Chair of Future Communication
Faculty of Computer Science
Prof. Dr. K. Tutschku

050069

VO Netzwerktechnologie für Multimedia Anwendungen

Slide set 6: Multimedia Networking

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

Endowed by

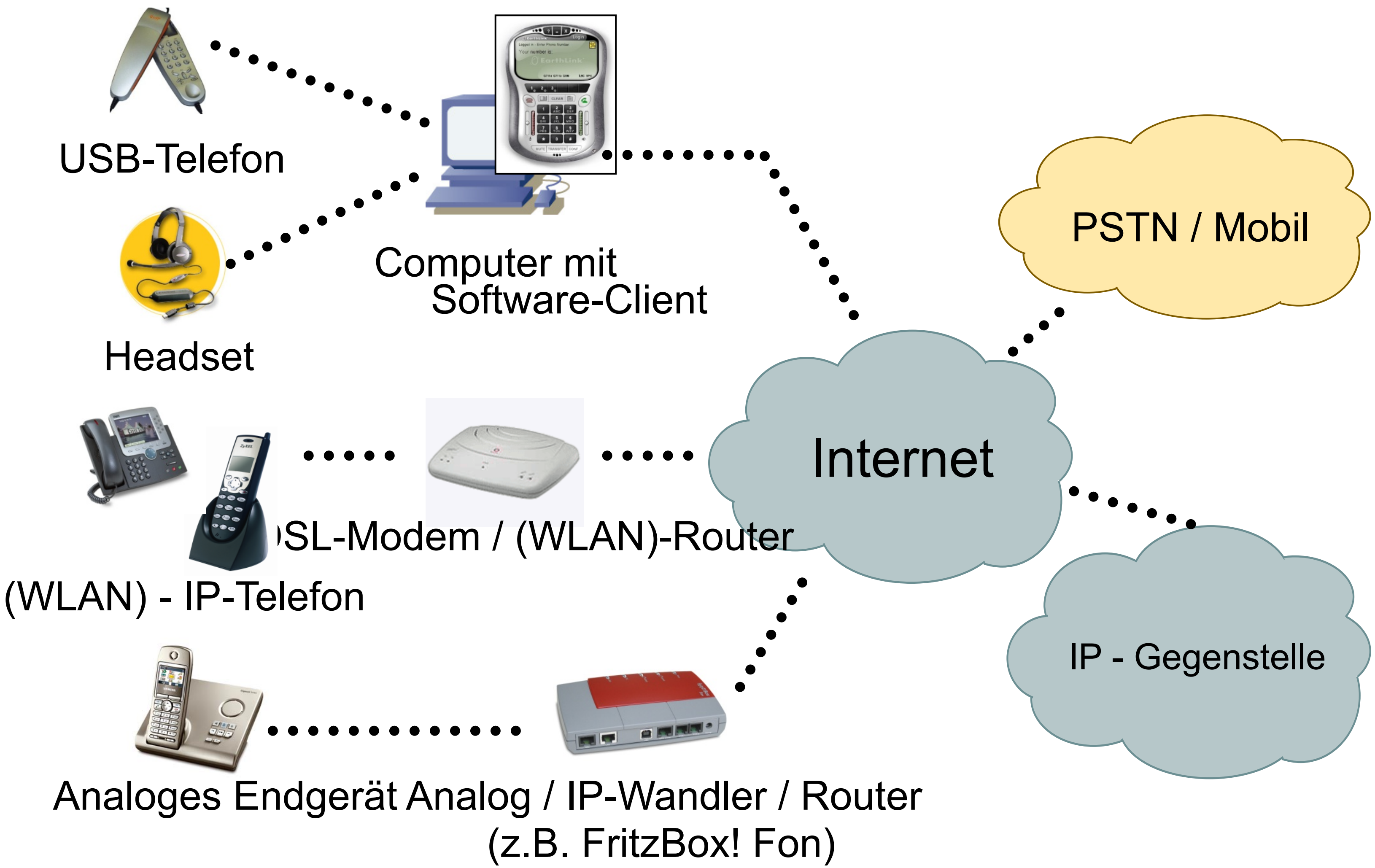
Bachelor Informatik (Medieninformatik)
WS 2011/12

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

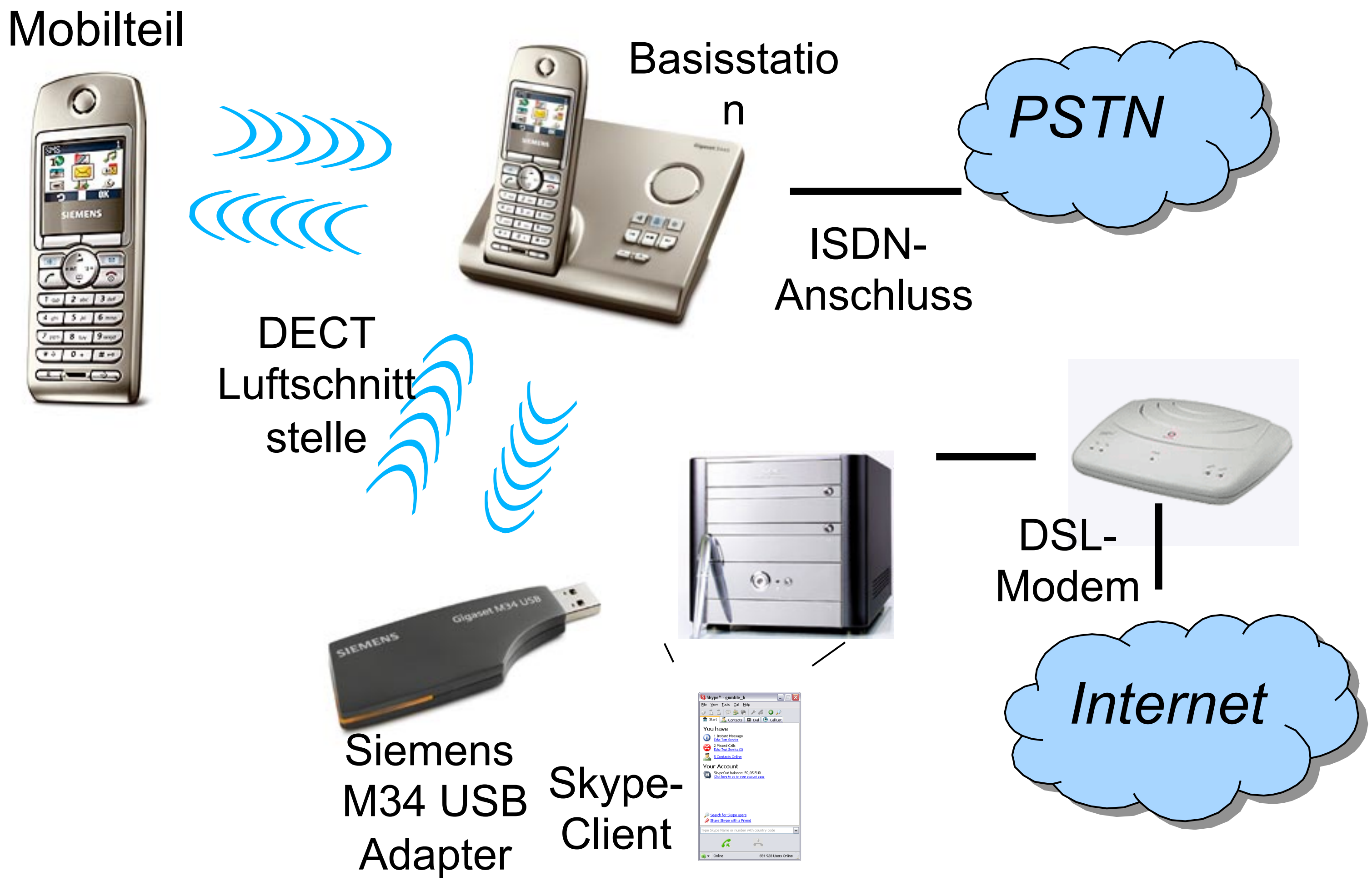
Zugangsmöglichkeiten zu VoIP



Chair of **Future Communication**

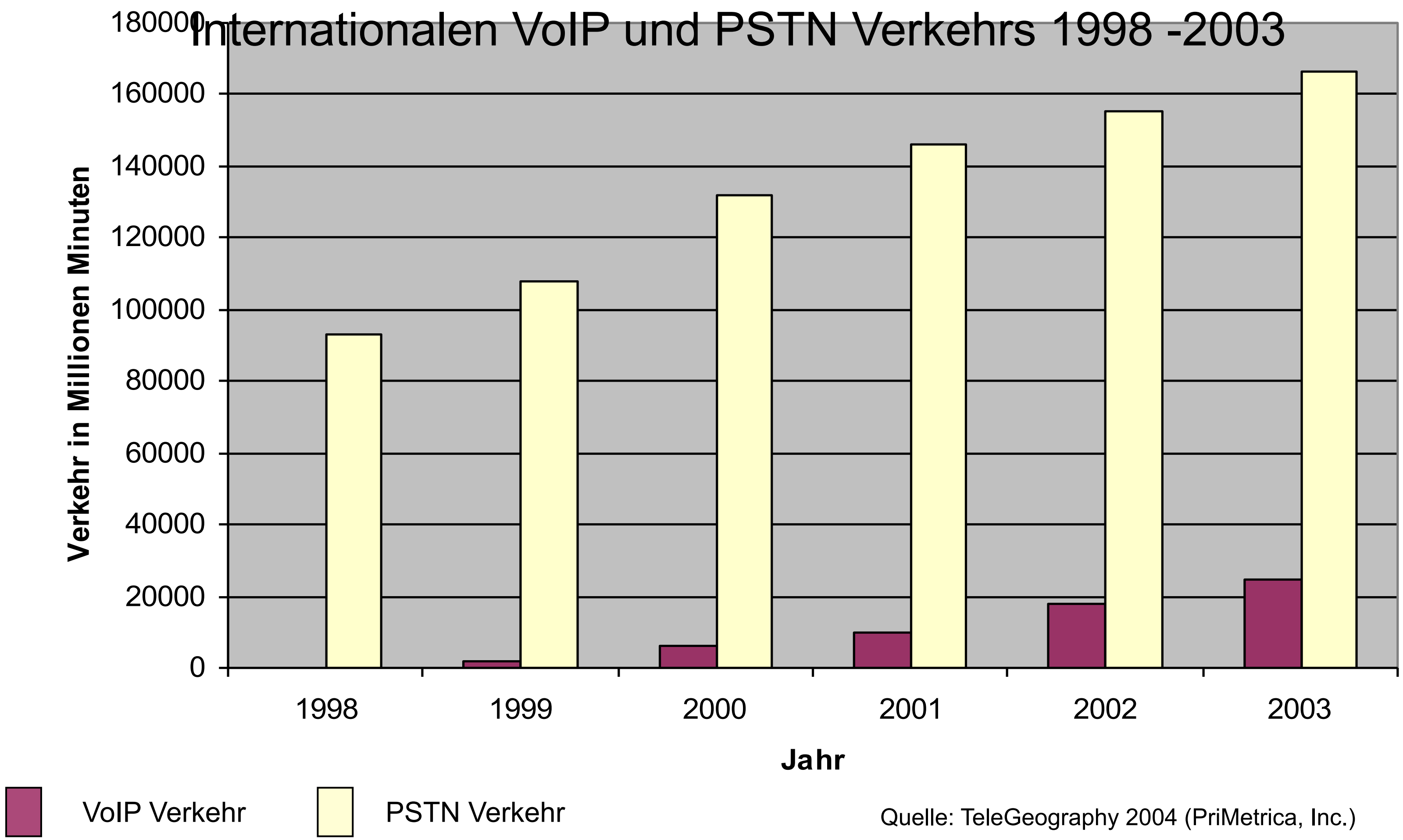
endowed by

Skype / DECT Beispiel-Installation



Chair of
Future Communication
endowed by

VoIP-Entwicklung in Zahlen



Quelle: TeleGeography 2004 (PriMetrica, Inc.)

Chair of **Future Communication**

endowed by

universität wien

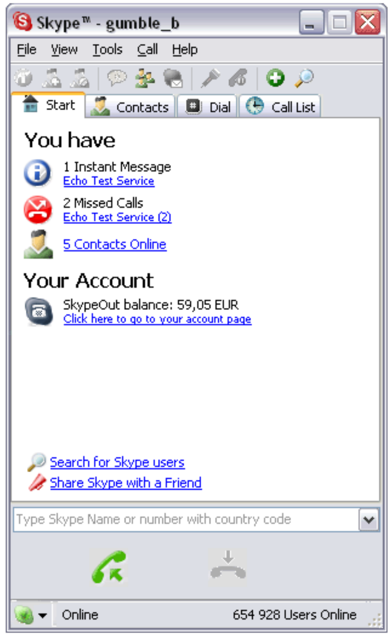
Freie Softwarekomponenten

Chair of **Future Communication**

endowed by A1

- Softwareunterstützung für VoIP

SIP-fähige Clients



Skype

ICQ

Netmeeting

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

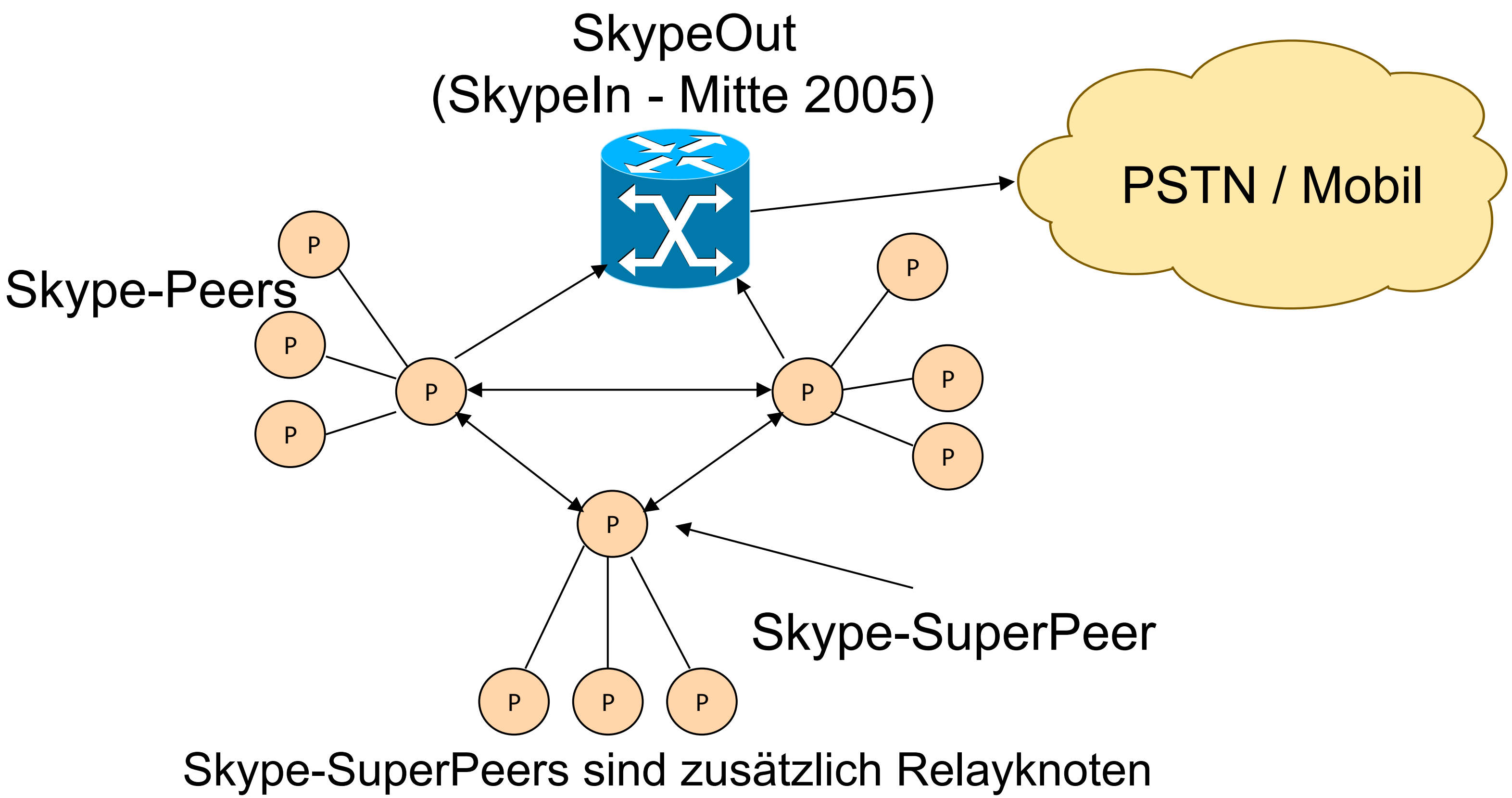
Quelle: pressetext deutschland

Chair of Future Communication

endowed by

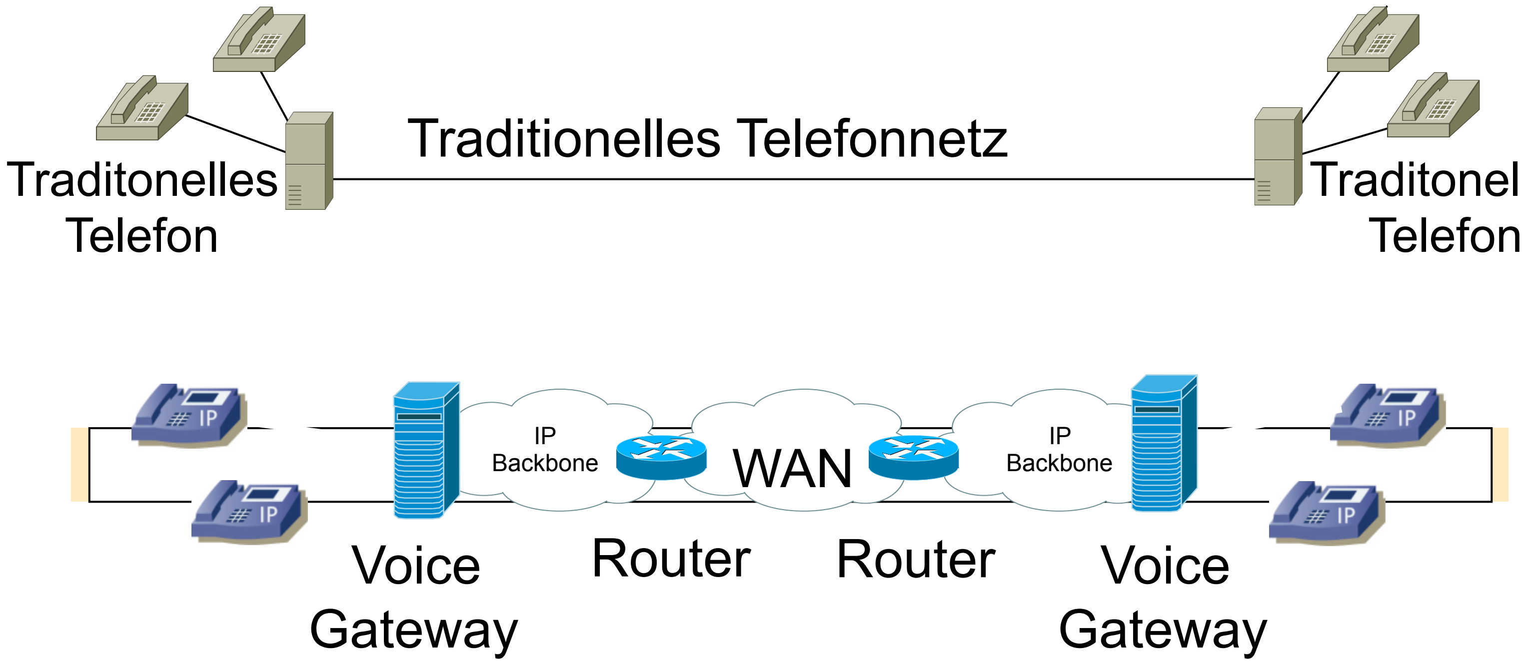
Skype – Ein P2P-Ansatz

- Skype ist aus der KaZaa-Entwicklung hervorgegangen



universität wien

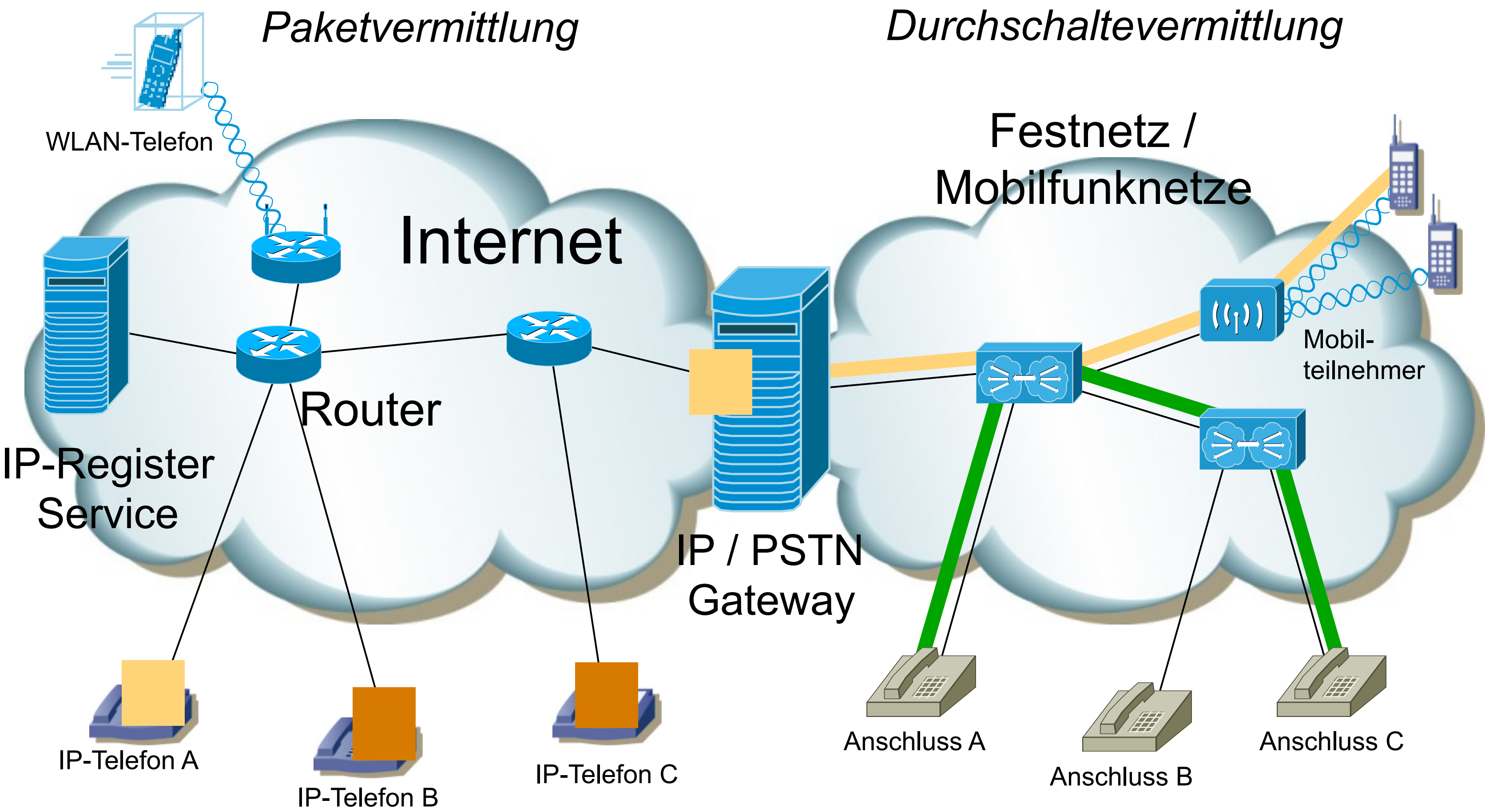
Voice over IP - Umgebung



Chair of **Future Communication**

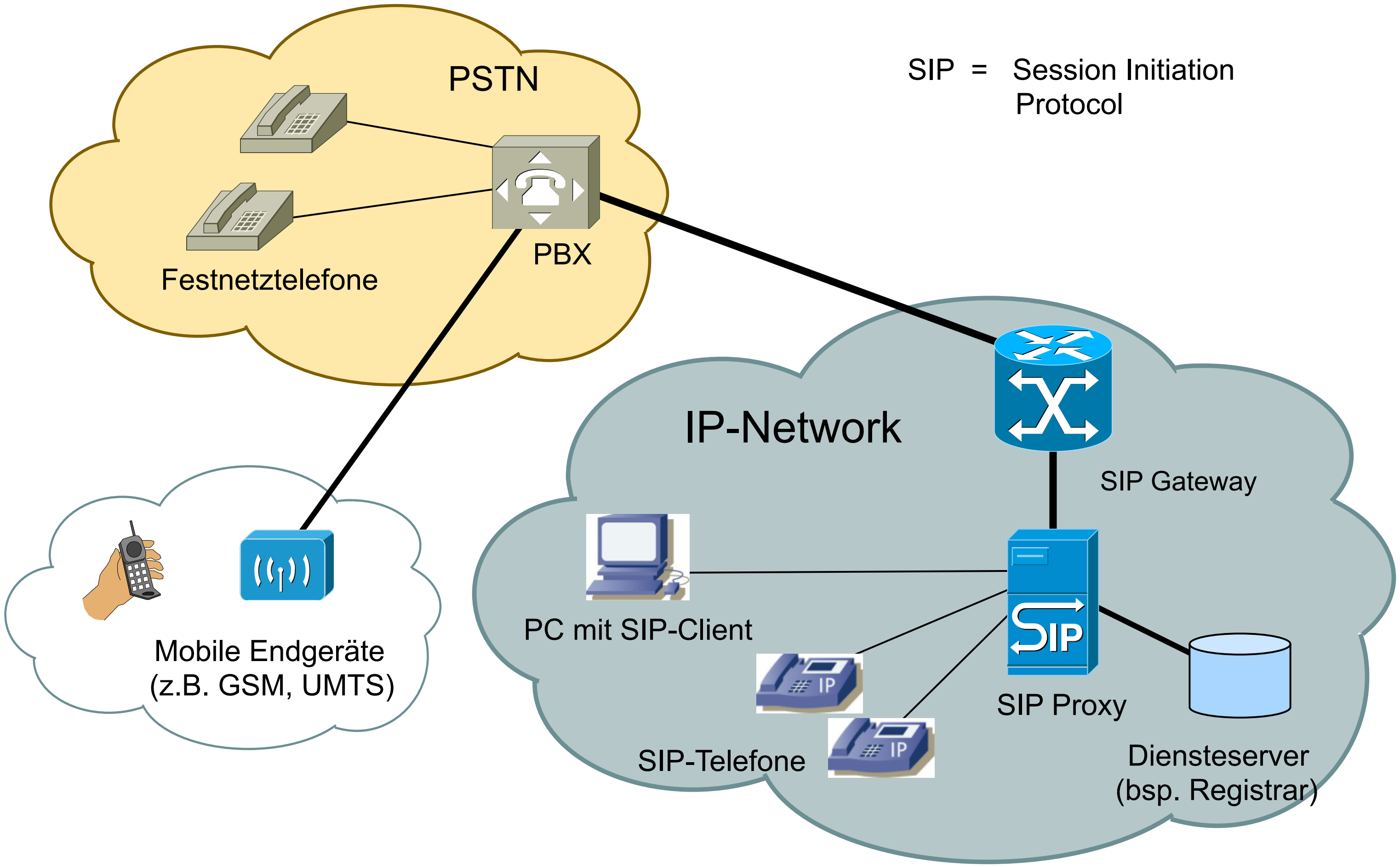
endowed by A1

Vergleich klassiche vs. VoIP-Telefonie



Chair of
Future Communication
endowed by

Aufbau eines SIP-basierten Netzes



SIP = Session Initiation Protocol

Chair of **Future Communication**

endowed by A1

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

Chair of
Future Communication

endowed by

- 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
- Serverlösungen für SIP-Telefonie

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

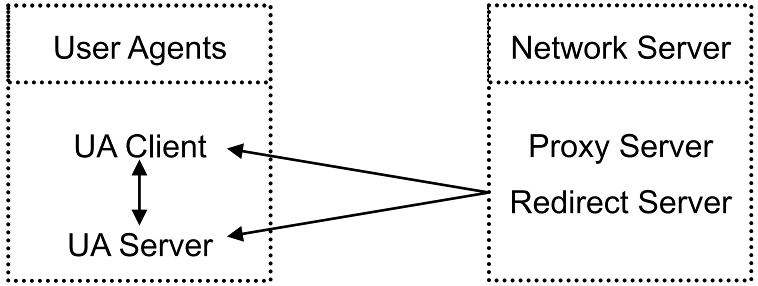
HTTP SIP SMTP	Application
TCP UDP	Transport
IP	Network
	Data Link
	Physical

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

SIP Elements

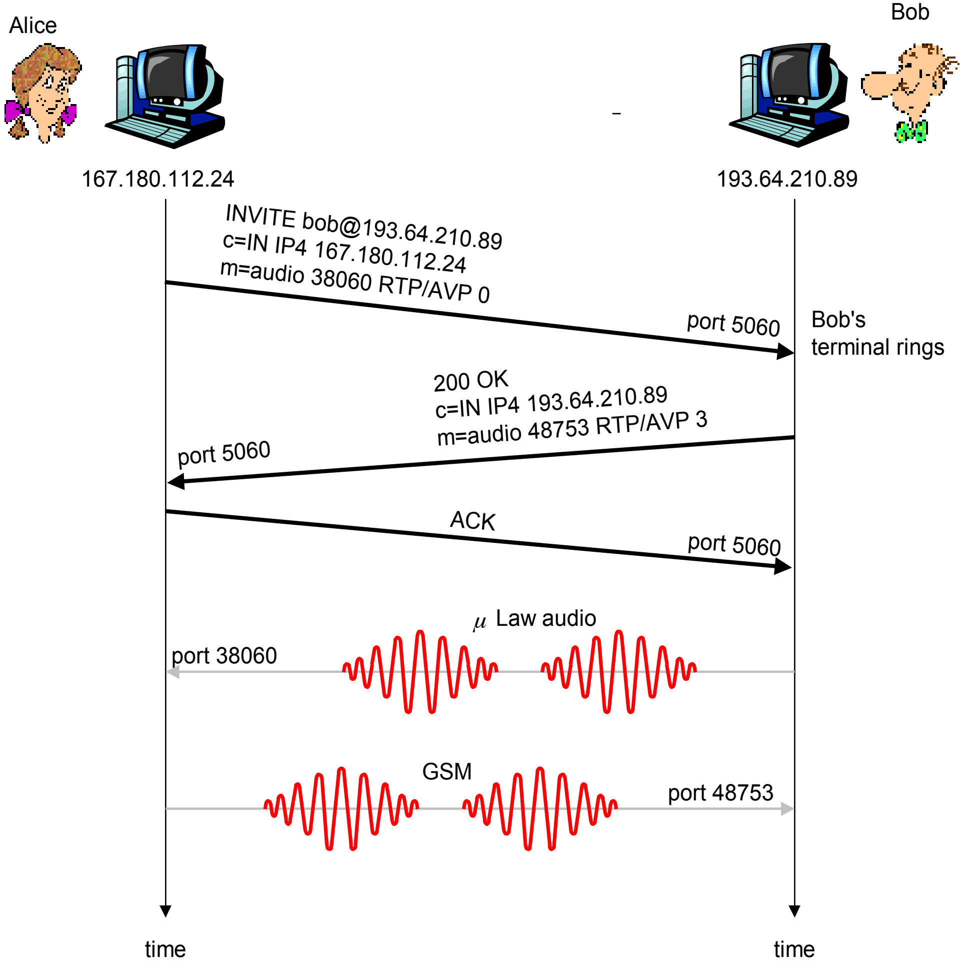
- User agents
  - Want to communicate with each other
  - Examples
    - Application on a user‘s computer
    - Cellular phone
    - PSTN gateway
- SIP proxies or SIP redirect servers
  - Help to find other users
- Registrars
  - Map mnemonic part of sip addresses to IP numbers
  - Usually collocated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



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.

Chair of Future Communication

endowed by

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

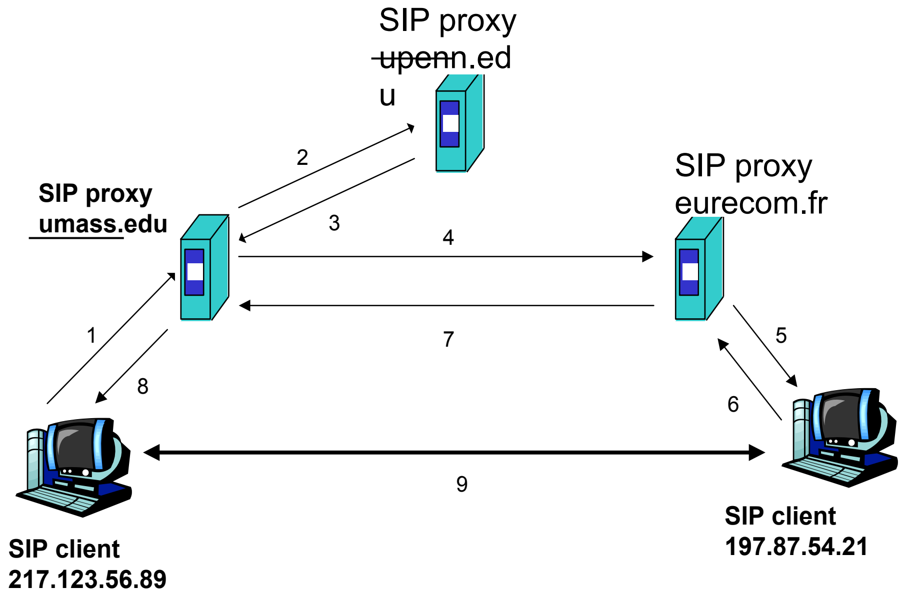
- Caller wants to call callee, but only has callee's name or e-mail address.
- Need to get IP address of callee's current host:
  - User moves around
  - DHCP protocol
  - User has different IP devices (PC, PDA, car device)
- Caller asks SIP server (like DNS server)
- Result can be based on:
  - Time 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



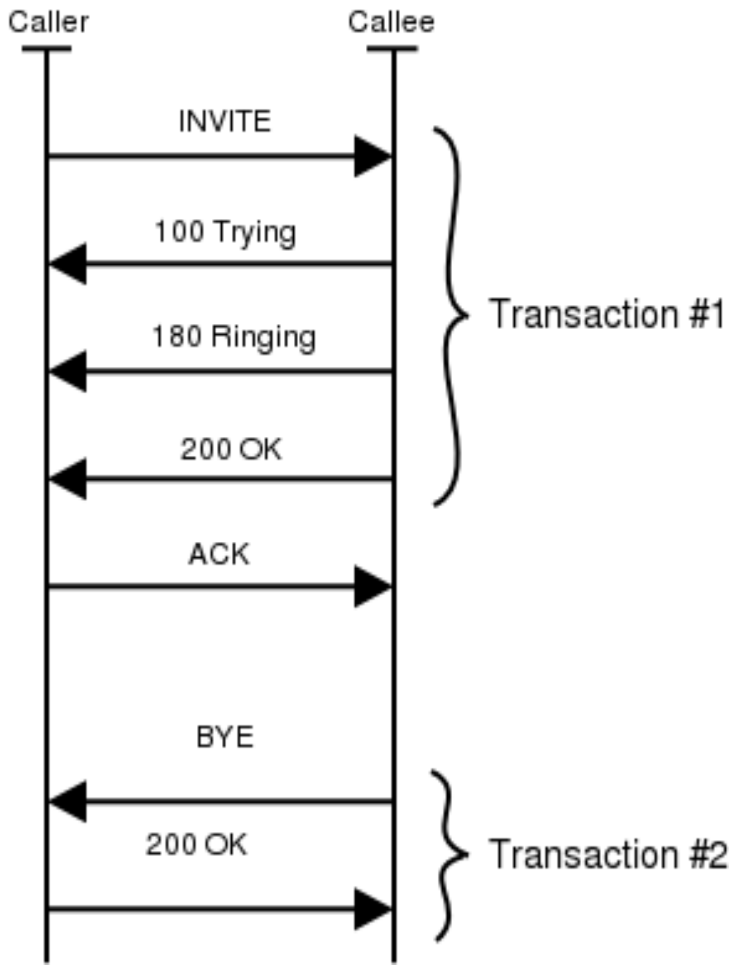
- (4) umass proxy sends INVITE to eurecom SIP proxy/registrar.
- (5) eurecom proxy/registrar forwards INVITE to 197.87.54.21, which is running keith's SIP client.
- (6-8) SIP response sent back
- (9) media sent directly between clients.
- Note: also a SIP ack message, which is not shown.

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.

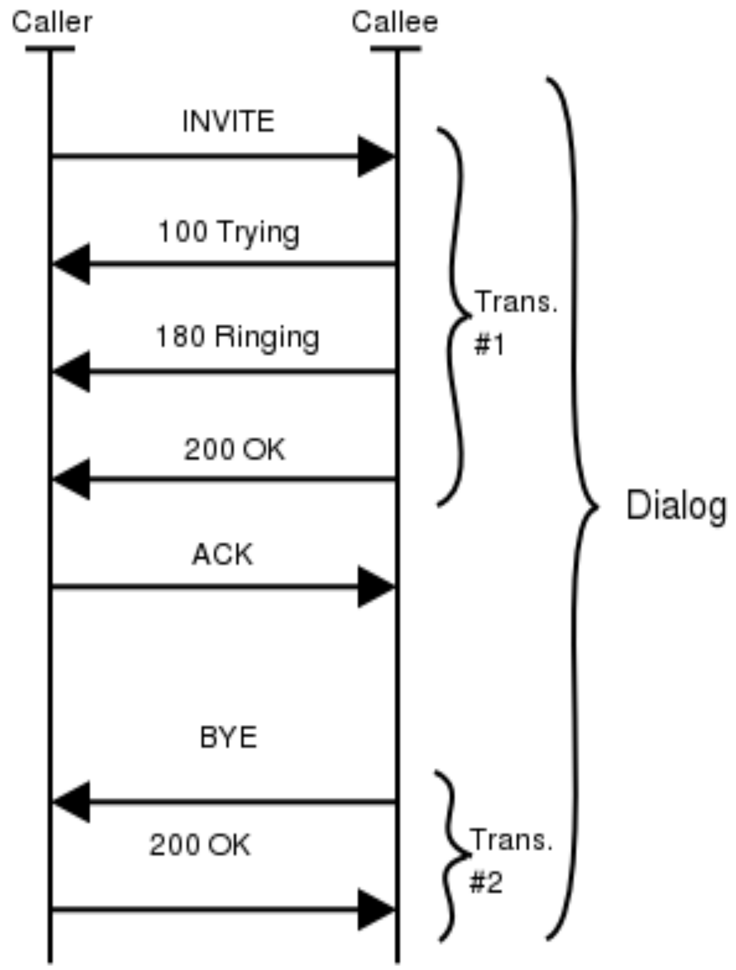
SIP Transactions

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



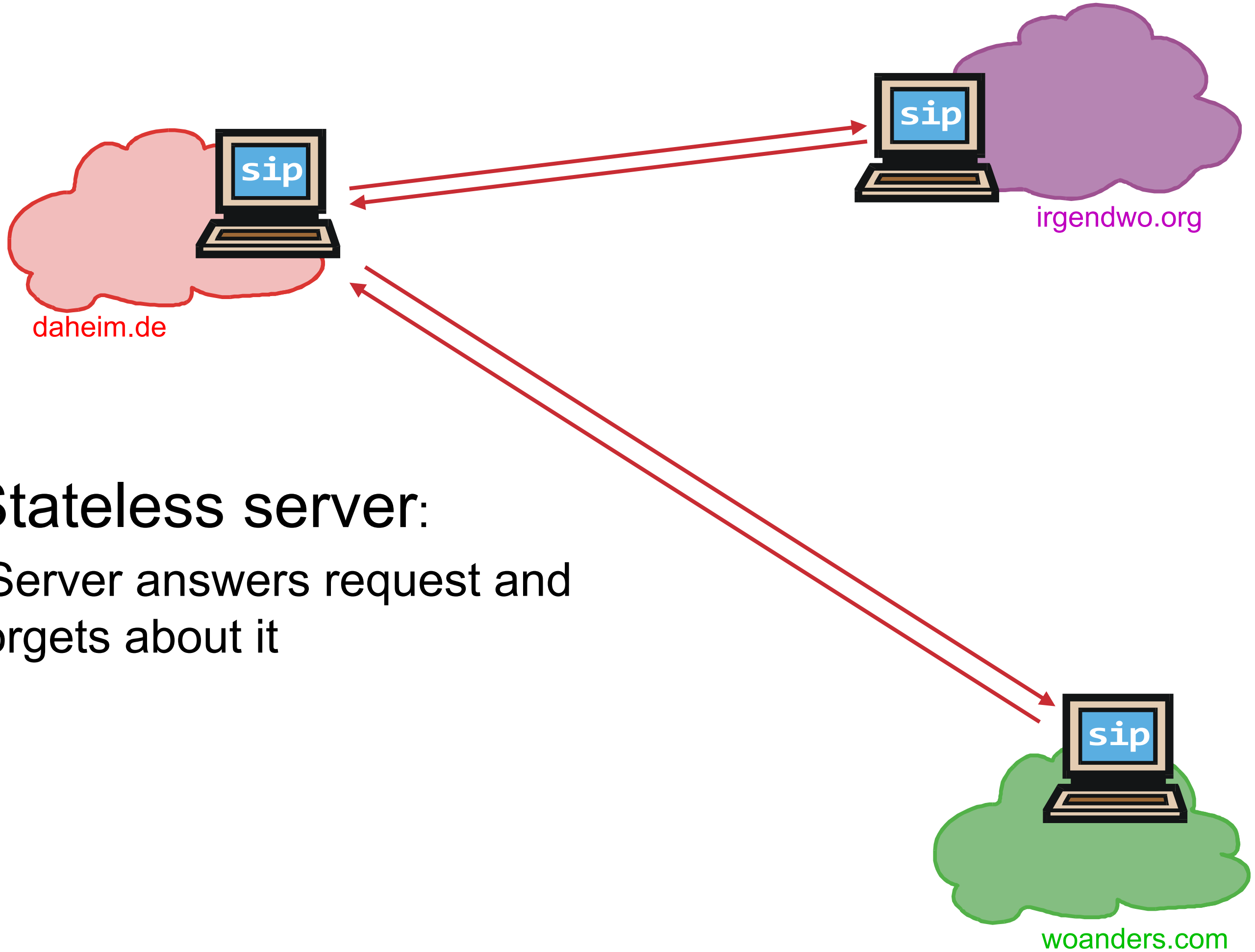
SIP Dialogs

- 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



universität wien

Network Server (1): Redirect Server



Stateless server:

•Server answers request and forgets about it

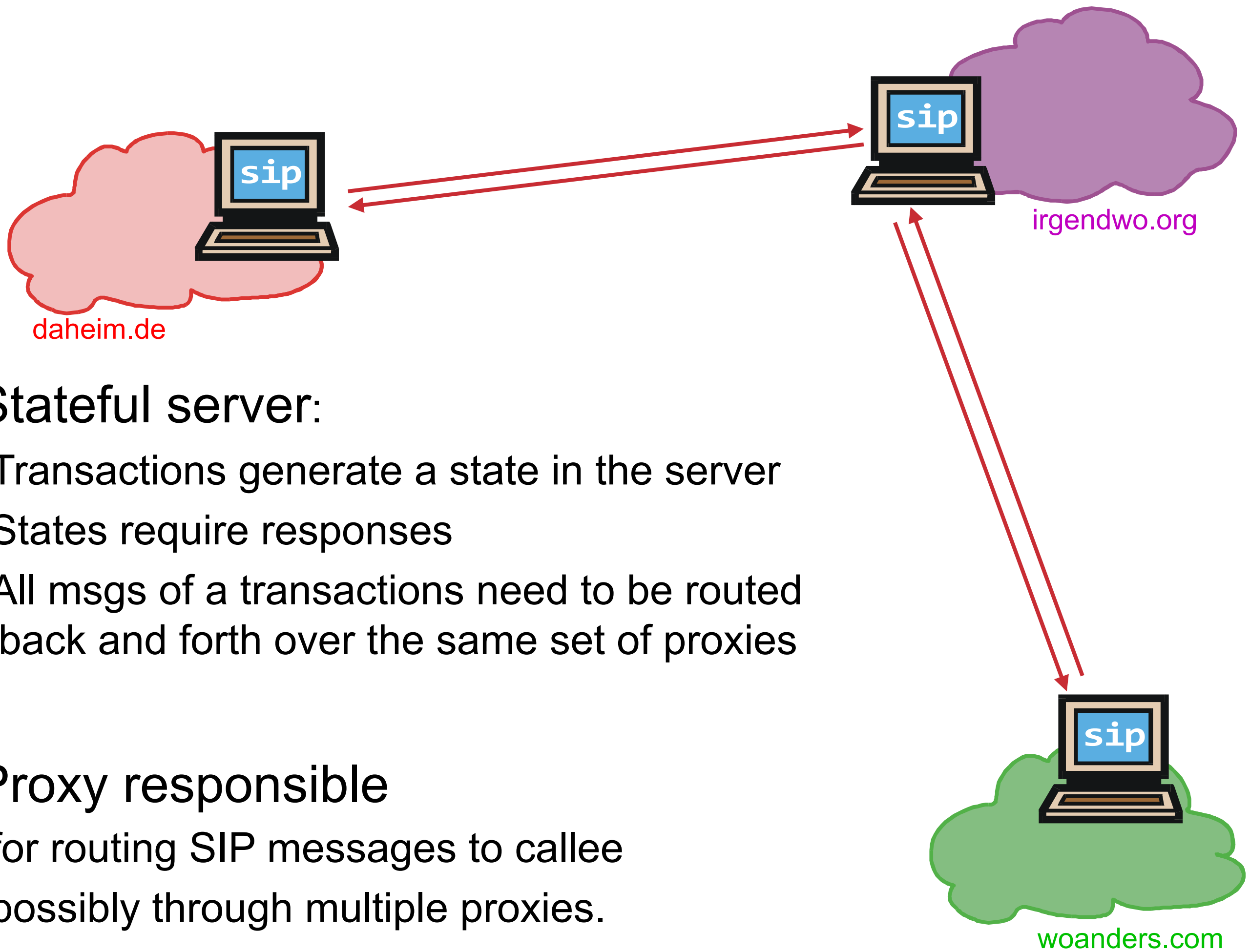
Chair of Future Communication

endowed by A1

universität wien

Network Server (2): Proxy Server

Chair of
Future Communication
endowed by A1



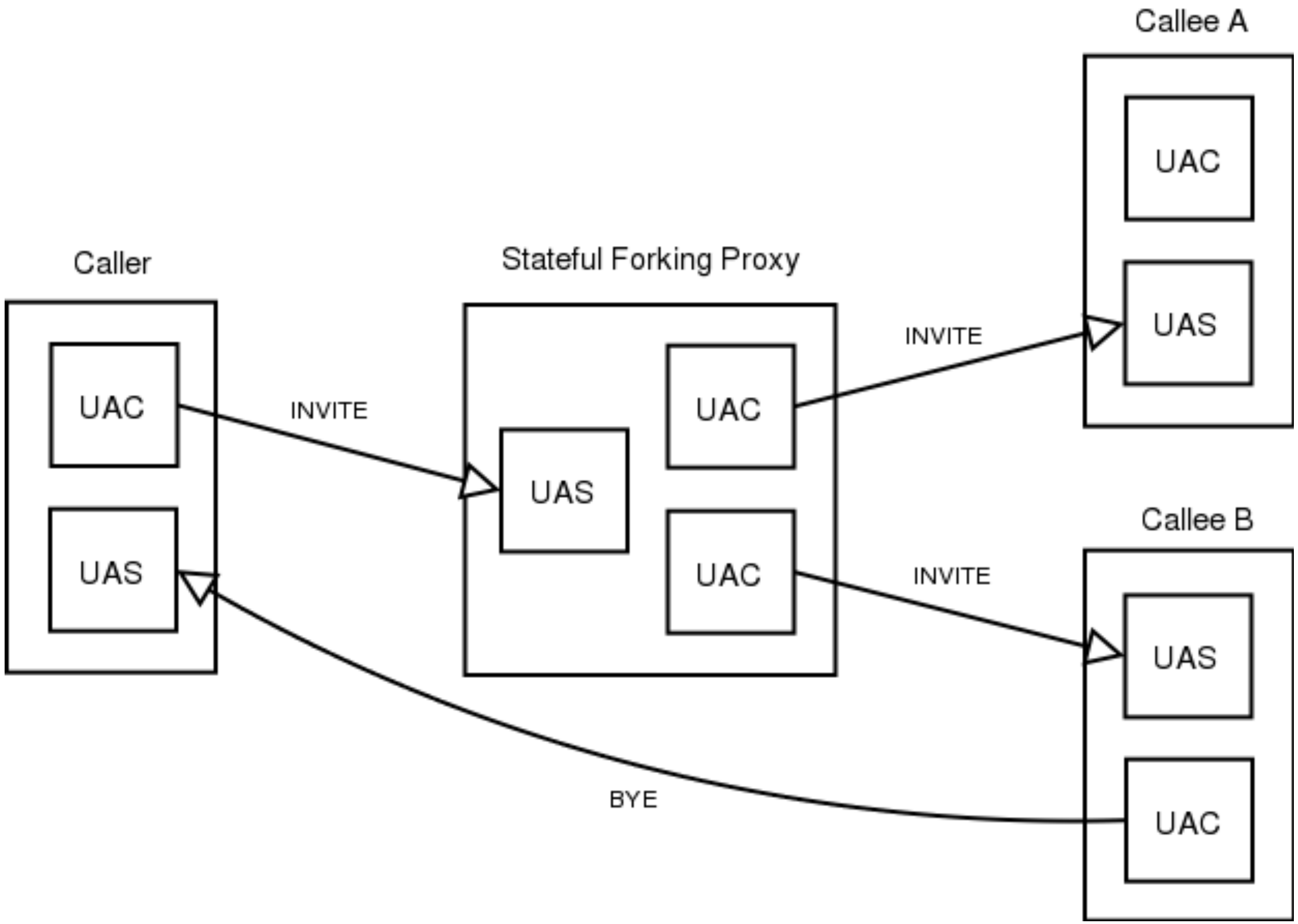
Stateful server:

- 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

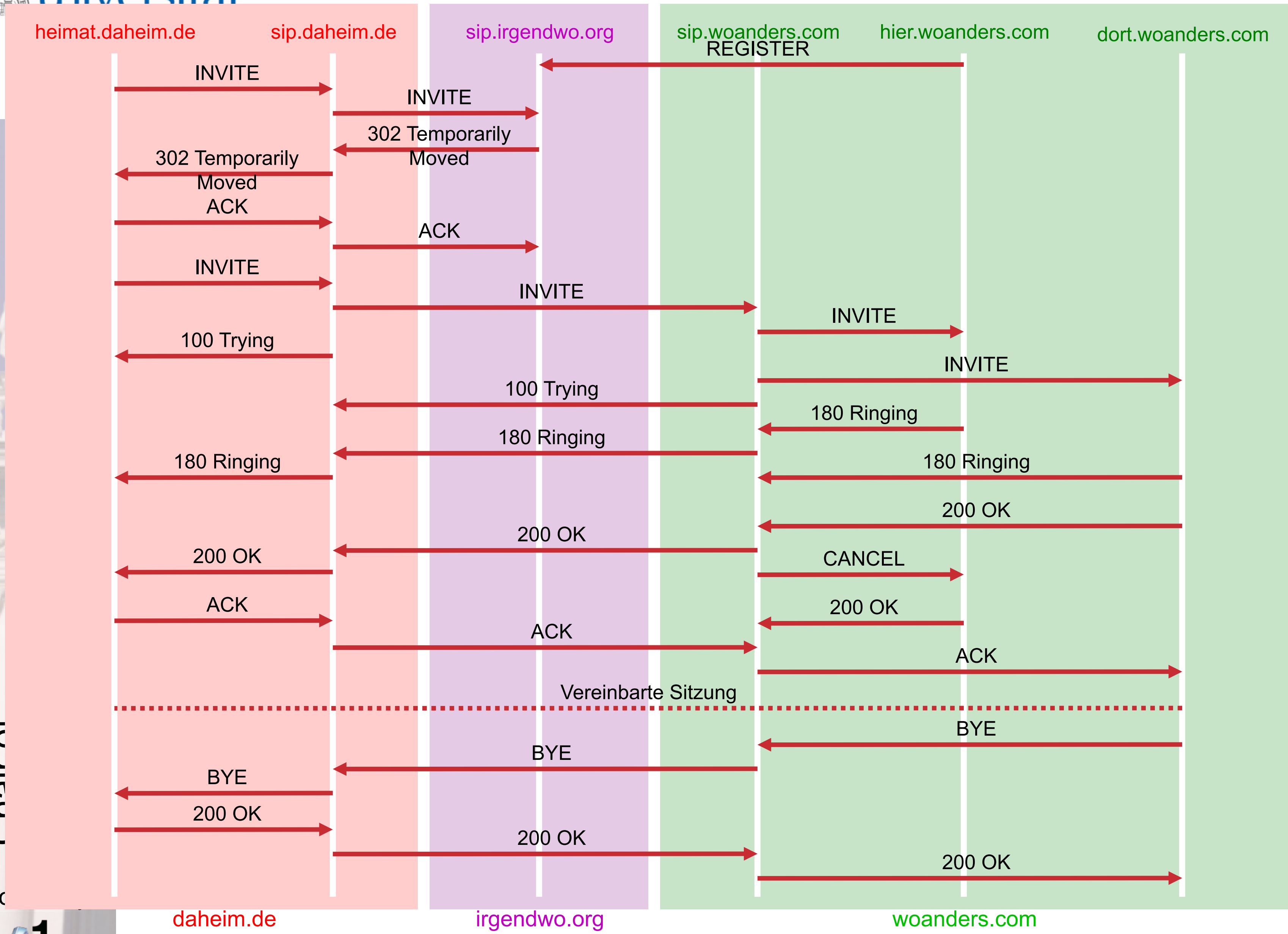


Proxy Types

- 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
CANCEL
200 OK
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.

Chair of **Future Communication**

endowed by

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

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

- Aufbau:

  <Start-Zeile>
  <Header_1>:<Wert_1>
  …
  <Header_n>:<Wert_n>

  < … >

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

SIP Header Types

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

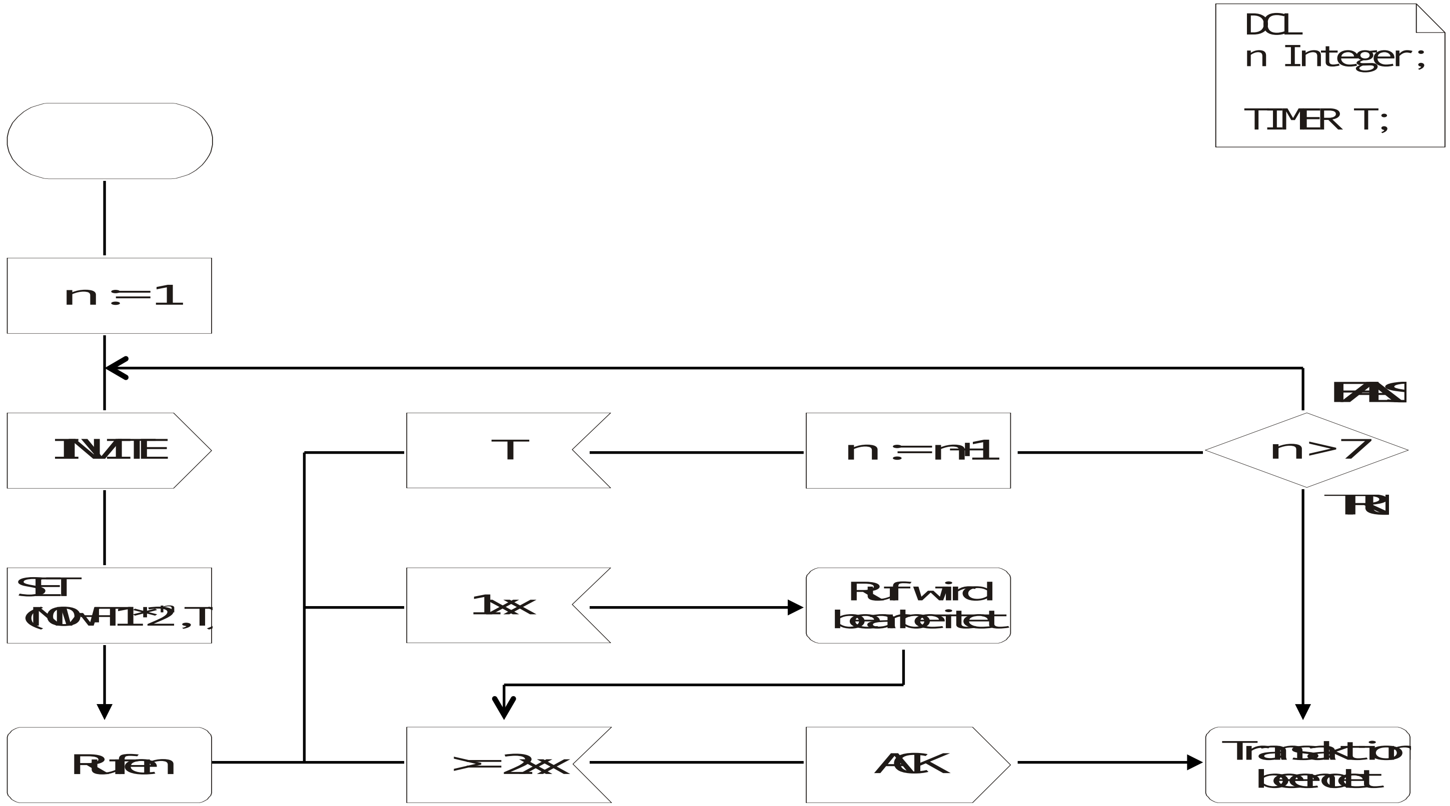
- Wichtige Standard-Header:
  - From: der anrufende Partner
  - To: der gerufene Partner
  - Call-ID: eindeutiger Bezeichner der Sitzung
  - CSeq: Bezeichner für eine Transaktion (Nr + Methode)
  - Via: Route der Nachricht (bisher)
  - Contact: Alternativ-Adresse(n)
  - Require: Erforderliche Optionen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

SDL Diagram for Client Finite State Machine

Chair of Future Communication

endowed by

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

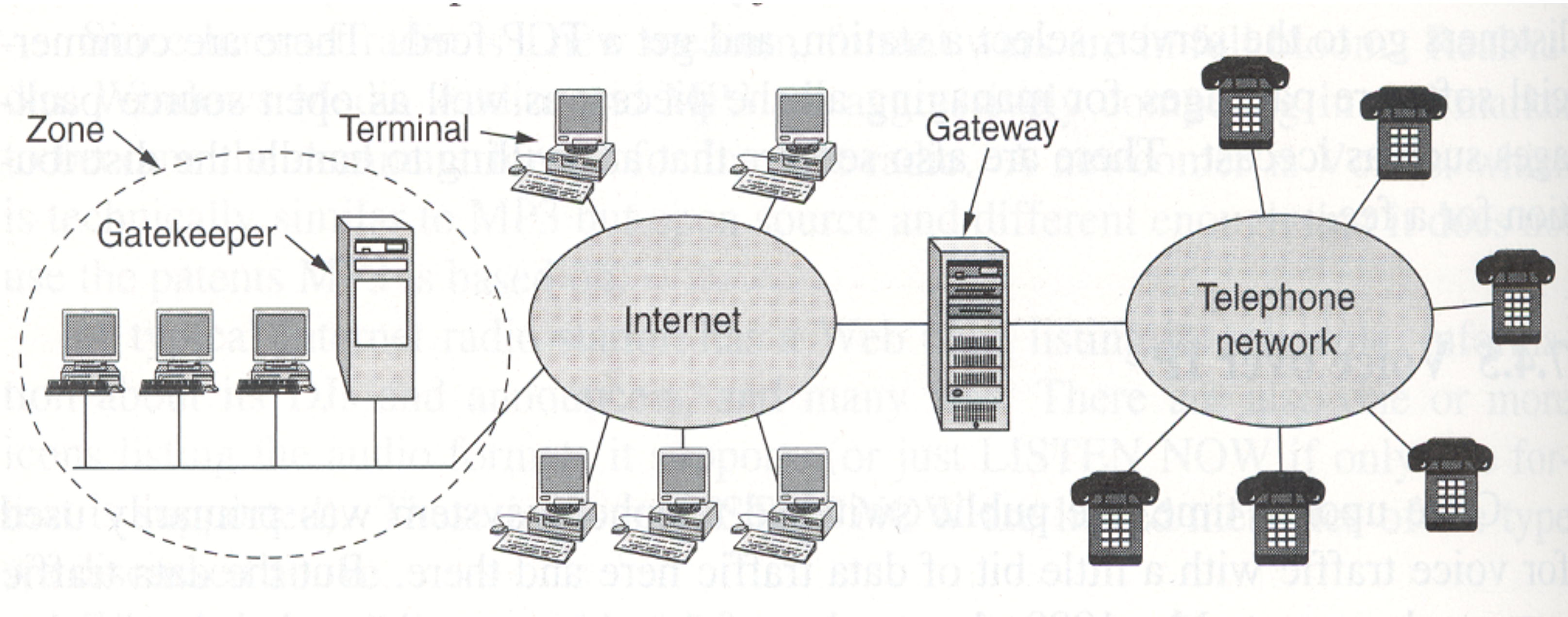
H.323 Overview

- 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

Chair of **Future Communication**

endowed by

H.323 Overview



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

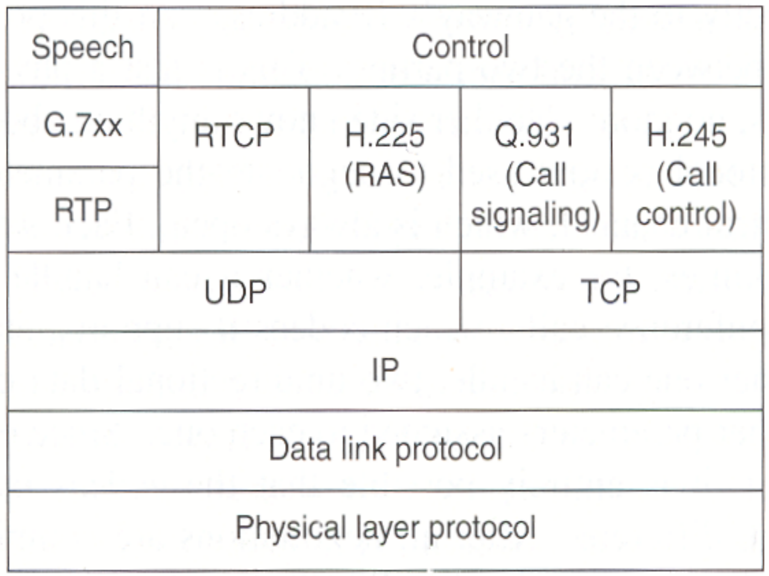
universität wien

H.323 Protocols

Chair of **Future Communication**

endowed by A1

- 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



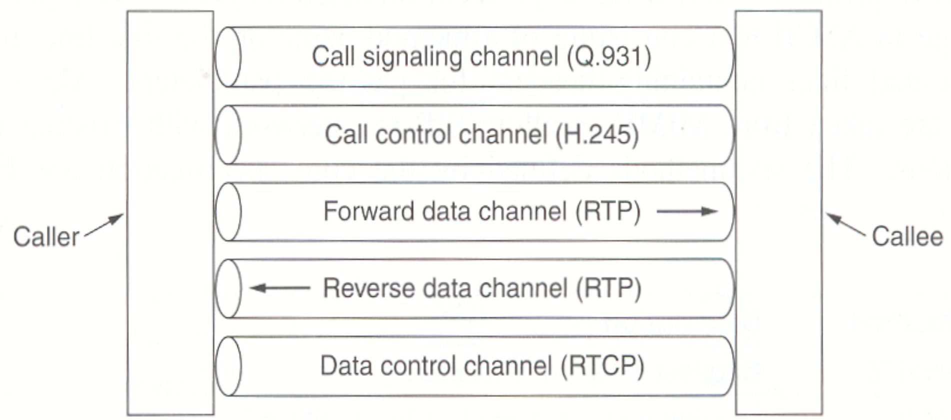
H.323 Signaling Example

- 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

universität wien

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



Chair of
Future Communication

endowed by
A1

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

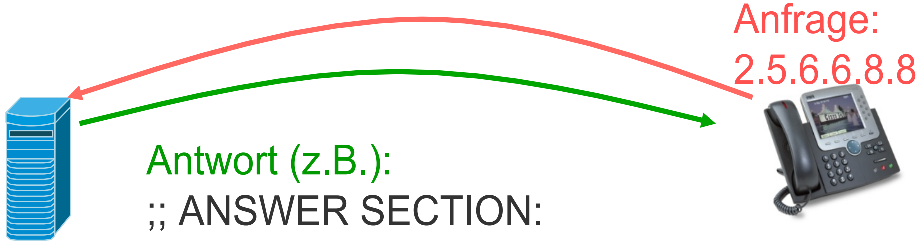
- tElephone NUmber Mapping (ENUM): Anwendung des DNS zur Übersetzung von Telefonnummern in Internet-Adressen (RFC3761)
- Klassische Telefonnummer durch ITU E.164 festgelegt
- ENUM Umsetzung:
  - Eingabe: E.164 konforme Nummer
  - Ausgabe: gültige „absoluteURI“ (RFC2396)
  - Beispiel:

+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:
2.5.6.6.8.8.8.1.3.9.9.4.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!" .

Chair of Future Communication

endowed by

Next Generation Network (NGN) – im engeren Sinne

- 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

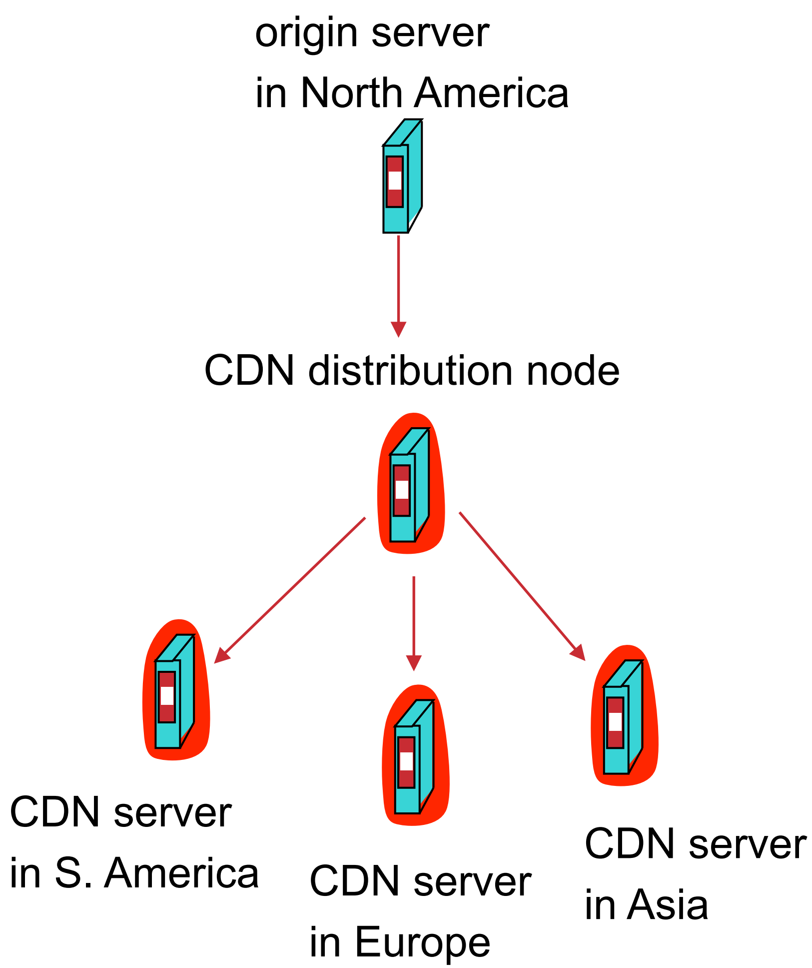
Overview:

- 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

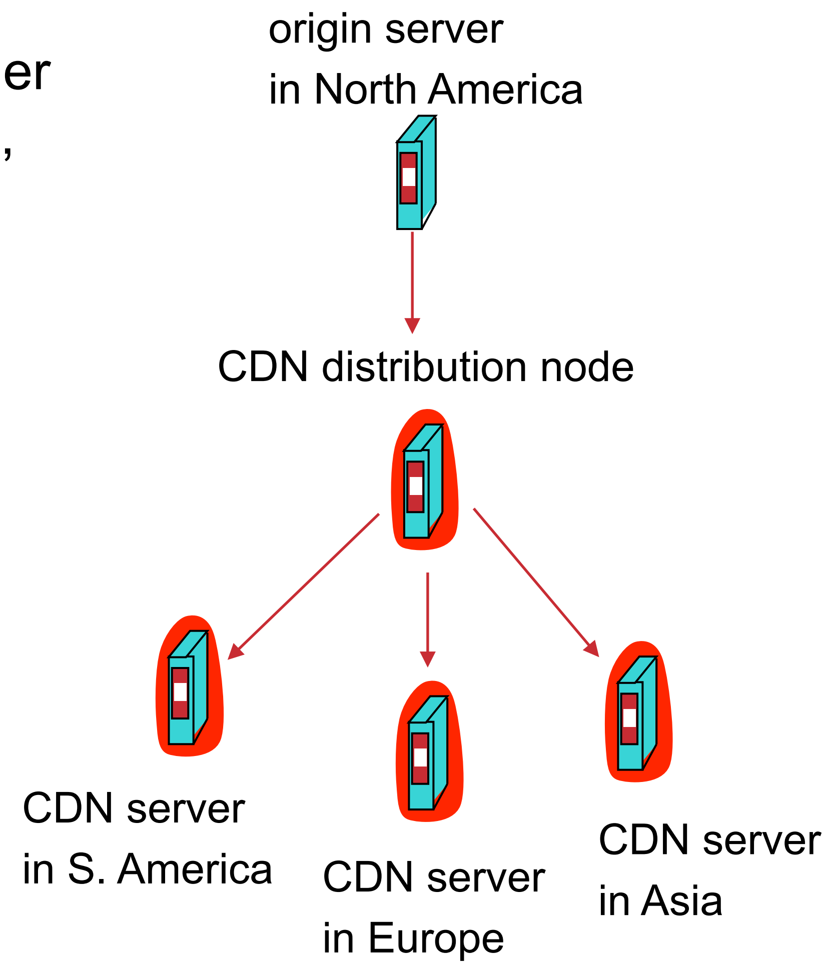
- Challenging to stream large files (e.g., video) from single origin server in real time
- Solution: replicate content at hundreds of servers throughout Internet
  - content downloaded to CDN servers ahead of time
  - placing content "close" to user avoids impairments (loss, delay) of sending content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



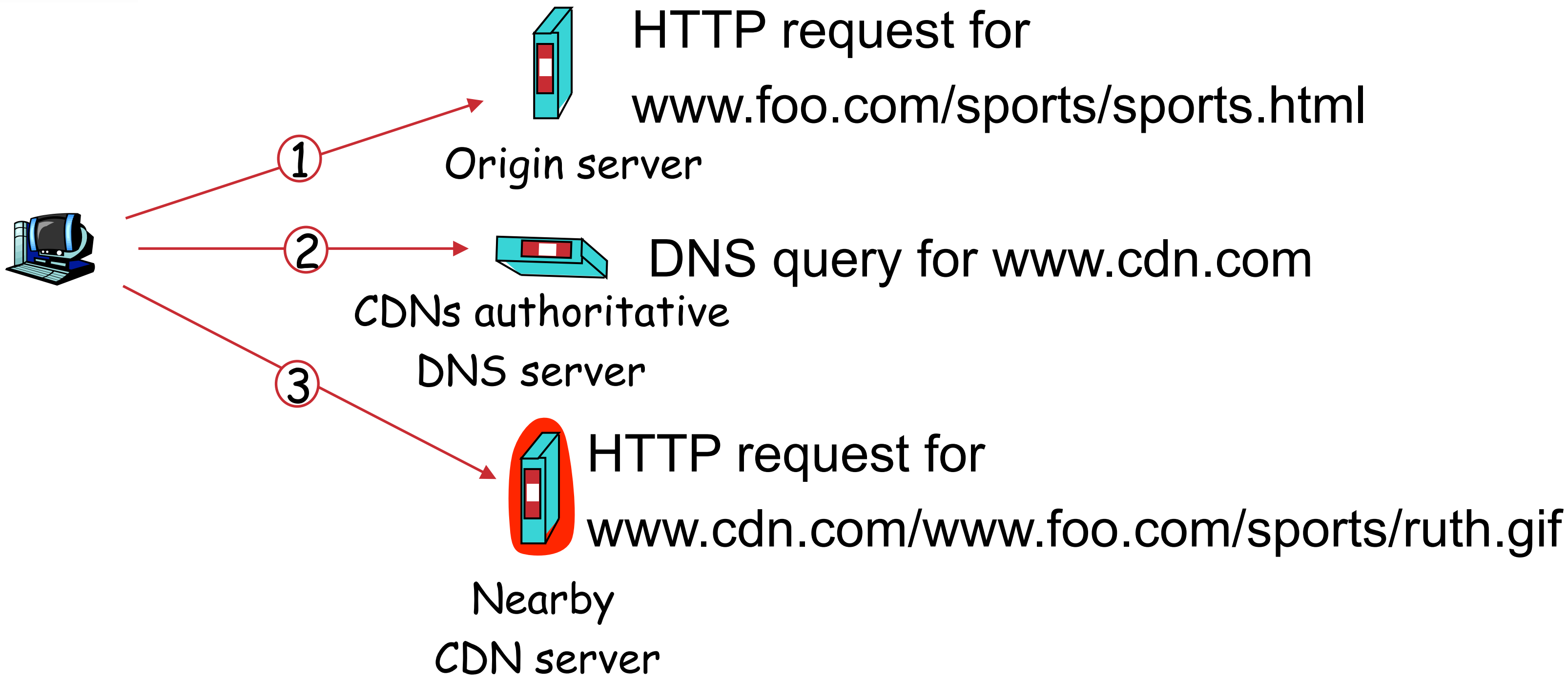
Content distribution networks (CDNs)

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



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