

IP Networks – a basic prerequisite for the cloud

Prof. Dr. Gerhard Schneider
direktor@rz.uni-freiburg.de

Albert-Ludwigs-Universität Freiburg

UNI
FREIBURG

DAAD

Deutscher Akademischer Austausch Dienst
German Academic Exchange Service

The world (for comparison)



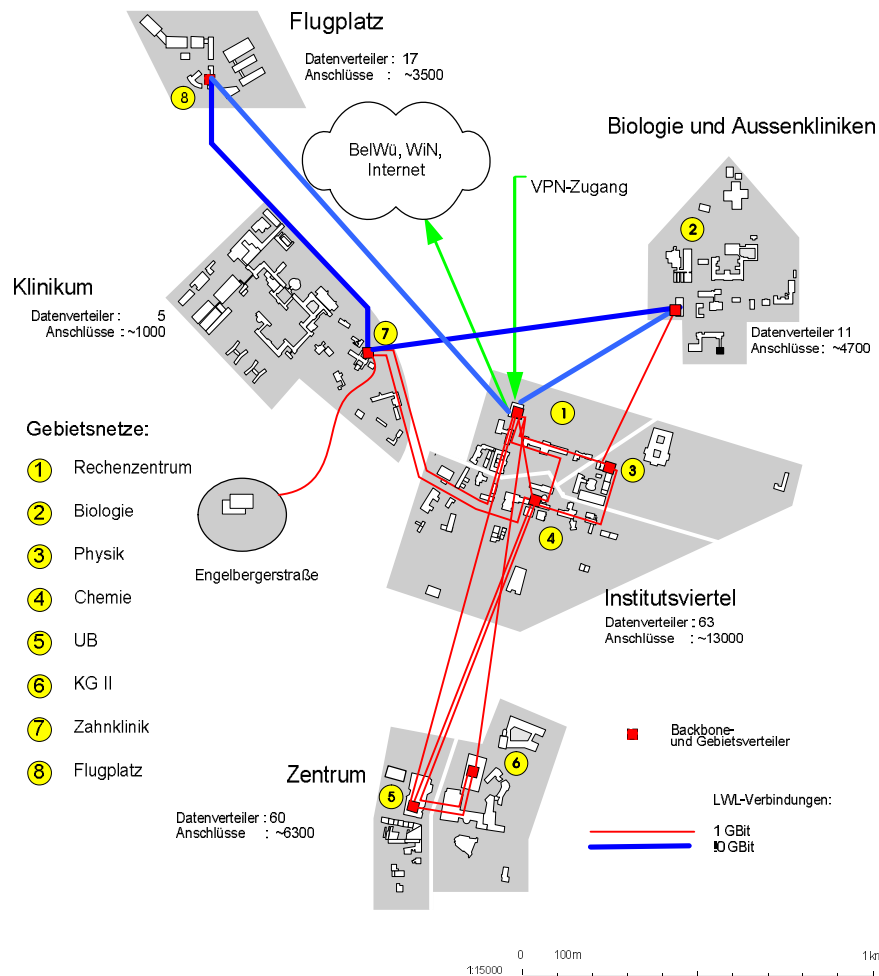
Europe in perspective



Network infrastructure – local view

Freiburger - Universitäts - Netz

Backbone (Mai 2008)



Gebietsnetze

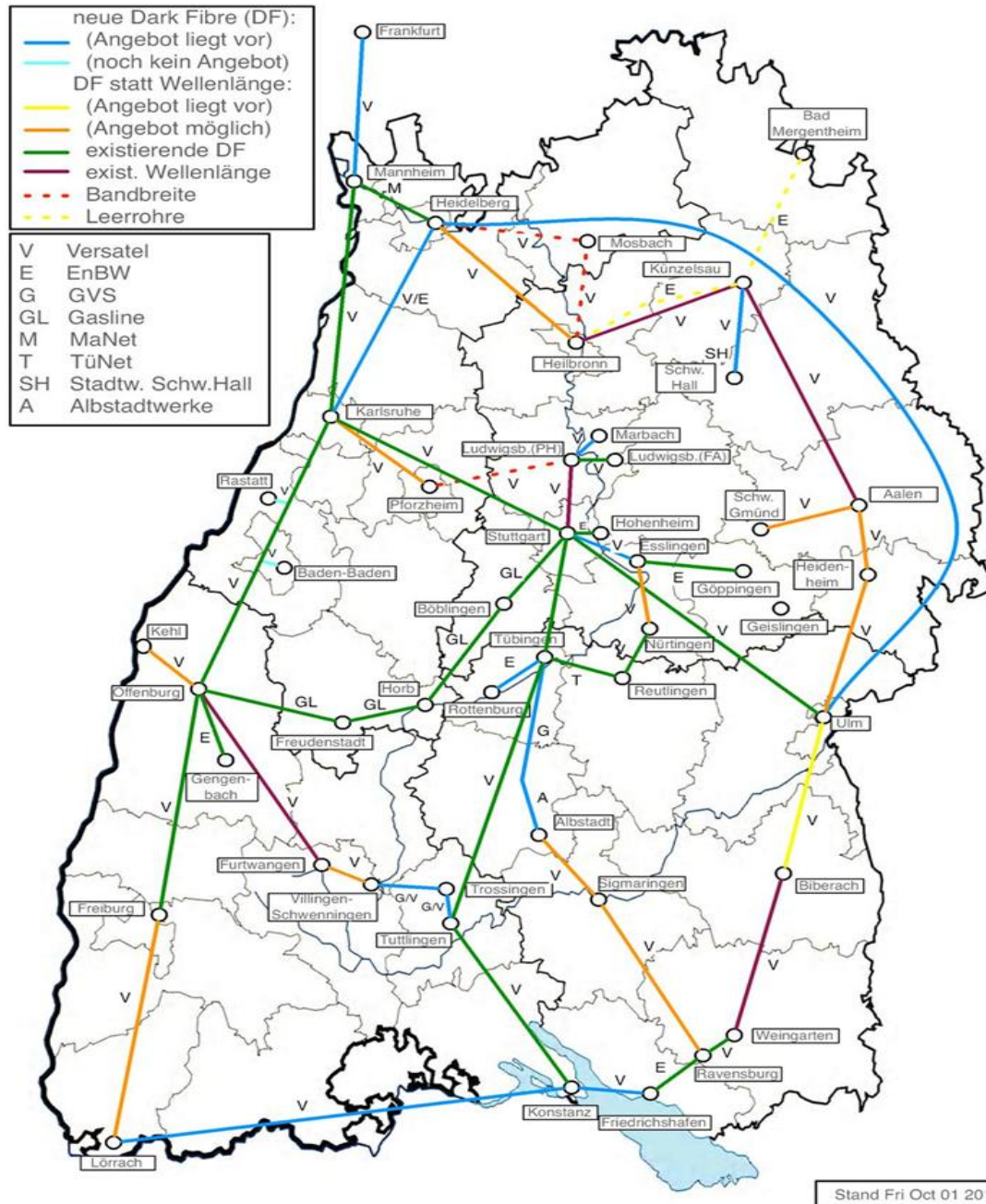
○ In den **172 Datenverteilern** stehen ca **32000 Anschlüsse** zur Verfügung.

✓ aktiv mit 100MBit / 1GBit sind ca. **19000 Anschlüsse**, davon wurden allein in den letzten 3 Monaten **2700 Anschlüsse** realisiert.

✓ Biologie	:	3182
✓ Chemie	:	2823
✓ Informatik	:	1890
✓ KG I.. KG IV	:	2408
✓ Klinik (Lehre)	:	379
✓ Physik	:	1554
✓ RZ	:	2264
✓ UB	:	1624

Belwue

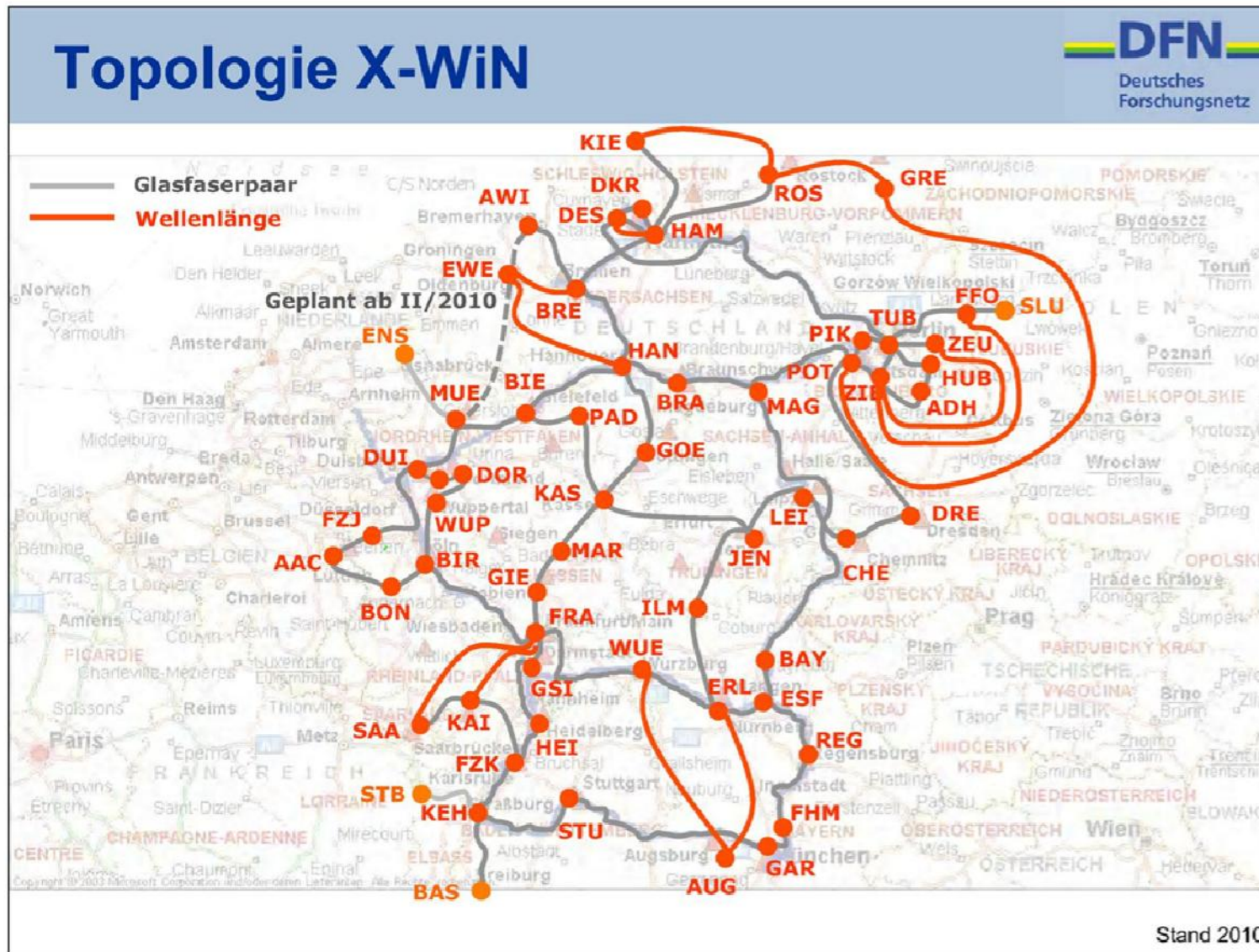
The State high speed network for science and research



Belwue structure

- Very interesting history 😊
 - Founded around 1988 as a non-OSI network
- Mainly funded by the state of Baden-Württemberg
 - The universities pay a small fee, e.g. Freiburg contributes 100.000€ a year for a 10 Gbit/s link
 - Idea is to support the smaller institutions as well – so that they get access to the infrastructure without being hampered by their small budgets
 - Operated by a network operation center in tight cooperation with the universities' IT centres
 - Planning is done mainly by the IT centre directors together with the MR from the ministry
- Very unusual construction – and fairly unique
- But highly efficient
 - Since it works, we are always ahead of the rest
- Based on dark fibre (leased or bought)
 - No dependency on commercial business plans / mergers / money making /

German science network

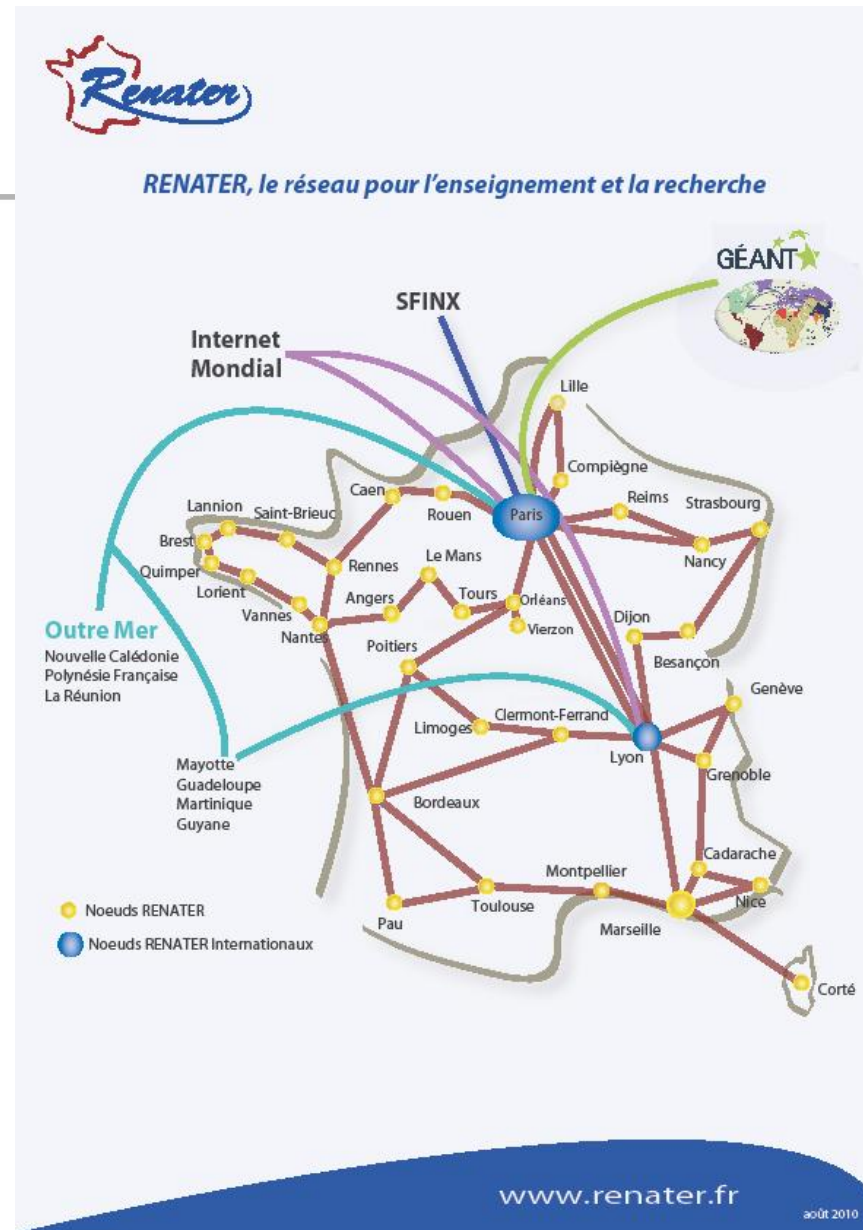


DFN – Deutsches Forschungsnetz

- Originally a union of interest of the German universities, founded around 1984, to organize a nation wide network on a research basis
 - Remember, we are a highly federal state
 - And the federal state must not interfere with education issues
- „research on networks“ allowed for federal funding
 - Required an OSI/X.25 focus ☹
- Now, DFN leases dark fibres and offers advanced services over the network
 - Backup of IT-centres, PKI, CERT, IPv6,
- Pretty large, some 60 staff (??)
- Today, DFN is autonomous – no government money required
 - At least for continuous evolution of day-to-day business
 - The government is confused.... ☺

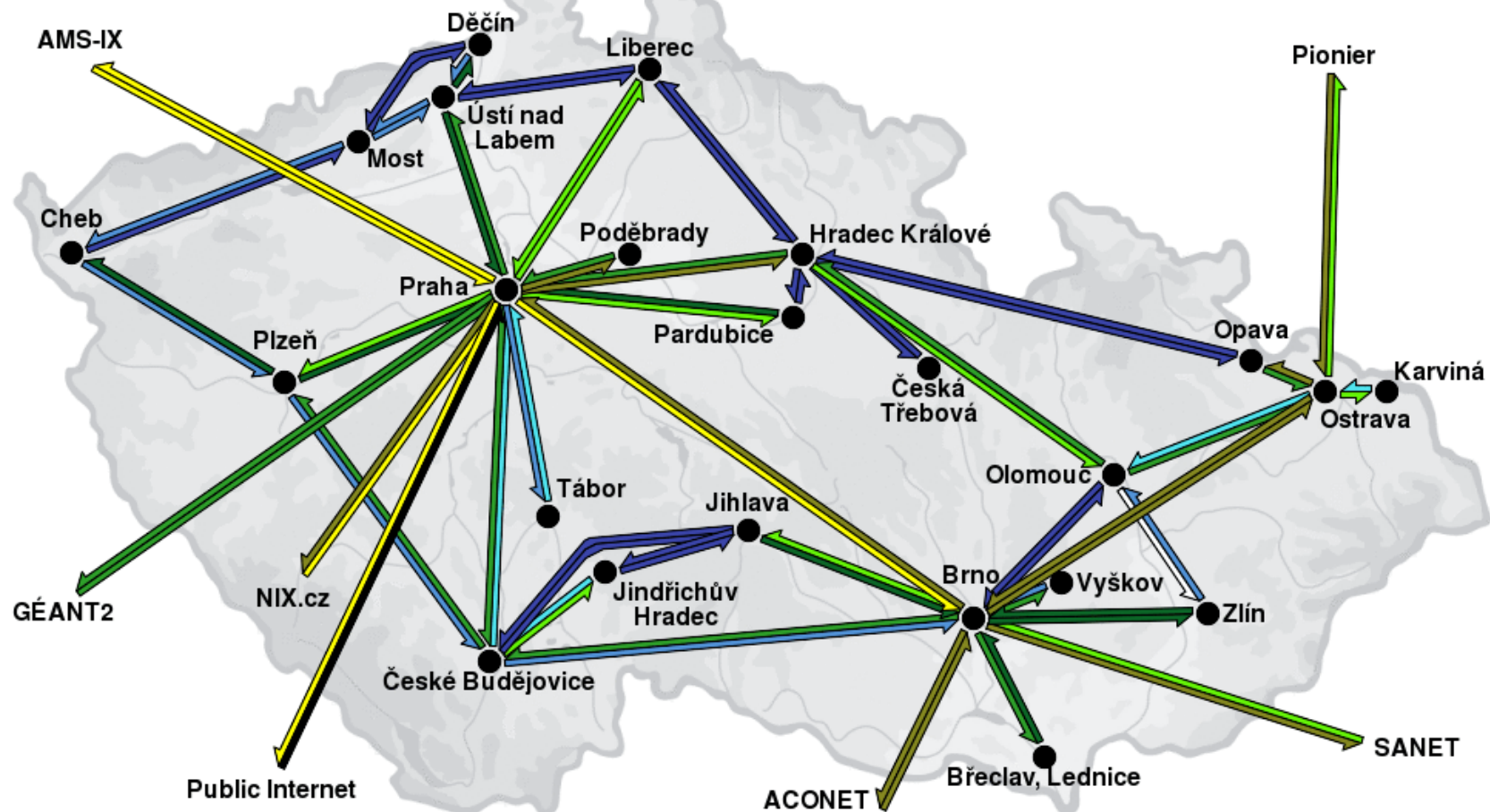
Examples of other national science networks

France



Examples of other national science networks

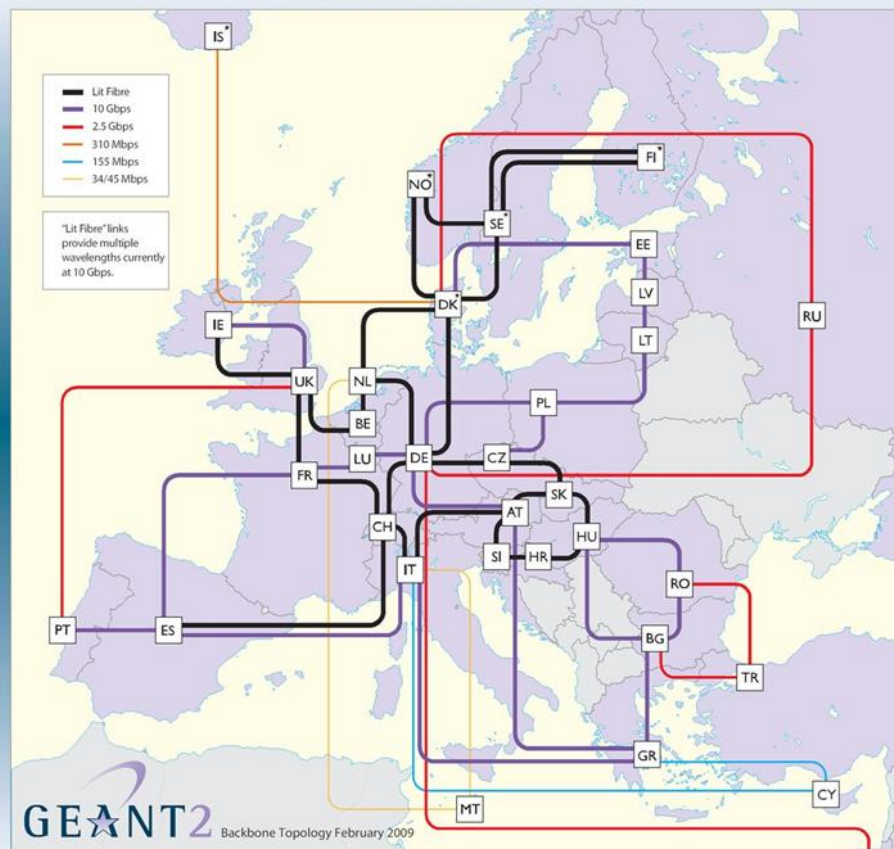
CSNET – Czech Republik



Europe

- European organisation with all national state networks as members
- Funding by the European Commission
- European backbone
 - Leads to some funny geographical connections
 - Regional improvements (eg Belwue)

The first international hybrid research and education network.
Lighting dark fibre for greater network performance.



GEANT2 is operated by DANTE on behalf of Europe's NRENs.

AT Austria	CZ Czech Republic	ES Spain	HR Croatia	IS Iceland*	LV Latvia	PL Poland	SE Sweden*
BE Belgium	DE Germany	FI Finland*	HU Hungary	IT Italy	MT Malta	PT Portugal	SI Slovenia
BG Bulgaria	DK Denmark*	FR France	IE Ireland	LT Lithuania	NL Netherlands	RO Romania	SK Slovakia
CH Switzerland	EE Estonia	GR Greece	IL Israel	LU Luxembourg	NO Norway*	RU Russia	TR Turkey
CY Cyprus							UK United Kingdom

*Connections between these countries are part of NORDUnet (the Nordic regional network)

GEANT2 is co-funded by the European Commission within its 6th R&D Framework Programme.



Géant – a few facts

The Project

The GÉANT2 network provides the high-performance, state-of-the-art network infrastructure that is fundamental to the European Union's vision of a European Research Area (ERA). The network is the core activity of a coherent set of initiatives that seek to develop all aspects of European research and education networking. The project within which the network is being built and developed also includes an integrated research programme, the development of support services for network users, initiatives to monitor and address disparities in the level of development of research and education networking around Europe, and a comprehensive study into the future of European research and education networking.

The partners in the project are [30 European NRENs](#), DANTE and TERENA.

Objectives

The project's overall objectives are:

To plan, build and operate a [multi-gigabit pan-European backbone research network](#) interconnecting Europe's national research and education networks (NRENs), over which a suite of advanced services will be offered to meet the increasingly demanding requirements of Europe's research and education community

To conduct [joint research into the development of networking technologies and services](#), with the primary aim of developing ideas from concept to production service to directly serve the users of GÉANT2 and its connected NRENs

To [support effectively and directly projects and users](#) who have advanced networking requirements

To pursue initiatives targeted at [closing the 'digital divide'](#), through both in-depth analysis of the picture of research networking in developing areas and the provision of direct support

To [examine the future of research networking](#), exploring the case for the sustaining of research and education networking beyond the conclusion of the project.

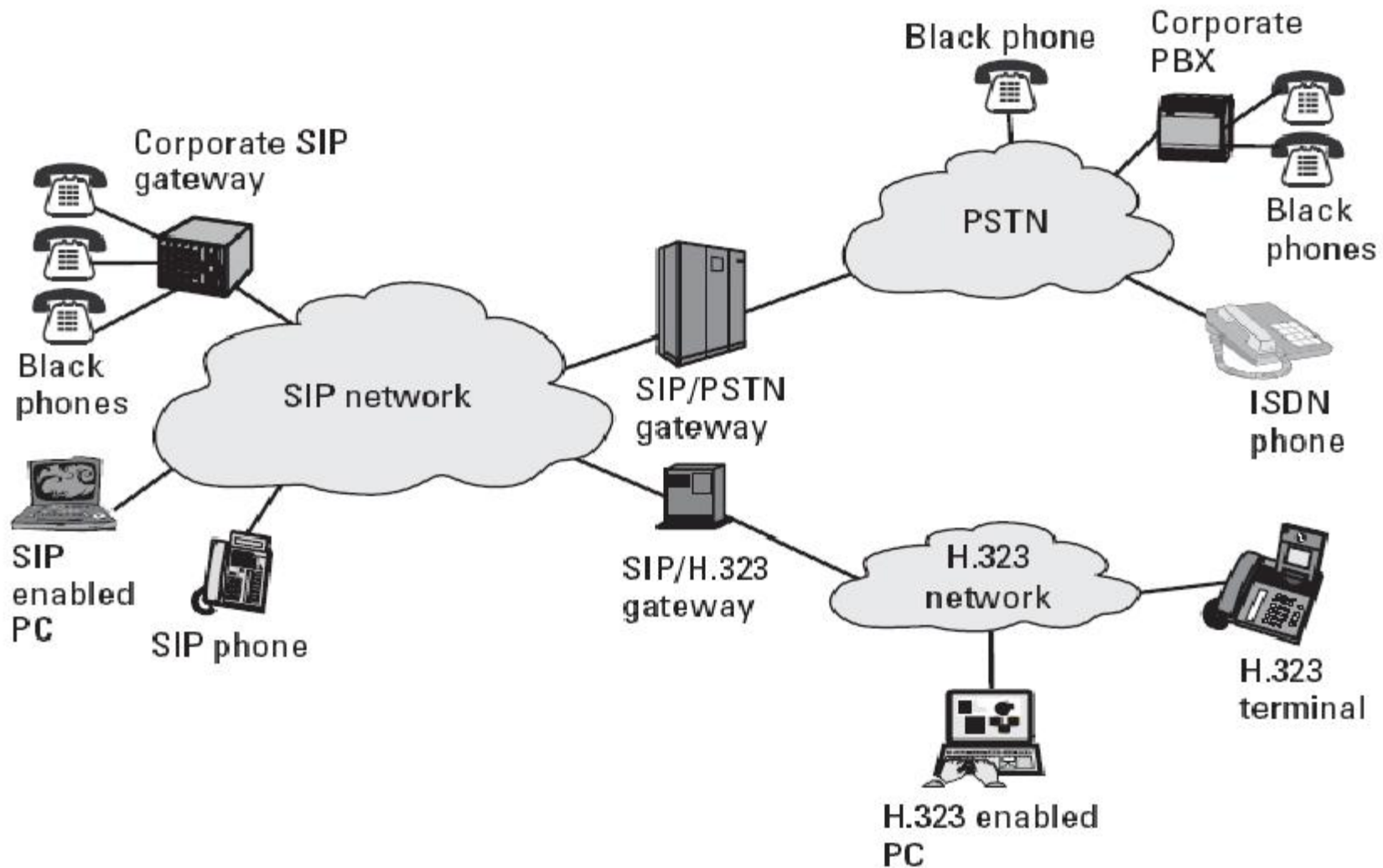
Funding

The programme of activities is co-funded by the European Commission within the GN2 contract, which is part of the EC's Sixth R&D Framework Programme (often referred to as FP6). The remainder of the funding is provided by the NRENs that will be connected to the network.

Question...

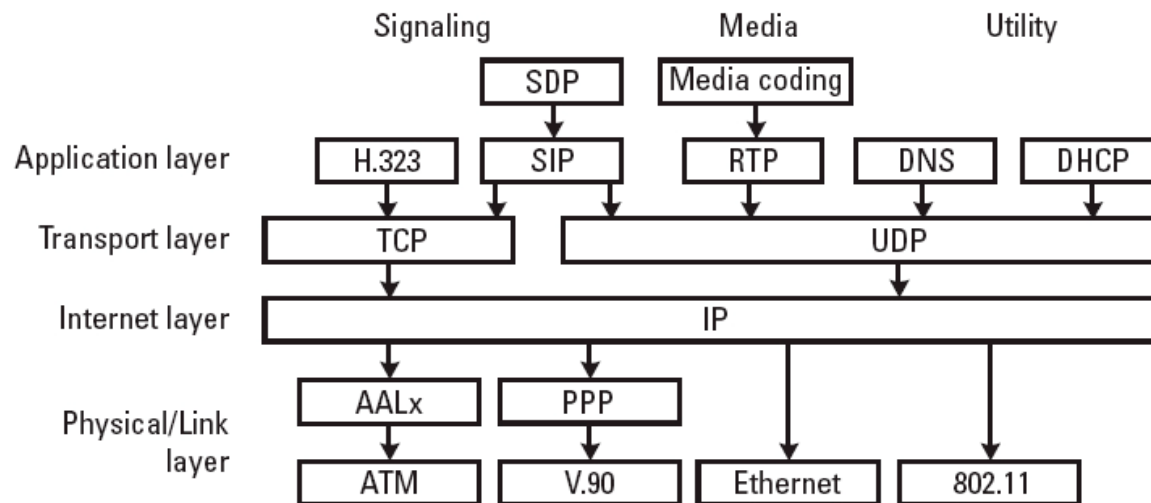
- How can we build a GRID or a Cloud, if the underlying infrastructure is highly complex and individually dependent on funding issues ??
 - Answer: ignore it.
- But it highlights the legal / commercial side: if a cloud provider signs a legal contract with a customer, he must make sure that all his components can be combined over a part of the Internet that can be legally administered
- Consequences for network neutrality?
- Consequences for market forces?
 - Who will be able to compete / who will survive?
 - Are we generating the worst monopolies we can imagine?
 - Suddenly Microsoft may appear as the „good guy“, selling us software for local installation 😊

VoIP as a cloud service



Internet telephony - SIP

- SIP just for session setup not for transport of multimedia streams
- inspired by HTTP
 - text based Peer-to-Peer application layer protocol
 - using requests and replies to set up a connection



Internet telephony - SIP

- Requirements for SIP.
 - localization of endpoints (network view, not geographical)
 - setup of connections
 - exchange of media and presence information
 - modification of sessions: rerouting and cancelling of calls
 - complete a session
 - scalability (more than one session should be possible)
- SIP addresses designed same way as email addresses
 - sip: “userID@sipgateway.site”

SIP - Entities

- Peers = User Agents (UA)
- a UA can fulfill one of the following roles
 - user agent client (UAC) = initiator of a request
 - user agent server (UAS) = application, which contacts the user and answers requests for him
- SIP clients
 - telephones: as UAC or UAS
 - Gateways: connections to other networks, translates between different audio and video codecs

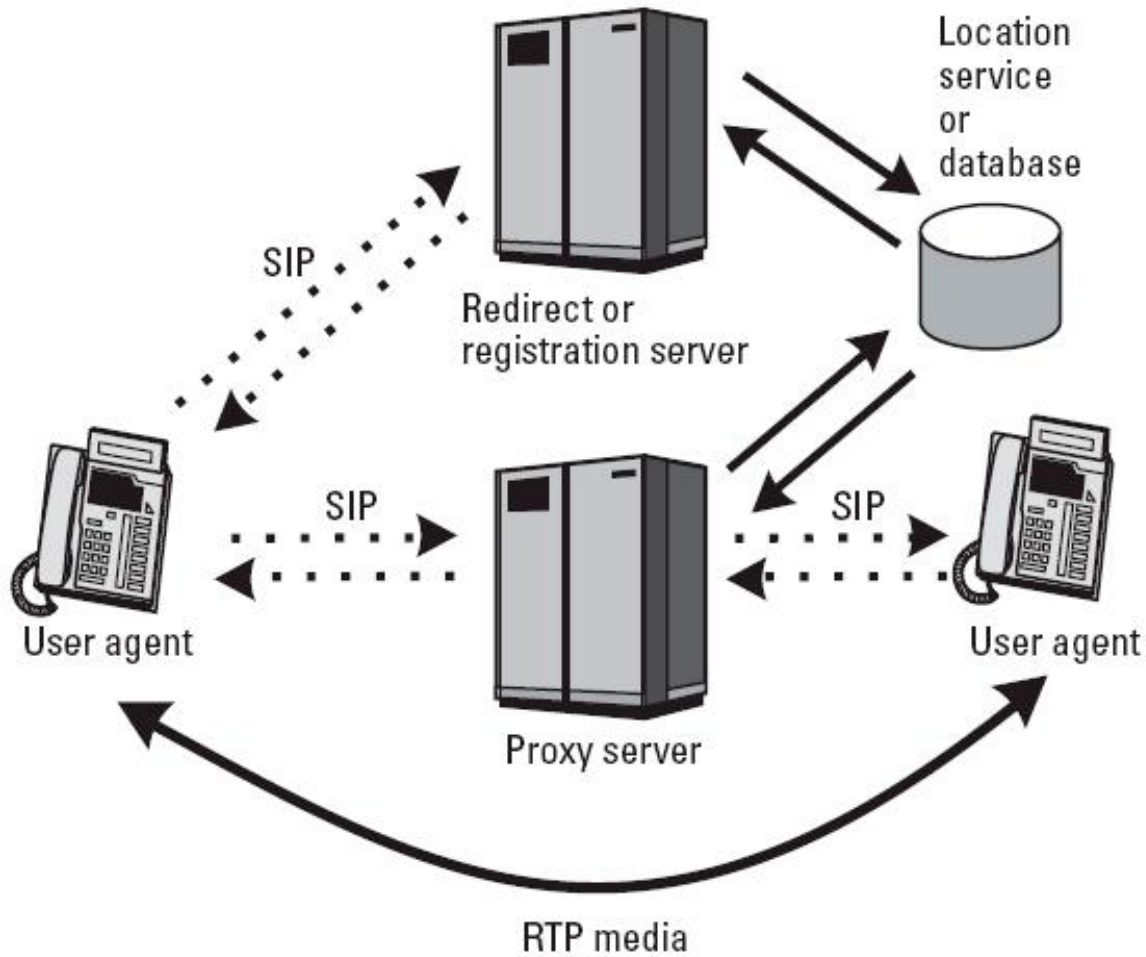
Communication Systems

SIP - Entities

SIP server

- might act as proxy server and could be used for
 - authentication, authorization
 - secure routing and rerouting
- redirect server = information service
- location server is the request address for the host on which a given user might be reached on
- registrar server acts as registration service
 - registers the current location of the clients
 - often at the same place as proxy or redirect
 - is not a required component for SIP, but useful in larger setups

SIP - Server



SIP – Message types

SIP defines messages for communication setup and ending

INVITE	Request to invite a user (called party) to a call
ACK	Acknowledgment to start reliable exchange of invitation messages
BYE	To terminate (or transfer) the call between the two endpoints
OPTIONS	Request to get information about the capabilities of a call
REGISTER	To register information of current location with a SIP registration server
CANCEL	Request to terminate search of a user or "ringing"
INFO	Mid-call information (e.g. ISUP, DTMF)
PRACK	Provisional Acknowledgement
COMET	Pre-condition met
SUBSCRIBE	Request to subscribe to an event
NOTIFY	Notify subscribers

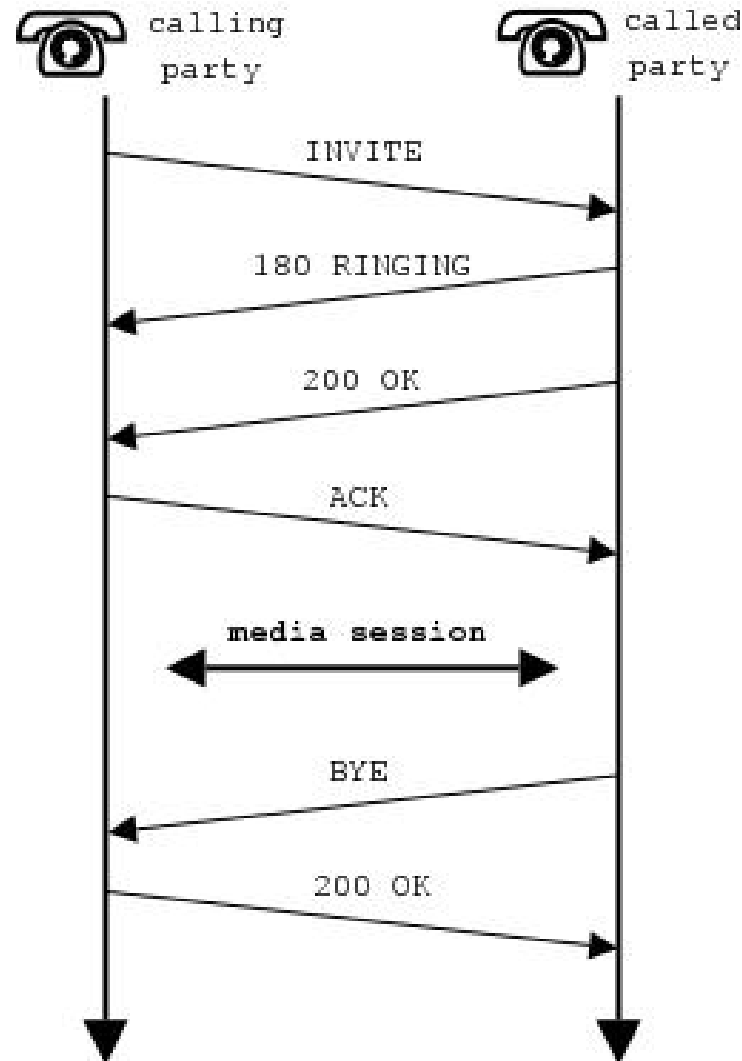
SIP – Direct example session

Direct SIP connection

Disadvantage:

- the calling party has to know the IP address of called party

INVITE message contains the details of which type of session is to be initiated



SIP – Direct example session

The screenshot displays a network analysis tool interface. The main window shows a packet capture of a SIP INVITE message. The packet list at the top shows a single packet at time 0.000000 from source 217.10.79.9 to destination 80.131.231.29, identified as a SIP/SDP Request: INVITE sip:2636639@80.131.231.29:5060, with session description.

The detailed view of the selected packet is as follows:

```

Session Initiation Protocol
  Request-Line: INVITE sip:2636639@80.131.231.29:5060 SIP/2.0
    Method: INVITE
  Message Header
    Record-Route: <sip:2636639@217.10.79.9;ftag=as2c42d209;lr=on>
    Record-Route: <sip:2636639@217.10.79.8;ftag=as2c42d209;lr=on>
    Max-Forwards: 8
    Record-Route: <sip:4920142636639@217.10.79.8;ftag=as2c42d209;lr=on>
    Via: SIP/2.0/UDP 217.10.79.9;branch=z9hG4bKaca7.d1ee4543.0
    Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.ba6faa92.0
    Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.aa6faa92.0
    Via: SIP/2.0/UDP 217.10.64.86:5060;branch=z9hG4bK2affe745
  From: "09119374209" <sip:09119374209@217.10.64.86>;tag=as2c42d209
    SIP from address: "09119374209" <sip:09119374209@217.10.64.86>
    SIP tag: as2c42d209
  To: <sip:4920142636639@sipgate.net>
  Contact: <sip:09119374209@217.10.64.86>
  Call-ID: 098d85e411b1bc0b2bf0900528510b31@217.10.64.86
  CSeq: 102 INVITE
  User-Agent: Asterisk PBX
  Date: Sun, 16 May 2004 16:17:04 GMT
  Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER
  Content-Type: application/sdp
  Content-Length: 255
  Siptgate-Authentication: accepted
  Message body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator: Session Id (s): root 26687 26687 TM IP4 217.10.64.86
  
```

At the bottom, the raw packet data is shown in hexadecimal and ASCII format. The ASCII portion shows the start of the INVITE message: "P... INVITE sip:2636639@80.131.231.29:5060 SIP/2.0..Record-Route: < sip:2636639@217.10.79.9;ftag=as2c42d209;lr=on>".

Communication Systems

SIP – Header fields

Request URI, SIP version number

VIA: SIP version number, protocol, every SIP entity adds host and port, which created or routed the message

Max-Forwards is decremented at every hop

To, From: tags as identifier

Call-ID: sender creates local non-ambiguous identifier which is globally unique in combination with the full qualified domain name

CSeq: command sequence is incremented with every new request

SIP – Optional header fields

More optional fields

- **Contact** contains the SIP address of the current host, if connected over proxy – messages could be sent directly
- **Content-Type** and **Content-Length** tell the style of the following SDP body

SIP – "trying message" (before ringing)

The screenshot shows a Wireshark interface with a packet capture list and a detailed view of a SIP message. The packet list shows:

No.	Time	Source	Destination	Protocol	Info
1	0.000000	217.10.79.9	80.131.231.29	SIP/SDP	Request: INVITE sip:2636639@80.131.231.29:5060, with session description
2	0.005955	80.131.231.29	217.10.79.9	SIP	Status: 100 trying
3	0.008325	80.131.231.29	217.10.79.9	SIP	Status: 180 ringing
4	0.057998	217.10.79.9	80.131.231.29	SIP/SDP	Request: INVITE sip:2636639@80.131.231.29:5060, with session description
5	0.064726	80.131.231.29	217.10.79.9	SIP	Status: 180 ringing
6	1.567673	80.131.231.29	217.5.112.21	ICMP	Echo (ping) request
7	4.552628	80.131.231.29	217.10.79.9	SIP/SDP	Status: 200 OK, with session description

The detailed view of Frame 2 (536 bytes on wire, 536 bytes captured) shows the following SIP message structure:

- Linux cooked capture
- Internet Protocol, Src Addr: 80.131.231.29 (80.131.231.29), Dst Addr: 217.10.79.9 (217.10.79.9)
- User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
- Session Initiation Protocol
 - Status-Line: SIP/2.0 100 trying
 - Status-Code: 100
 - Message Header
 - Via: SIP/2.0/UDP 217.10.79.9;branch=z9hG4bKaca7.dlee4543.0
 - Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.ba6faa92.0
 - Via: SIP/2.0/UDP 217.10.79.8;branch=z9hG4bKaca7.aa6faa92.0
 - Via: SIP/2.0/UDP 217.10.64.86:5060;branch=z9hG4bK2affe745
 - From: "09119374209" <sip:09119374209@217.10.64.86>;tag=as2c42d209
 - SIP from address: "09119374209" <sip:09119374209@217.10.64.86>
 - SIP tag: as2c42d209
 - To: <sip:4920142636639@sipgate.net>
 - Call-ID: 098d85e411b1bc0b2bf0900528510b31@217.10.64.86
 - CSeq: 102 INVITE
 - User-Agent: Grandstream 1.0.4.39
 - Content-Length: 0

The hex dump at the bottom shows the raw bytes of the message, with ASCII characters visible on the right side, including "E...P...", ".0...SIP/", "2.0 100 trying..", "Via: SIP /2.0/UDP", "217.10. 79.9;bra", "nch=z9hG 4bKaca7.", "dlee4543 .0..Via:", "SIP/2.0 /UDP 217", ".10.79.8 ;branch=", "z9hG4hka ca7 ba6f".



SIP – "ringing message"

File Edit View Go Capture Analyze Statistics Help

No.	Time	Source	Destination	Protocol	Info
1	0.000000	80.131.245.242	217.10.79.9	SIP/SDP	Request: INVITE sip:021158006489@sipgate.de, with session description
2	0.133041	217.10.79.9	80.131.245.242	SIP	Status: 100 trying -- your call is important to us
3	0.256306	217.10.79.9	80.131.245.242	SIP	Status: 180 ringing
4	2.525077	80.131.245.242	217.5.112.21	ICMP	Echo (ping) request
5	3.496069	80.131.245.242	217.10.79.9	SIP	Request: CANCEL sip:021158006489@sipgate.de
6	3.602265	217.10.79.9	80.131.245.242	SIP	Status: 200 cancelling
7	3.609524	217.10.79.9	80.131.245.242	SIP	Status: 487 Request cancelled
8	3.613375	80.131.245.242	217.10.79.9	SIP	Request: ACK sip:021158006489@sipgate.de
9	3.720011	217.10.79.9	80.131.245.242	SIP	Status: 487 Request cancelled

Via: SIP/2.0/UDP 10.8.4.20;rport=5060;received=80.131.245.242;branch=z9hG4bK57cdd8e750b2e2ba
 Record-Route: <sip:8006489@217.10.79.9;ftag=a1f109c28a5cd049;lr=on>
 Record-Route: <sip:8006489@217.10.79.8;ftag=a1f109c28a5cd049;lr=on>
 Record-Route: <sip:4921158006489@217.10.79.8;ftag=a1f109c28a5cd049;lr=on>
 Record-Route: <sip:021158006489@217.10.79.9;ftag=a1f109c28a5cd049;lr=on>
 From: "Gerhard Schneider" <sip:2636639@sipgate.de>;tag=a1f109c28a5cd049
 SIP from address: "Gerhard Schneider" <sip:2636639@sipgate.de>

```

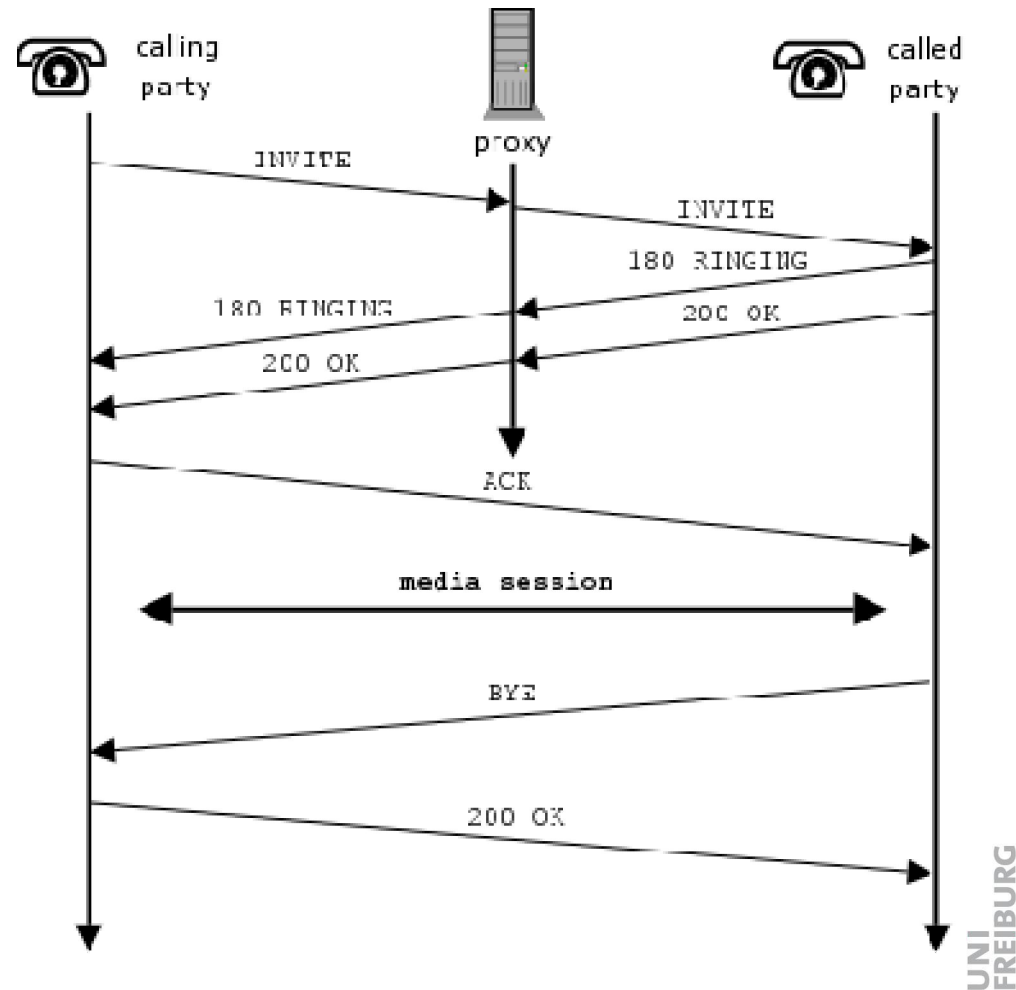
0000  00 00 02 00 00 00 00 00 00 00 00 00 00 08 00  .....
0010  45 10 02 a5 00 00 40 00 3a 11 cf ae d9 0a 4f 09  E.....@.....0.
0020  50 83 f5 f2 13 c4 13 c4 02 91 9e ad 53 49 50 2f  P..... SIP/
0030  32 2e 30 20 31 38 30 20 72 69 6e 67 69 6e 67 0d  2.0 180 ringing.
0040  0a 56 69 61 3a 20 53 49 50 2f 32 2e 30 2f 55 44  .Via: SI P/2.0/UD
0050  50 20 31 30 2e 38 2e 34 2e 32 30 3b 72 70 6f 72  P 10.8.4 .20:rpor
  
```

Filter: + Expression... Clear Apply File: (Untitled) 56. P: 15 D: 15 M: 0

SIP – "ringing" (cont.)

To and **From** fields are the same as in INVITE

- direction of the initiating request is important



SIP – "ringing" (cont.)

Connection over a proxy

- only answers to requests, does not send requests by itself
- no media abilities (does not handle media sessions)
- reads header and does not analyse body+

Proxy may send request for clients location to location server

SIP – OK (200) message

The screenshot displays a network traffic capture in Wireshark. The main focus is on a SIP 200 OK message (packet 7). The packet list pane shows the following entries:

No.	Time	Source	Destination	Protocol	Info
7	4.552620	80.131.231.29	217.10.79.9	SIP/SDP	Status: 200 OK, with session description
8	4.702762	217.10.79.9	80.131.231.29	SIP	Request: ACK sip:2636639@80.131.231.29:5060
9	4.720004	80.131.231.29	217.10.79.9	RTP	Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=23686, Time=4044144421
10	4.739986	80.131.231.29	217.10.79.9	RTP	Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=23687, Time=4044144581
11	4.759968	80.131.231.29	217.10.79.9	RTP	Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=23688, Time=4044144741
12	4.779988	80.131.231.29	217.10.79.9	RTP	Payload type=ITU-T G.711 PCMU, SSRC=901137530, Seq=23689, Time=4044144901

The packet details pane for the selected SIP packet (No. 7) shows the following structure:

- SIP tag: as2c42d209
 - To: <sip:4920142636639@sipgate.net>;tag=9bd267a4bb45229f
 - SIP to address: <sip:4920142636639@sipgate.net>
 - SIP tag: 9bd267a4bb45229f
 - Call-ID: 098d85e411b1bc0b2bf0900528510b31@217.10.64.86
 - CSeq: 102 INVITE
 - User-Agent: Grandstream 1.0.4.39
 - Contact: <sip:2636639@10.8.4.20>
 - Allow: INVITE,ACK,CANCEL,BYE,NOTIFY,REFER,OPTIONS,INFO,SUBSCRIBE
 - Content-Type: application/sdp
 - Content-Length: 140
 - Message body
 - Session Description Protocol
 - Session Description Protocol Version (v): 0
 - Owner/Creator, Session Id (o): 2636639 8000 8000 IN IP4 10.8.4.20
 - Session Name (s): SIP Call
 - Connection Information (c): IN IP4 10.8.4.20
 - Time Description, active time (t): 0 0
 - Media Description, name and address (m): audio 5004 RTP/AVP 0
 - Media Attribute (a): rtpmap:0 PCMU/8000
 - Media Attribute (a):ptime:20

The packet bytes pane at the bottom shows the raw data of the message, including headers and the SDP body.

SIP – Redirect, registering & instant messaging

Redirection

- client sends INVITE to the SIP redirect server
- redirect server sends a request to the location server or requests the IP of the client to call
- current data is sent to the client, which ACK's
- from now on further on like direct connection

Registration

- REGISTER message to SIP registration server
- binding of the SIP URI with IP the users client/machine
- 200 OK

SIP – Redirect, registering & instant messaging

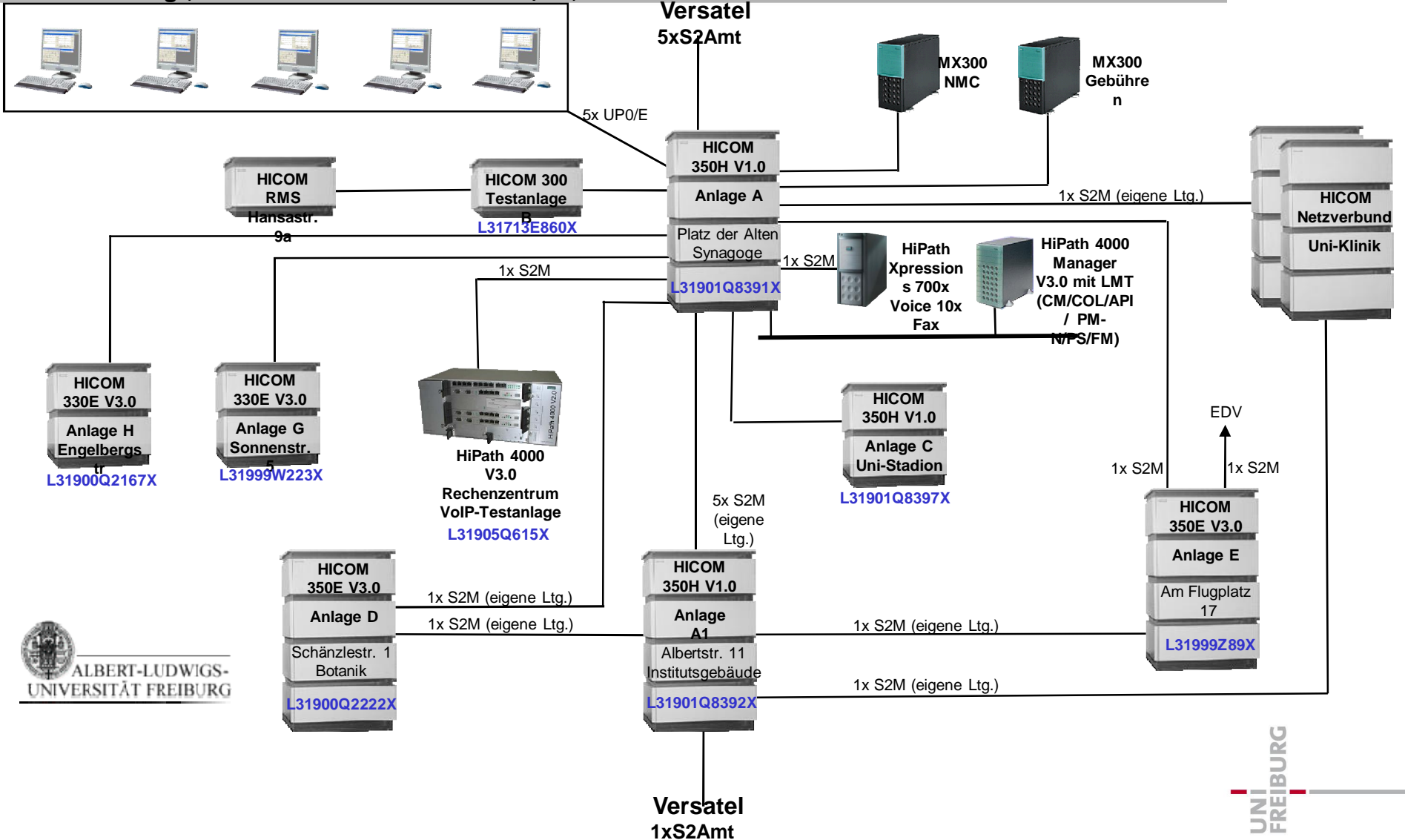
Instant messaging like the wellknown tools in that sector

- online status, buddy lists ...

- Let us introduce a VoIP network into an existing environment
- How? Finances? Results?

Classical telephone since 1994

Vermittlung (5x AC WIN XP, davon 1x Blindenarbeitsplatz)



Available infrastructure (Freiburg)

- WiFi everywhere (literally) – access to digital data (library) seamlessly possible
 - Some 700 access points
 - Uniform access on university premises + other institutions
 - Roaming: statewide access, nationwide access, eduroam
- Rektorate knows about IT (via CIO)
- Classical telephony crew merged into computer centre
 - Even trade union considered this to be a good idea
 - Controlled refocus to new technology (under guidance)
 - VoIP is primarily an issue for data networks
 - And stays a „take care of a telephone user“ issue (i.e. know how required!)

VoIP – typical arguments from the past

- **Expensive operation** of two infrastructures (data, voice)
 - The classical world even requires additional external service personel
- **Internal staff training problem**
 - Technicians for old technology – what if the old technology vanishes? No “hire&fire”
- **Technical idea is “too easy”** and does not reflect the necessities for speech
 - Session Initiation Protokoll → “ringing”
 - Breakout to classical network
 - Mapping of IP-adresses and phone numbers
- No “classical telephone system” required.

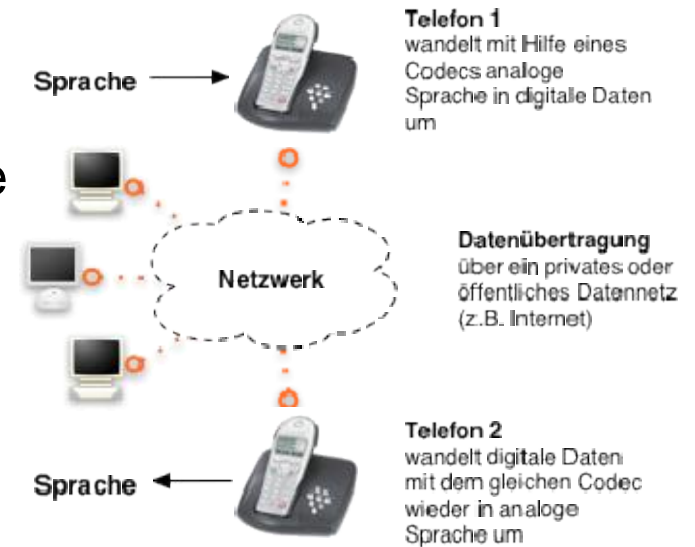
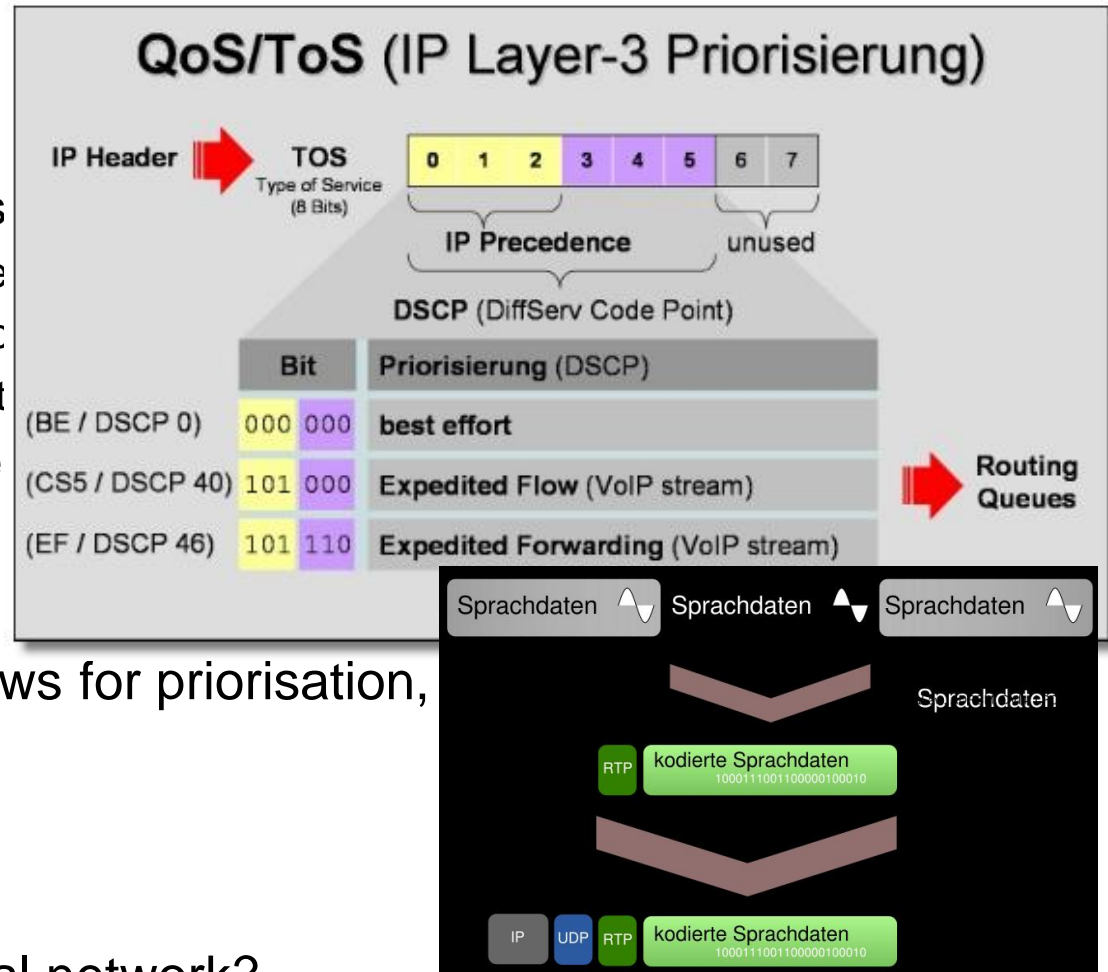


Bild: wikipedia

VoIP – technical arguments from the past

- A sensitive service cannot
 - Best effort = no SLA poss
 - No bandwidth guarantee
 - No guarantee for turnarc
 - No guarantee against jitt
 - The human ear only acce
- If VoIP then you may have
- RTP/SRTP over UDP allows for prioritisation, LAN (if at all)
- Emergency calls?
- localisation?
- Connectivity to the classical network?
 - Ample chances for discussion during the selection process



VoIP at Freiburg University

our way of introducing a new service

- New building for the university: ZBSA
 - 69.000 € to dig up the botanical garden to install classical telephone cables (no alternative due to length restrictions)
 - Local telephone system could not easily expanded (all slots full)
 - The IT centre (no in charge of telephony) decided to have VoIP in that building – no length restrictions for data fibre optics
- We explicitly did not think of Siemens
 - Hicom 300 nightmares...
- Open standards as priority
 - No alternative if you teach SIP and open standards in class
 - Who believes you if you do not follow your own preaching?



Selection process ☺

Presentation of HiPath8000 showed:

- Surprise: absolut SIP conformity
- Telephone system: just an IBM-PC with software
 - Ok – no real surprise
- Independent VoIP↔ISDN breakout
 - Breakout feels like a VoIP-phone, operates without HiPath8000
- Official phones run smoothly on Asterisk
- Billing can still be done by the old system (while available)
- „special features“ realisable via SIP realisiert (CheSe)

System can easily be replaced (if SEN goes broke, or....)

- Worst-case-scenario showed: no risk
- Instead of digging: buy a HiPath8000, it is cheaper and more modern
 - Well, cheaper at first...

Money \$\$~~€€€~~



OpenStage 20

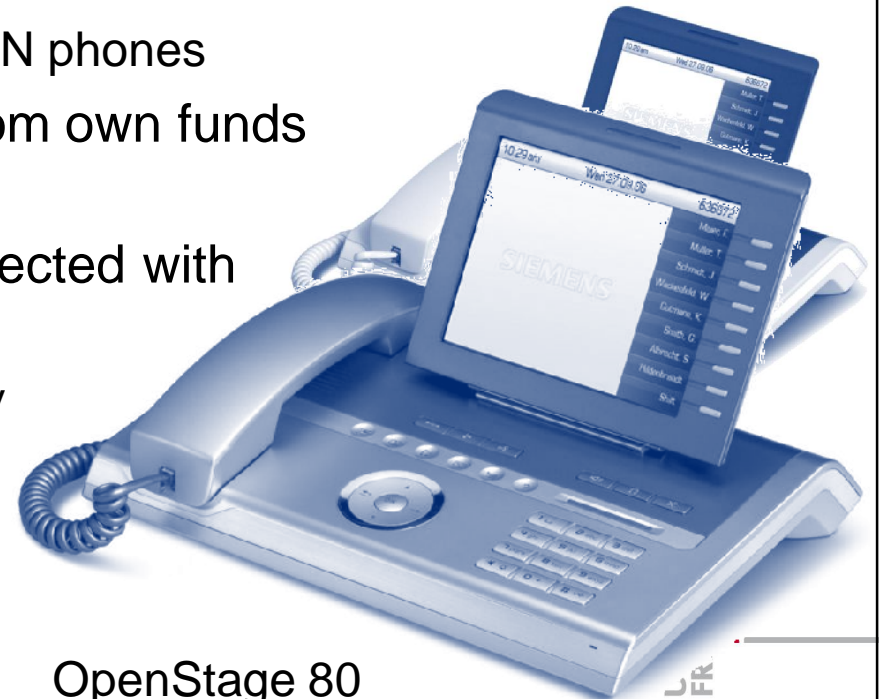


Openstage 40

Telephones cost money ☹️

Definition by the Rektorate:

- Replace defunct old phones with old ones
- New phones (40/60) only for new staff
 - Phones are cheaper than the ISDN phones
- Anybody may buy new phones from own funds
 - publish the price tag
- Non-system phones may be connected with a service charge of 100€
 - Great – no real system dependency
- Major problem:



OpenStage 80



Money – and experiences

- The (new) assistant gets a new phone 😊
 - His/her professor asks for rearrangement of all phone connections so that he gets the new phone ☹️ -> work
- In the natural sciences people are „happy“ to spend their money on the new phones.
- Little interest for other phones
we can offer central maintenance only for system phones – not seen as problem
 - The view is different for computers 😊
- Plans: slowly phase out the old phones
 - Perhaps the majority will be replaced by 2015

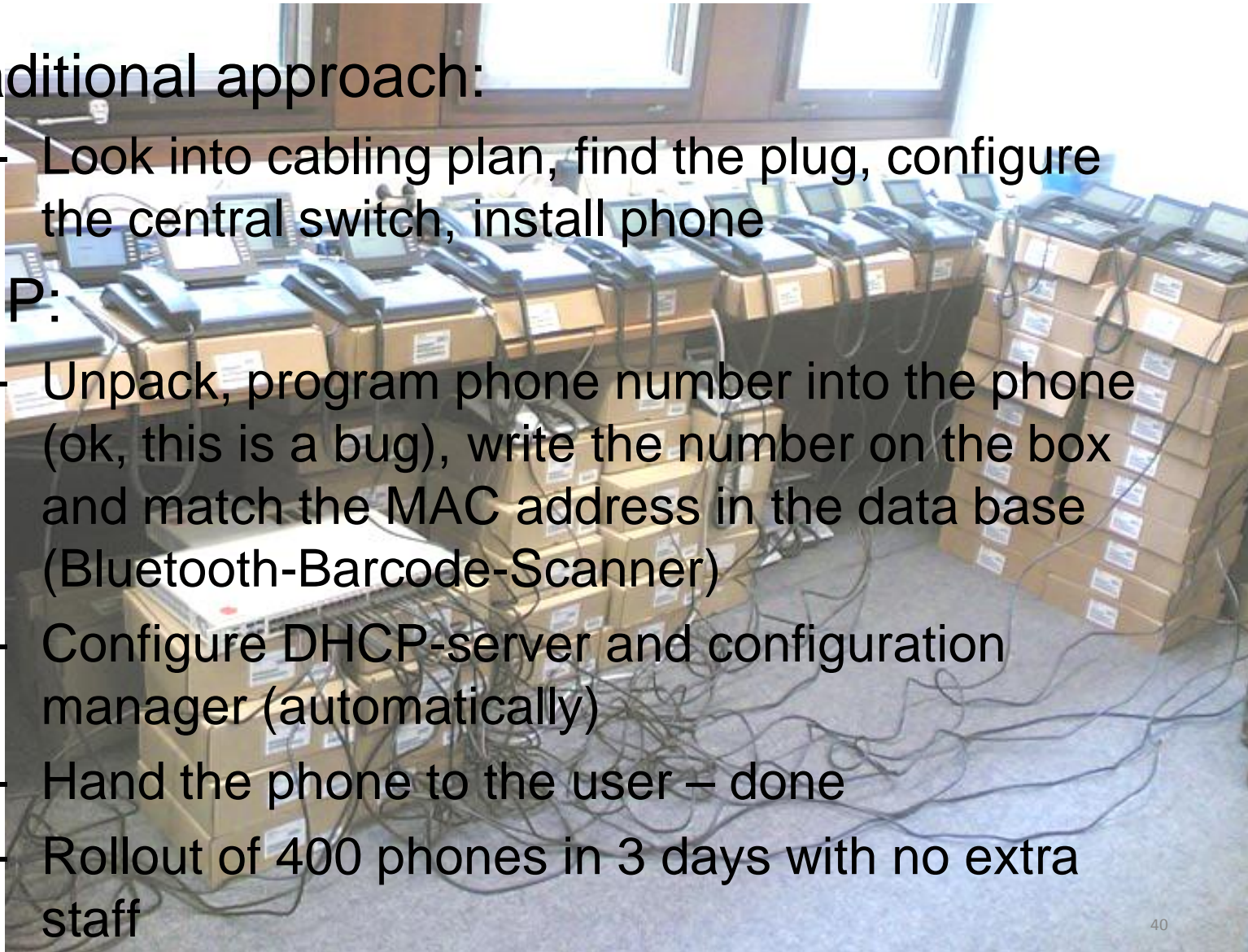
Rollout – experiences of a different kind

Traditional approach:

- Look into cabling plan, find the plug, configure the central switch, install phone

VoIP:

- Unpack, program phone number into the phone (ok, this is a bug), write the number on the box and match the MAC address in the data base (Bluetooth-Barcode-Scanner)
- Configure DHCP-server and configuration manager (automatically)
- Hand the phone to the user – done
- Rollout of 400 phones in 3 days with no extra staff



Energy issues

VoIP-phones need electricity

- Others don't?
- What is required by an old W48-dialer, intelligent phones, DECT, ISDN-phones, UP0-phones at Hicom??



SEN reacted to our complaints

Freiburg = Green City, green mayor reelected

New firmware with a reasonable deep sleep mode

- Instead of 168h per week the phone is now only some 50h „fully awake“

Effects on buildings

PoE really necessary

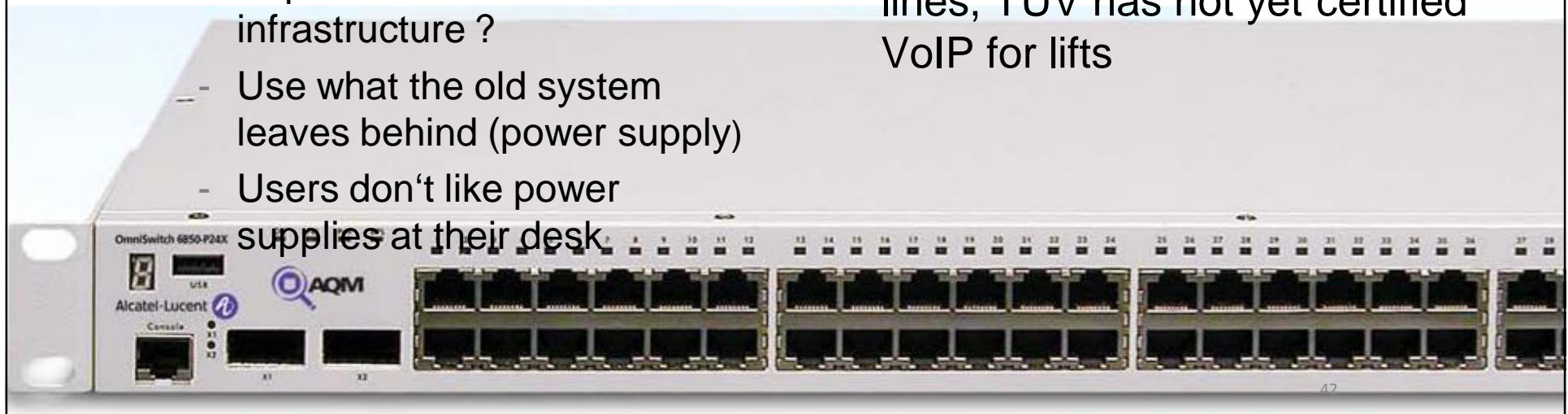
- Now there are affordable solutions
 - We use: Alcatel-Lucent
- UPS in each switching centre?
 - 150 UPS??
Costly in terms of purchase and maintenance
 - Separate/additional electrical infrastructure ?
 - Use what the old system leaves behind (power supply)
 - Users don't like power supplies at their desk.

Emergency phones (lift,.....) still remain on the old system

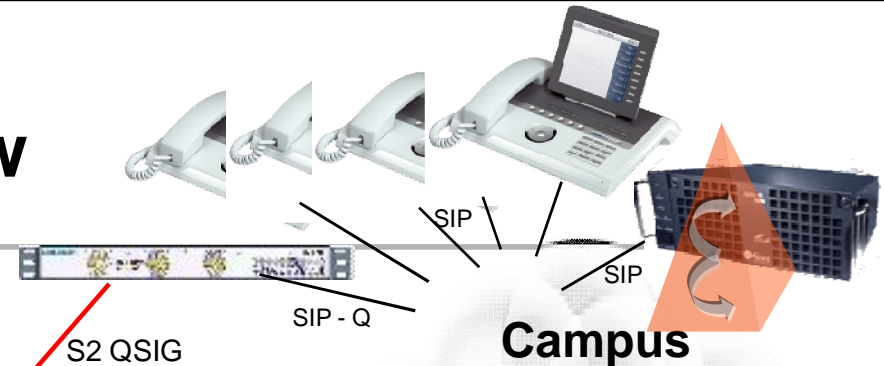
- Coward or smart guy?
 - Rome not built in one day
- GSM: possible backup structure?

Unexpected problem:

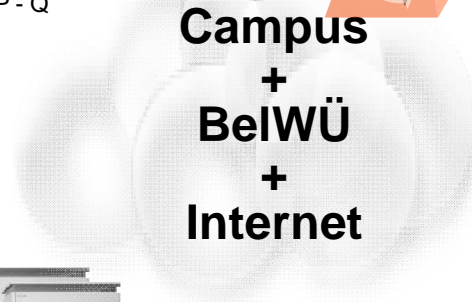
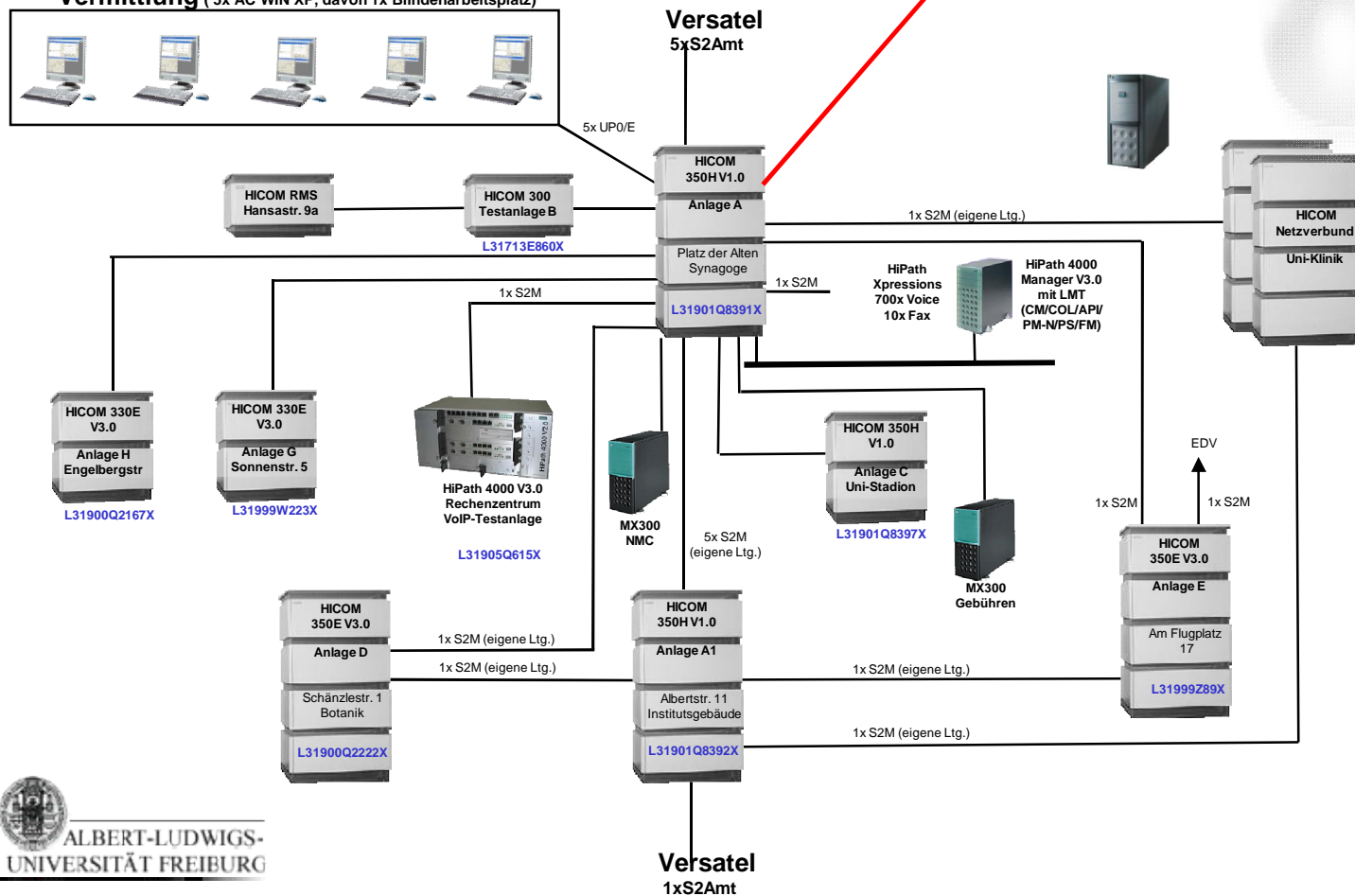
Telekom abandons analogue lines, TÜV has not yet certified VoIP for lifts



Integrating old and new



Vermittlung (5x AC WIN XP, davon 1x Blindenarbeitsplatz)



Successfully connected devices

Nokia
E65



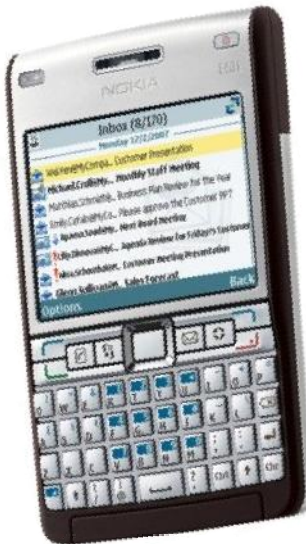
Siemens
WL2



AVM 7270



Nokia
E61/E61i



**Just a small
selection!**

Loox T830



Successfully connected devices (2)

Integration of the video conference system

- Lifesize-System with SIP
- just call the phone number it has on the H8K
- the boxes negotiate the right capabilities
- VoIP-Videophones may now take part in a video conference



Other aspects

Trade union (Personalrat)

- Real concerns!
- VoIP-communication may now be encrypted
- Phone can be secured via PIN
 - Unauthorised people can no longer read the history
 - Features can be switched on centrally, with local override
 - Great advantage over old system
- Integrated in all major decisions
 - RZ-staff has union representatives... 😊

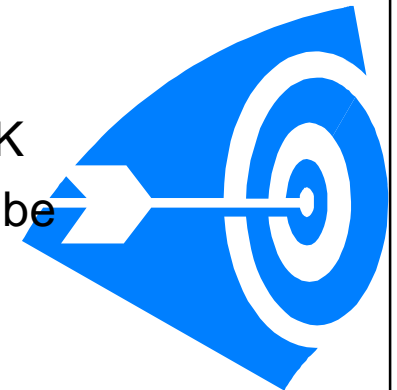
- Evolutionary training of „classical“ staff ensured
- No problem....

Universität

- Report to the senate early!
- Give VoIP to VIPs who have influence
 - Including 2x BVerfG

phones

- Are independent of H8K
- Mixture of devices can be good and bad



Projects – thanks to VoIP



University „work at home“

- Up to 25% of your work time (GenderMainstreaming)
- Enforce clear regulations when critical data is used
 - If data is very critical, the university will provide the hardware
 - bw-pc with cryptobox, configuration via IT-centre
 - User provides internet connectivity
- H8K can serve up to 5 phone with one number
 - No more call rerouting, just connect another (soft)phone

Beeper system → WLAN

- Use the money for the beeper system to expand WiFi
now you can call maintenance staff directly via VoIPoWLAN
 - GSM won't work in the cellar
- Technician may take a remote control camera with him
 - For emergencies – have yourself monitored by a friend
- Localisation of users

Other plans

- Configuration of personal data in myaccount
 - Part of our Identity-Management-Project
 - Users may enter:
 - What should be published
 - What should be shown in the phone display
 - Phone needs LDAP-access 😊
- VoIP coupling with other universities
- Use the multi provider functionalities as service for other institutions
- Configure phones for at least two SIP-providers
 - Simplifies charging of private calls

Takes longer than expected – due to internal civil service structures

VoIP projects

Why do we still need phones?

- Use soft phones!
- At last a device under the control of the IT centre
- Display a motd
 - Be careful – do not SPAM!
- Public loudspeaker system
 - Chemical accidents, power failures,

Novel usage of the „device“:

- Exchange an electronic business card
- Quick email check

XML-Platform example

Forms:

10:45 Wed 09/05/07 4300

Example Applications Settings 3400

Form String Lucy Takeshi

Options Select →

Numeric Text 123456

Any Text abc123

A Flower 

Mobility Shift

10:48 Wed 09/05/07 4300

Example Applications Settings 3400

ChoiceGroup Lucy Takeshi

Options Select →

Choice1 Option 1-1
 Option 1-2
 Option 1-3

Choice2 Option 2-1
 Option 2-2

Mobility Shift

Dropdown lists

12:17 Tue 13.11.07 4300

Examples Applications 3400

Implicit List Lucy Takeshi

Options

Item 1

Item 2 →

Item 3

Mobility Shift

9:50 Wed 09/05/07 4300

Example Applications Settings 3400

Confirmation Alert Lucy Takeshi

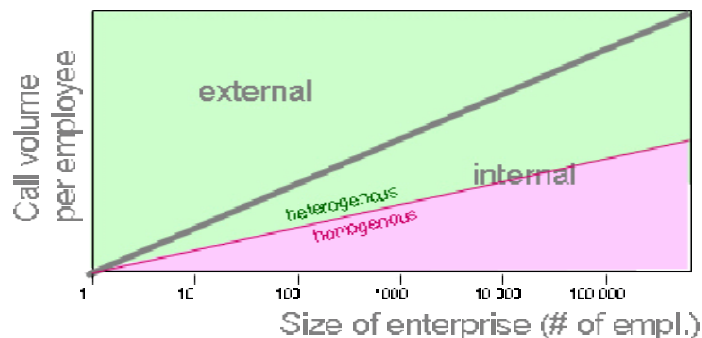
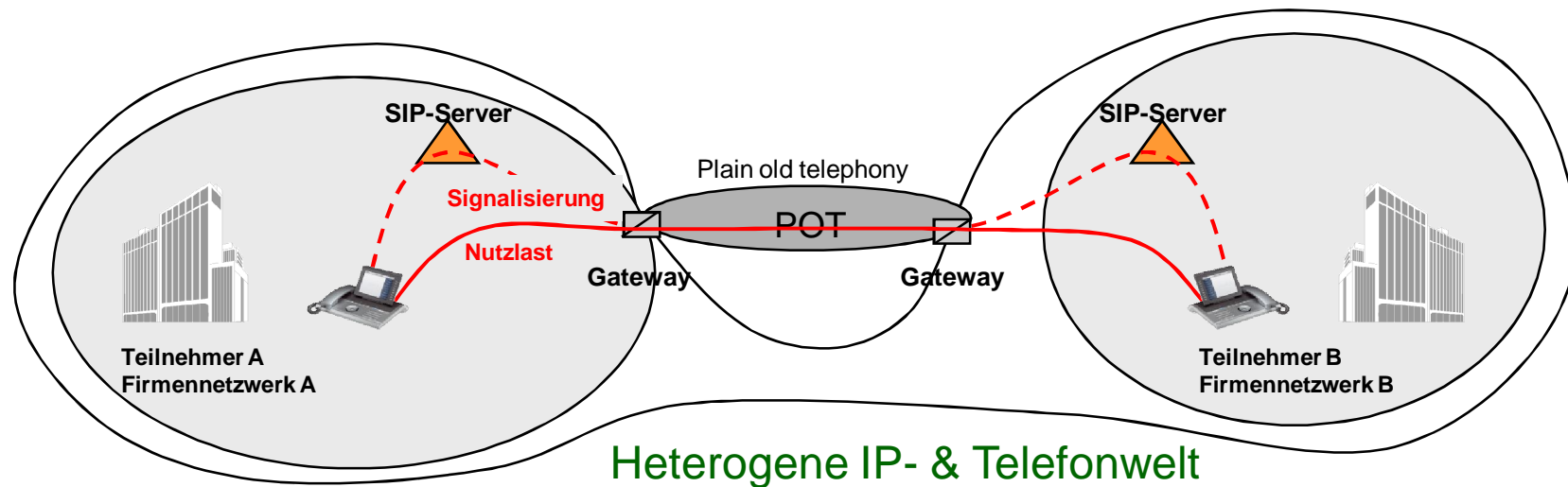
Options Alert

Press "Alert": An alert will appear here:

Mobility Shift

Exchange of electronic business card while making phone calls

Challenge: heterogeneous network infrastructure



0 – 50 % of connections
50 – 100 % of connections

Summary of projects:

- Can use the phone for quick email checks
 - Energy saving way of soothing your mind ☺
- Vcard can be exchanged over any network
- Proof that phones do offer interesting added values
 - As we know from mobile phones and smartphones
- ... gadgets to attract students
 - And thus later members of staff

General summary

- Introduction of VoIP can be done step by step
 - If you have a clear aim
 - If the people in charge work for the system
 - And do not stick to the written plan for the sake of sticking to the plan – but are open to target improvements
 - Collect new ideas on the way
- **The dependence on the manufacturer must be minimised through adherence to open standards**
- **Then we can offer a reliable cloud service**
 - At least in our own mini-cloud, where we own everything, including the infrastructure