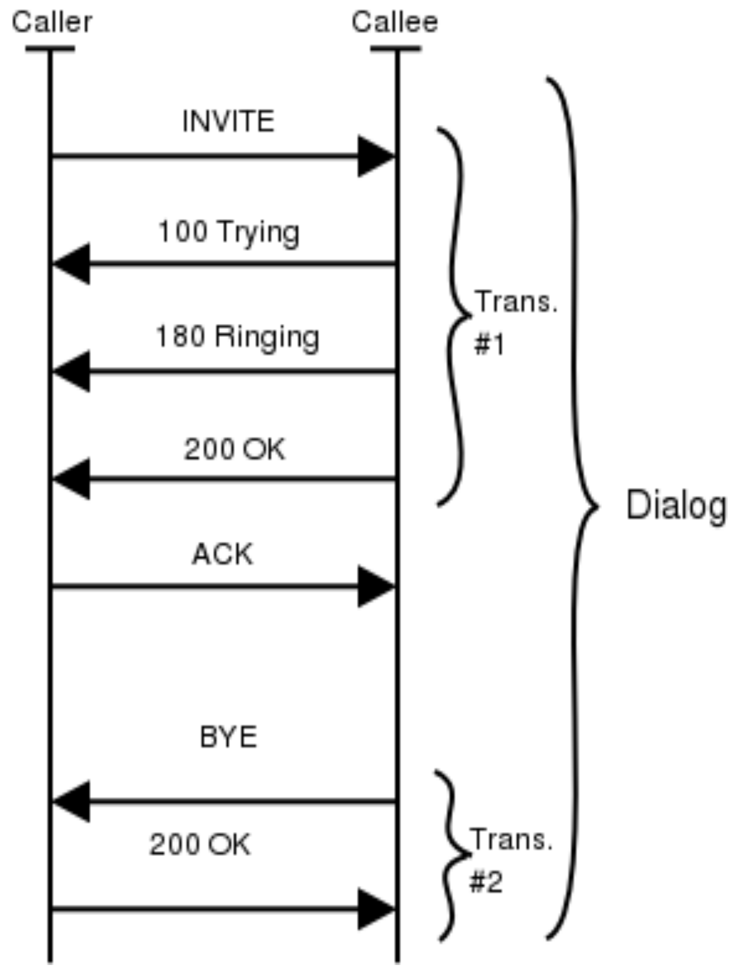
SIP Transactions

- **Transaction**
  - Sequence of SIP messages
  - One request and all responses tc that request
  - Zero or more provisional responses and one or more final responses (e.g. when proxy server forks)
  - Final positive ack may be omittec but not final negative ack
  - Transaction identifier
    - hash of all important message header fields (obsolete)
    - Directly contained in msg.



SIP Dialogs

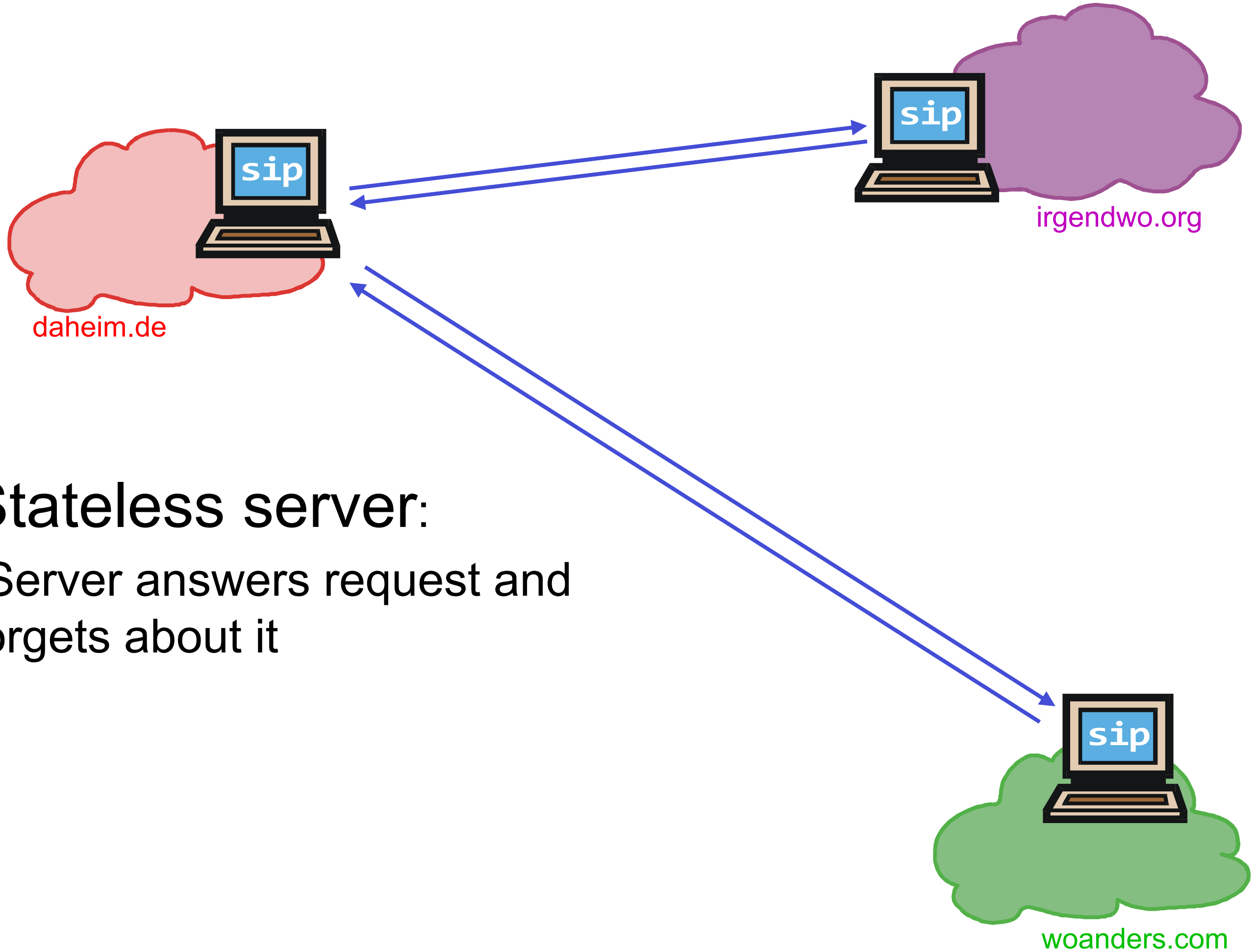
- **Represents a peer-to-peer SIP relationship between two user agents**
- ***A dialog is a sequence of transactions***
- **Dialogs are identified using Call-ID, From tag, and To tag**
- **CSeq header field numbers request / messages within a dialog.**
- **The number must be monotonically increased for each message sent within a dialog otherwise the peer will handle it as out of order request or retransmission**



Chair of **Future Communication**

endowed by

Network Server (1): Redirect Server



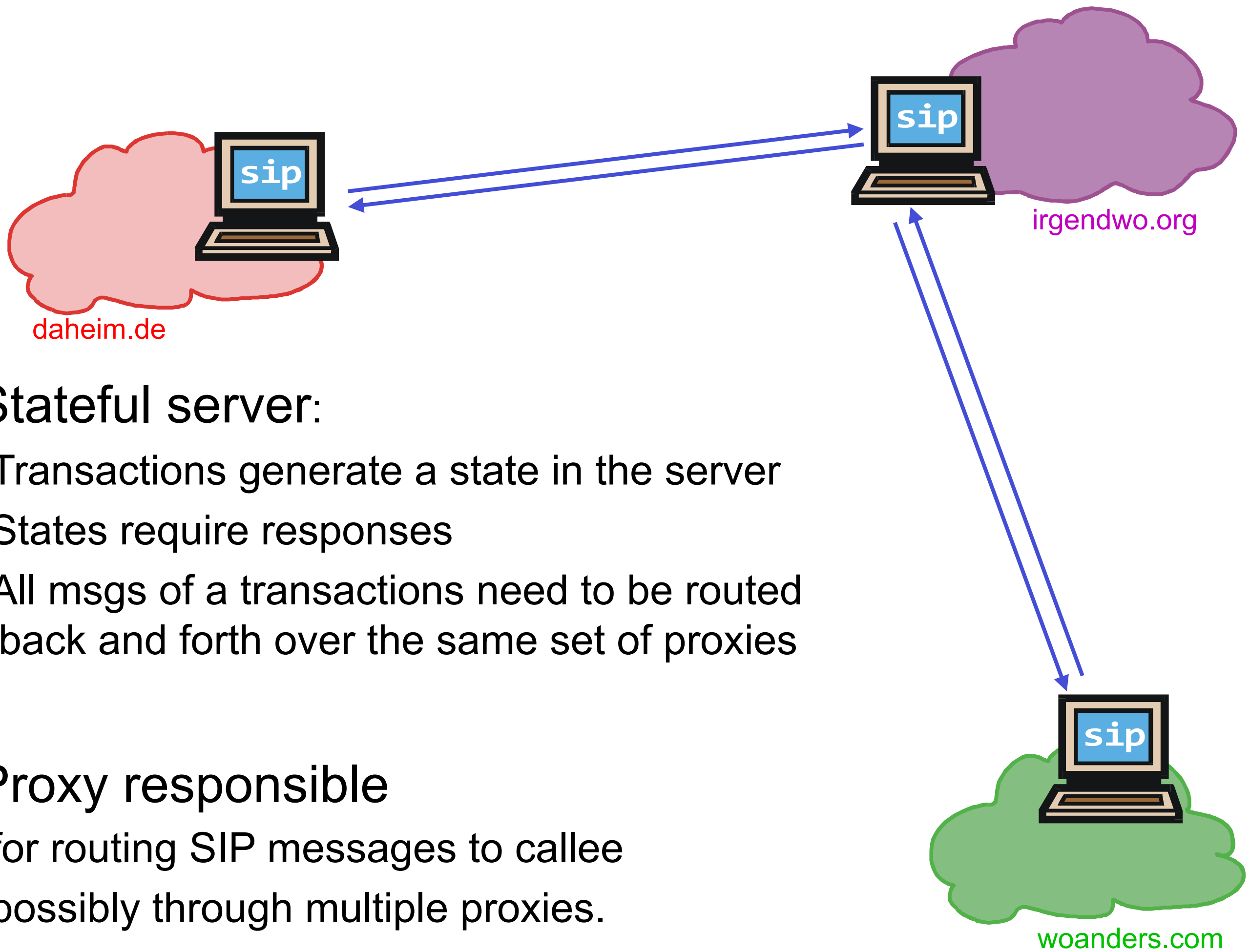
Stateless server:

•Server answers request and forgets about it

Chair of **Future Communication**

endowed by

Network Server (2): Proxy Server



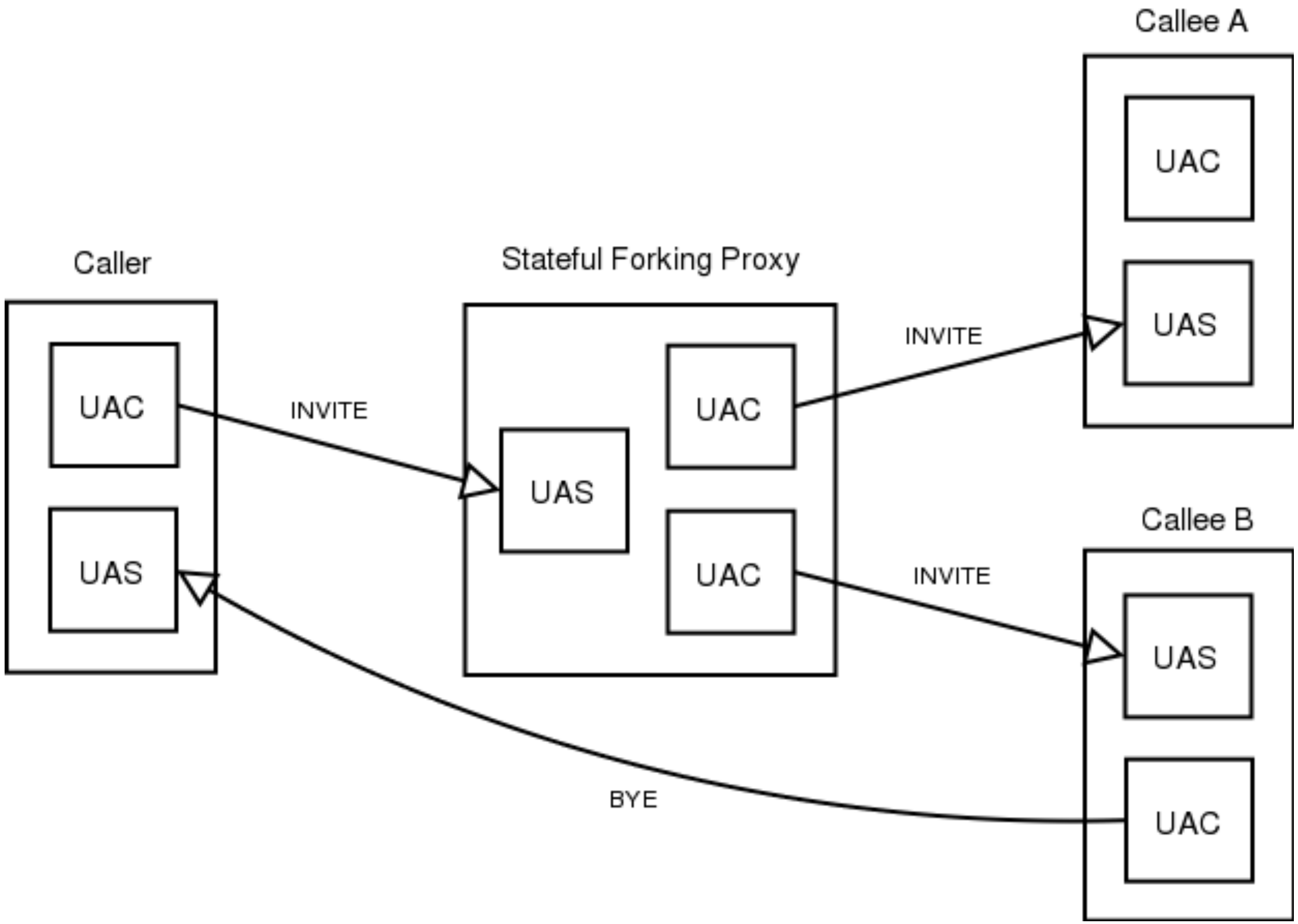
Stateful server:

- Transactions generate a state in the server
- States require responses
- All msgs of a transactions need to be routed back and forth over the same set of proxies

Proxy responsible

- for routing SIP messages to callee
- possibly through multiple proxies.

Forking Proxy

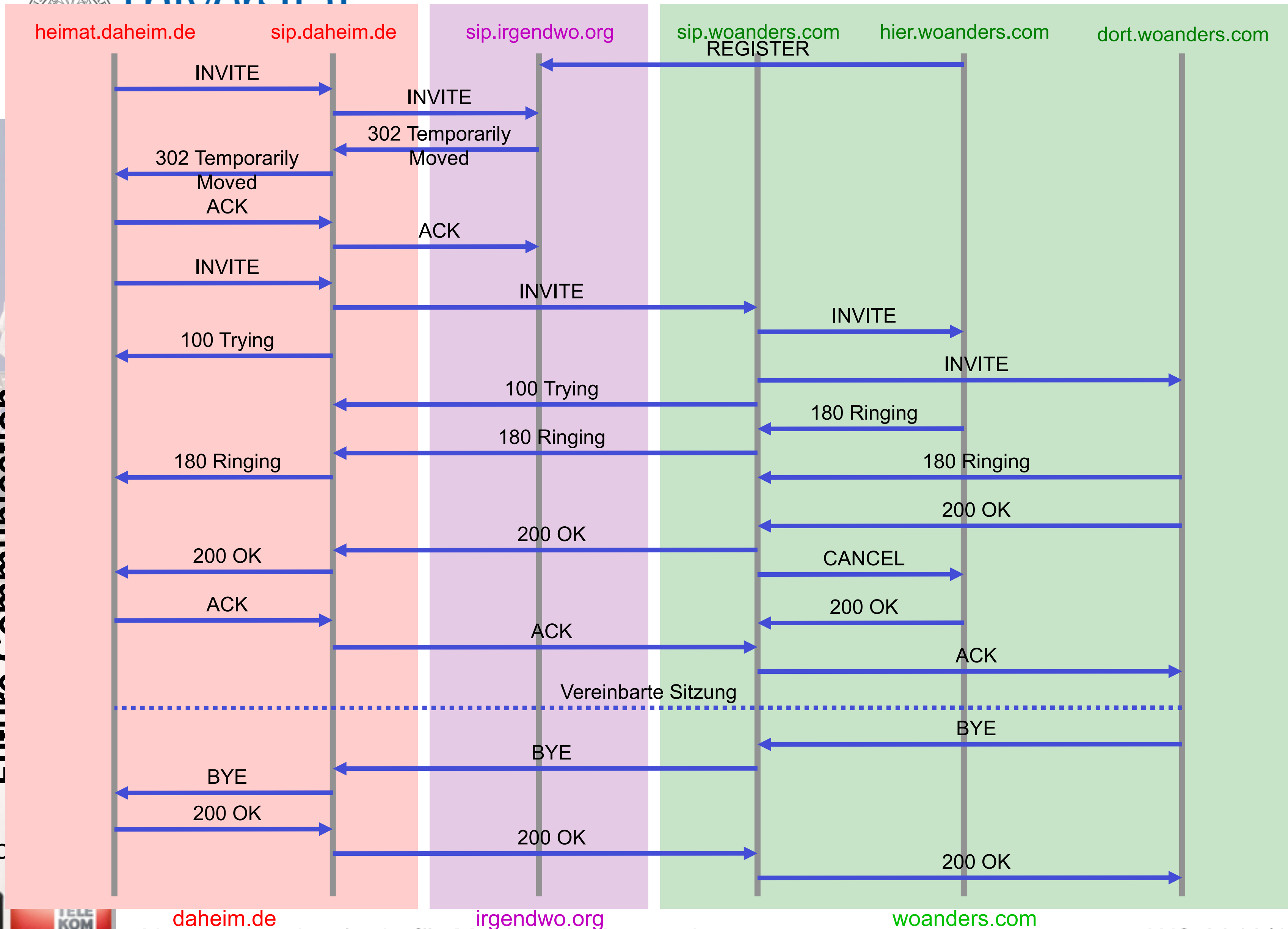


Proxy Types

- **Stateless proxy**
  - Simple message forwarder
  - Do not take care of transactions
  - Simpler and faster than stateful proxies
- **Stateful proxy**
  - Most of today's proxies are stateful
  - Creates a state upon reception of a request and keeps it until the transaction finishes
  - Advanced functionalities
    - Absorb retransmissions
    - Advanced message routing
      - Forking (msg forwarding to different destinations)
      - Recursive traversal
        (try different locations to find a user)
    - Accounting
    - NAT traversal aid

How do we find appropriate SIP proxy?

- **Local SIP server usually knows users within its domain**
- **How does it find appropriate SIP proxy?**
- **Remember: Domain Name System!**
  - Translates mnemonic addresses into IP numbers
  - Hierarchical structure
  - Resource records specify request (Name, Value, Type, TTL)
  - Type may be, e.g., A, NS, CNAME, MX, …
  - MX returns mail exchange server
  - Similar type for SIP server (SRV?)


heimat.daheim.de
sip.daheim.de
sip.irgendwo.org
sip.woanders.com
hier.woanders.com
dort.woanders.com
REGISTER
INVITE
INVITE
302 Temporarily Moved
302 Temporarily Moved
ACK
ACK
INVITE
INVITE
INVITE
100 Trying
INVITE
100 Trying
180 Ringing
180 Ringing
180 Ringing
180 Ringing
200 OK
200 OK
200 OK
CANCEL
200 OK
ACK
ACK
ACK
Vereinbarte Sitzung
BYE
BYE
BYE
200 OK
200 OK
200 OK
daheim.de
irgendwo.org
woanders.com

Example of SIP message

```
INVITE sip:bob@domain.com SIP/2.0
Via: SIP/2.0/UDP 167.180.112.24
From: sip:alice@hereway.com
To: sip:bob@domain.com
Call-ID: a2e3a@pigeon.hereway.com
Content-Type: application/sdp
Content-Length: 885

c=IN IP4 167.180.112.24
m=audio 38060 RTP/AVP 0
```

- **Here we don't know Bob's IP address. Intermediate SIP servers will be necessary.**
- **Alice specifies in Via: header that SIP client sends and receives SIP messages over UDP**
- **Alice sends and receives SIP messages using the SIP default port number 5060.**

- **HTTP-like message syntax**
- **Format: Unicode-Text in UTF-8-Codierung (8-bit unicode transformation format, RFC3629)**
- **sdp = session description protocol**
- **Call-ID is unique for every call.**

SDP - Session Description Protocol

- **Von MMUSIC / IETF, veröffentlicht als RFC 2327 (April 1998)**

- **Textformat: <Bezeichner>=<Wert>**

- **Informationen:**
  - Name und Zweck der Session
  - Zeit(en), in der (denen) die Session aktiv ist
  - Medien, die an der Session beteiligt sind
  - Informationen zum Empfang dieser Medien (Adresse, Port, Format, …)

  zusätzlich:
  - Bandbreite
  - Ansprechpartner

SIP Message Structure

Chair of **Future Communication**

endowed by

- *Transaktion* besteht aus *Request - Response (- Acknowledgement)*

- Aufbau:

<Start-Zeile>
<Header_1>:<Wert_1>
... Kopf
<Header_n>:<Wert_n>

< ... > Körper

- <Start-Zeile> = <Methode> <Request-URI> <SIP-Version> Request
  | <SIP-Version> <Status-Code> <Begründung> Response

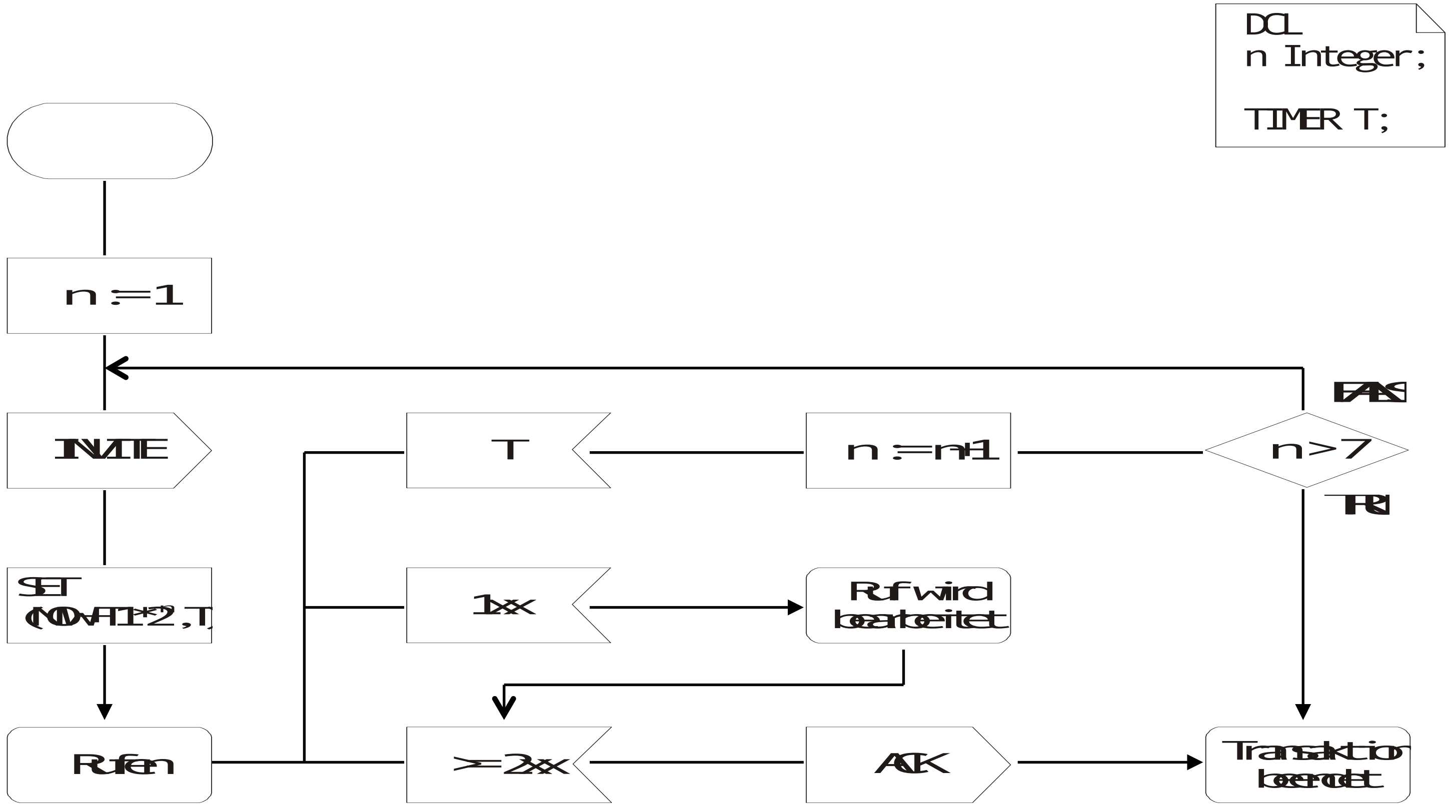
SIP Header Types

- **SIP erweiterbar um neue Header durch hierarchische Namensgebung oder Registrierung bei der Internet Assigned Numbers Authority (IANA)**

- **Wichtige Standard-Header:**
  - From: der anrufende Partner
  - To: der gerufene Partner
  - Call-ID: eindeutiger Bezeichner der Sitzung
  - CSeq: Bezeichner für eine Transaktion (Nr + Methode)
  - Via: Route der Nachricht (bisher)
  - Contact: Alternativ-Adresse(n)
  - Require: Erforderliche Optionen
  - Unsupported: Nicht unterstützte Optionen
  - Content-Type/-Length/-Encoding: Nachrichten-Körper

SDL Diagram for Client Finite State Machine

Aus Sicht des Client, Server-Sicht analog!



Operations for Other Methods

- **Nachrichtenfluss bei den übrigen Methoden:**
  - Request/Response bei BYE, CANCEL, OPTIONS, REGISTER analog
  - Retransmission des Request ebenfalls mit Back-Off-Algorithmus
  - Keine Bestätigung mit ACK
  - ⇨ keine period. Wiederholung des Response
  - Wiederholung des Response nur nach Wiederholung des Request

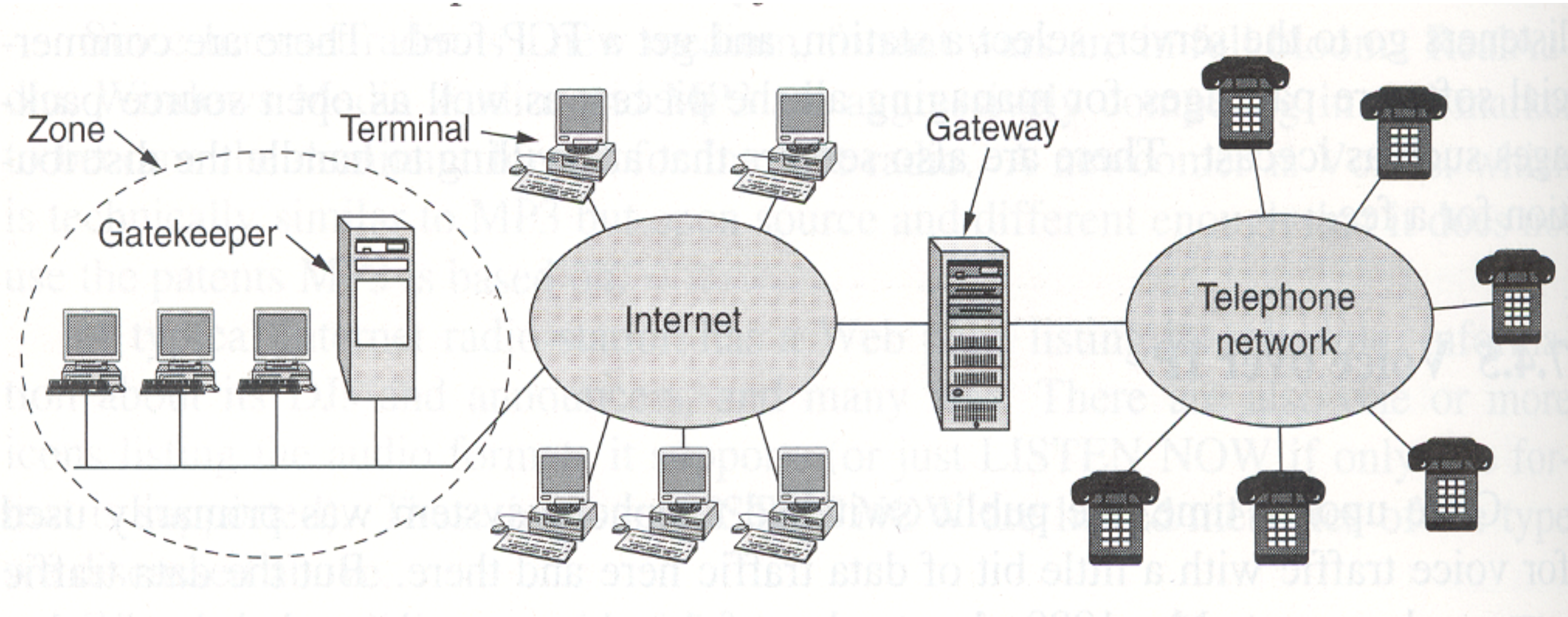
- ***Bemerkung:***

  Wiederholungen bei TCP i.d.R. unnötig, da zuverlässige Verbindung

H.323 Overview

- **Multimedia communication standard by ITU**
  - Development start in 1996 to make IP telephony equipment from different vendors interoperable
  - Revised in 1998
  - Basis for first widespread Internet telephony systems
  - Not a single protocol but an architectural overview comprising many different protocols and other stuff
    - Speech codecs
    - Call setup
    - Signalling
    - Data transport
    - Interoperability with telephone network
  - Network elements
    - Terminals
    - Gatekeper
    - Zone
    - Gateway

H.323 Overview



Chair of **Future Communication**

endowed by

H.323 Protocols

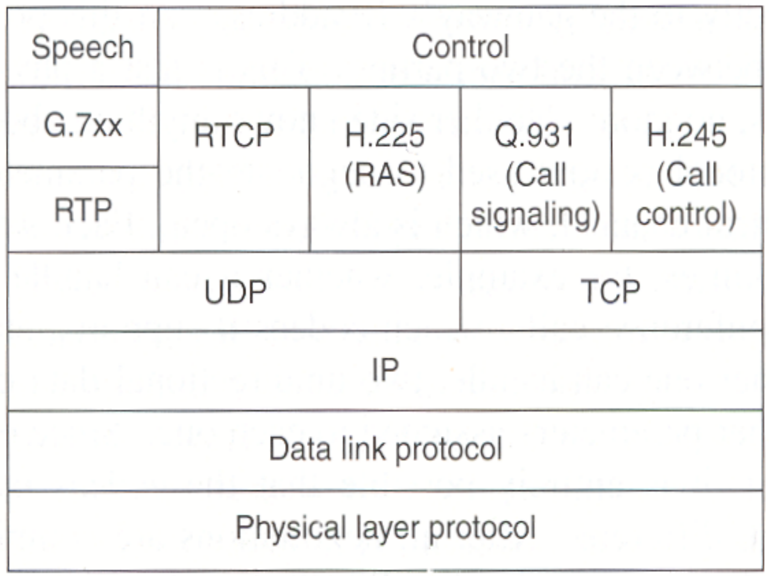
- **Codecs**
  - Requirement: G.711 (64 kbps PCM voice, uncompressed)
  - Many others permitted, e.g., G.723.1 (predictive coding to compress speech to 24 or 20 bytes / 30 ms = 6.4 and 5.3 kbps, compression factor 10 and 12!)
- **H.245**
  - Capability information and parameter negotiation
  - Codecs, bit rates, …
- **RTP and RTCP required for data transport**
- **ITU Q.931**
  - Standard telephony signalling
    - Establishing and releasing connections
    - Providing dial tones, making ringing sounds
    - Rest of standard telephony features

H.323 Protocols

Chair of **Future Communication**

endowed by

- **H.225**
  - Used for communication with gatekeeper
  - Manages PC-to-gatekeeper channel „RAS“ (Registration/ Admission/Status), allows terminals
    - To join and leave the zone
    - To request and return bandwidth
    - To provide status updates and other stuff

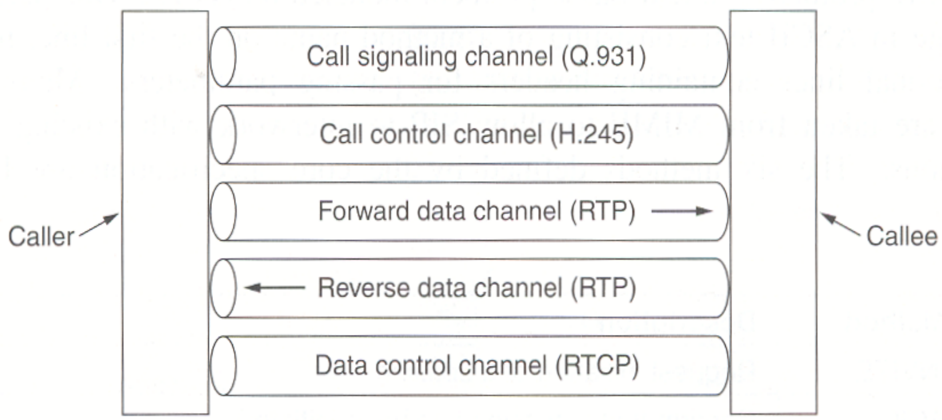


H.323 Signaling Example

- **PC discovers gatekeeper by broadcasting a UDP gatekeeper discovery packet to port 1718**
- **Gatekeeper responds and PC learns gatekeeper‘s IP address**
- **PC registers at gatekeeper with a UDP msg**
- **After acceptance, PC requests bandwidth with a RAS msg in UDP**
  - QoS achieved by local admission control
- **If successful, connection setup may start over TCP towards gatekeeper**
  - PC sends Q.931 SETUP (including telephone number / IP address)
  - Gatekeeper responds with Q.931 CALL PROCEEDING and forwards SETUP towards gateway
  - Gateway is half computer and half telephone switch
  - Forwards SETUP in appropriate way and forwards Q.931 ALERT to PC (ringing has begun)
  - If remote peer picks up the telephone, a CONNECT msg is sent back to the PC

H.323 Signaling Example

- **From now on, the gatekeeper is no longer in the loop and communication is directly PC – Gateway**
  - H.245: capability and parameter (e.g. codec) negotiation
  - Different codecs may be used for both directions
  - Two unidirectional data channels are set up
  - Data flow begins over RTP
  - RTCP helps to control congestion and to synchronize audio and video
- **Q.931 channel is finally used to tear down the connection**
- **When call is terminated, PC contacts gatekeeper to realease the reserved bandwidth**



Comparison SIP vs. H.323

- **SIP is a single component. Works with RTP, but does not mandate it. Can be combined with other protocols and services.**
- **SIP comes from IETF: Borrows much of its concepts from HTTP.**
- **SIP has a Web flavor.**
  - States kept in end devices
  - Proxy just helps but it is not required
- **SIP uses the KISS principle: Keep it simple stupid.**

- **H.323 is a complete, vertically integrated suite of protocols for multimedia conferencing: signaling, registration, admission control, transport and codecs.**
- **H.323 comes from the ITU (telephony).**
- **H.323 has a telephony flavor.**
  - End devices (telephones) are very primitive
  - States kept in network devices in PSTN; H.323 uses also a gatekeeper

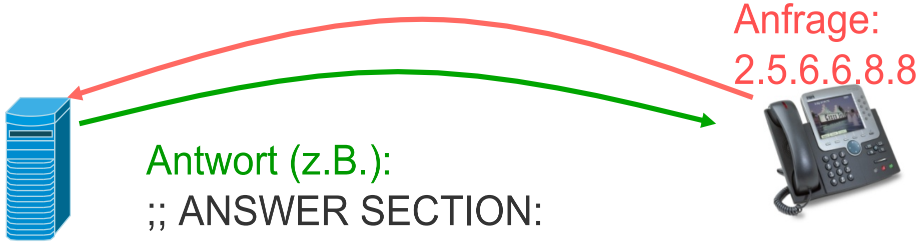
Zentraler Rufnummerndienst mit ENUM

- tElephone NUmber Mapping (ENUM): Anwendung des DNS zur Übersetzung von Telefonnummern in Internet-Adressen (RFC3761)
- Klassische Telefonnummer durch ITU E.164 festgelegt
- ENUM Umsetzung:
  - Eingabe: E.164 konforme Nummer
  - Ausgabe: gültige „absoluteURI“ (RFC2396)
  - Beispiel:

+49-931-888-6652 → 499318886652 → 2.5.6.6.8.8.8.1.3.9.9.4.e164.arpa

Gültige Telefonnummer — Nur Zahlen —
• Punkte zwischen jede Zahl
• Reihenfolge umkehren
• Nummern Identifizierung anhängen

- ENUM Funktionsweise:
  - NAPTR (Naming Authority Pointer) im Resource Record des DNS
  - IN: Internet



Anfrage:
2.5.6.6.8.8.1.3.9.9.4.e164.arpa

DNS-Server
(bspw. DENIC)

Antwort (z.B.):
;; ANSWER SECTION:
2.5.6.6.8.8.8.1.3.9.9.4.e164.arpa.
IN NAPTR 100 10 "u" "e2u+sip" "!^.*$!sip:test@info3.de!" .
IN NAPTR 100 10 "u" "e2u+http" "!^.*$!http://www.info3.de/staff/test!" .
IN NAPTR 100 10 "u" "e2u+email" "!^.*$!mailto:test@info3.de!" .

Next Generation Network (NGN) – im engeren Sinne

Chair of **Future Communication**

endowed by

- **Ziel: Konvergenz von Telefon- und Datennetzen**
  - paketvermittelte Verbindungen zwischen zwei und mehr Teilnehmern
  - Zusammenarbeit zwischen der leitungsvermittelnden und der paketorientierten Domäne
  - eine End–to–End-Aushandlung der Dienstgüte (Quality of Service)
  - dienstabhängige Kostenabrechnung
  - Bereitstellung der Heimnetzumgebung in Fremdnetzen
  - Unterstützung verschiedener Medientypen
  - schnelle und flexible Erstellung von Diensten durch Service Enabler (vordefinierte Dienstbausteine)
  - Dienste sollen unabhängig vom Zugangsnetz sein
- **Ansätze in der Standardisierung**
  - TC TISPAN (Technical Committee Telecoms & Internet converged Services & Protocols for Advanced Networks) von ETSI (European Telecommunications Standards Institute)
  - IP Multimedia Subsystem (IMS) von 3GPP (3rd Generation Partnership Project, weltweite Kooperation von 5 Normierungsorganisationen zur Standardisierung im Mobilfunk, darunter auch ETSI)
  - Y.2001 der ITU-T (International Telecommunication Union – Stanardization Sector)
  - Mehrheitliche Benutzung von SIP

Quelle: Wikipedia, 29.6.2009

Chapter 3: Multimedia Networking

Chair of
Future Communication
endowed by
A1 TELEKOM AUSTRIA

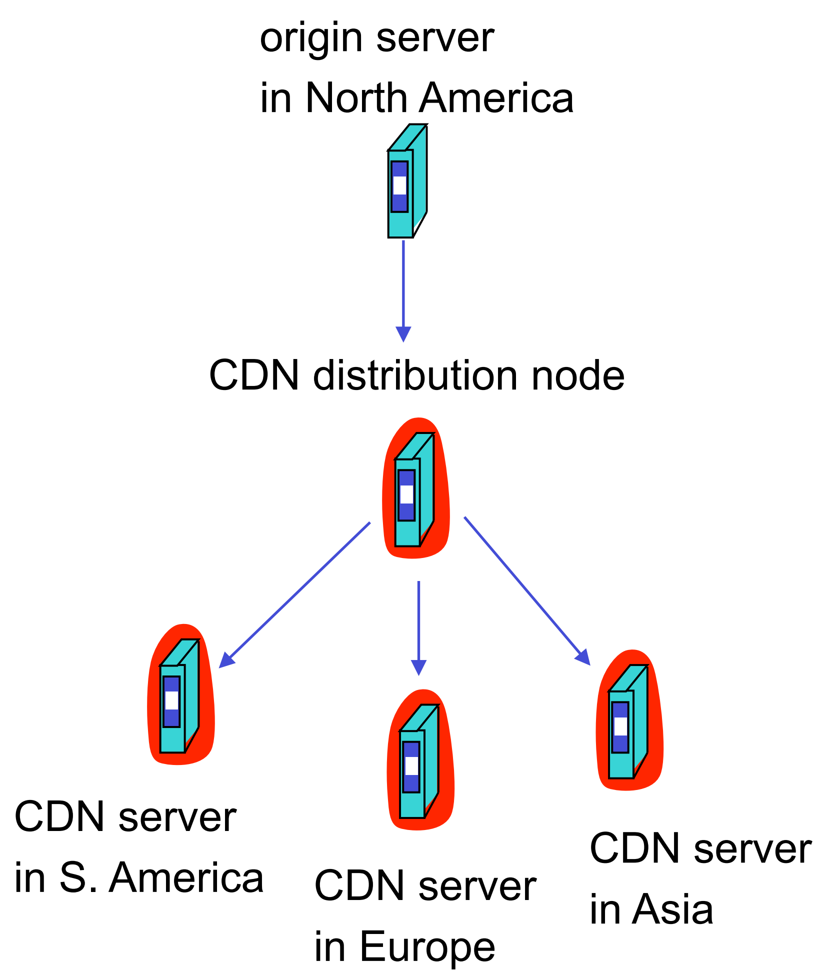
Overview:

- 2.1 Multimedia Networking Applications
- 2.2 Streaming stored audio and video
- 2.3 Real-time Multimedia: Internet Phone study
- 2.4 Protocols for Real-Time Interactive Applications
  - RTP,RTCP
- 2.5 IP Telefony, SIP, and H.323
- 2.6 Distributing Multimedia: content distribution networks

Content distribution networks (CDNs)

Content replication

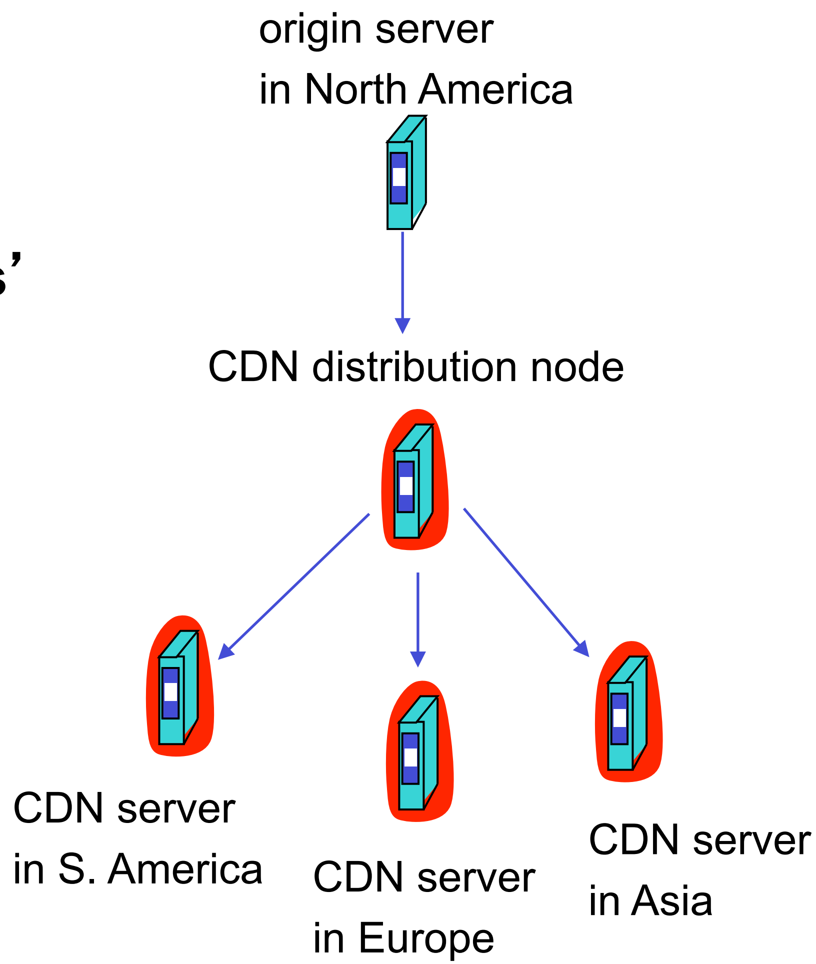
- **Challenging to stream large files (e.g., video) from single origin server in real time**
- **Solution: replicate content at hundreds of servers throughout Internet**
  - content downloaded to CDN servers ahead of time
  - placing content “close” to user avoids impairments (loss, delay) of sending content over long paths
  - CDN server typically in edge/ access network



Content distribution networks (CDNs)

Content replication

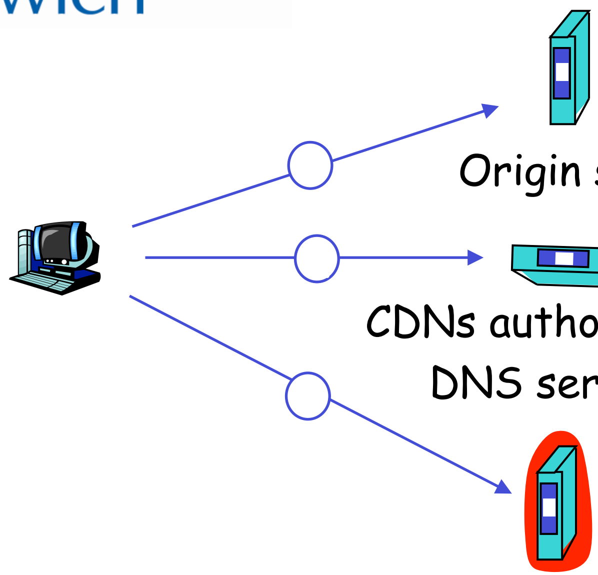
- **CDN (e.g., Akamai) customer is the content provider (e.g., CNN)**
- **CDN replicates customers' content in CDN servers. When provider updates content, CDN updates servers**



Chair of
Future Communication

endowed by

CDN example



origin server (www.foo.com)

- **distributes HTML**
- **replaces:**

  http://www.foo.com/sports/ruth.gif

  with

  http://www.cdn.com/www.foo.com/sports/ruth.gif

CDN company (cdn.com)

- **distributes gif files**
- **uses its authoritative DNS server to route redirect requests**

More about CDNs

routing requests

- **CDN creates a "map", indicating distances from leaf ISPs and CDN nodes**
- **when query arrives at authoritative DNS server:**
  - server determines ISP from which query originates
  - uses "map" to determine best CDN server
- **CDN nodes create application-layer overlay network**