Voice over IP (VoIP)

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Introduction

- Student since 1999 in the European Study Course “Mechatronics” at the Niederrhein University of Applied Sciences in Krefeld (Germany) and the Fontys University of Professional Education in Venlo (The Netherlands)
- Practical trainee abroad from September 2001 till February this year at the Hungarian telephony-provider Matáv (subsidiary of Deutsche Telekom)
- This presentation has been hold already at 18C3

The basis of Voice over IP

Part I

Table of contents Part I

Wherefore Voice over IP?

- Use IP to carry both control and media streams
- Enable integration of voice and data networks
- Replaces traditonal telephone network
- Offers well-known telephony services
  - simple phone calls, conference calls
  - call transfer, call waiting, call hold, consultation
- Seamless transition to (new) multimedia services

Protocols

- MGCP and MEGACO/H.248
- H.323
- SIP

MGCP and MEGACO/H.248

by the *telco community*

- for SS7/VoIP integration
- master/slave technology
- similar to digital PBX

H.323

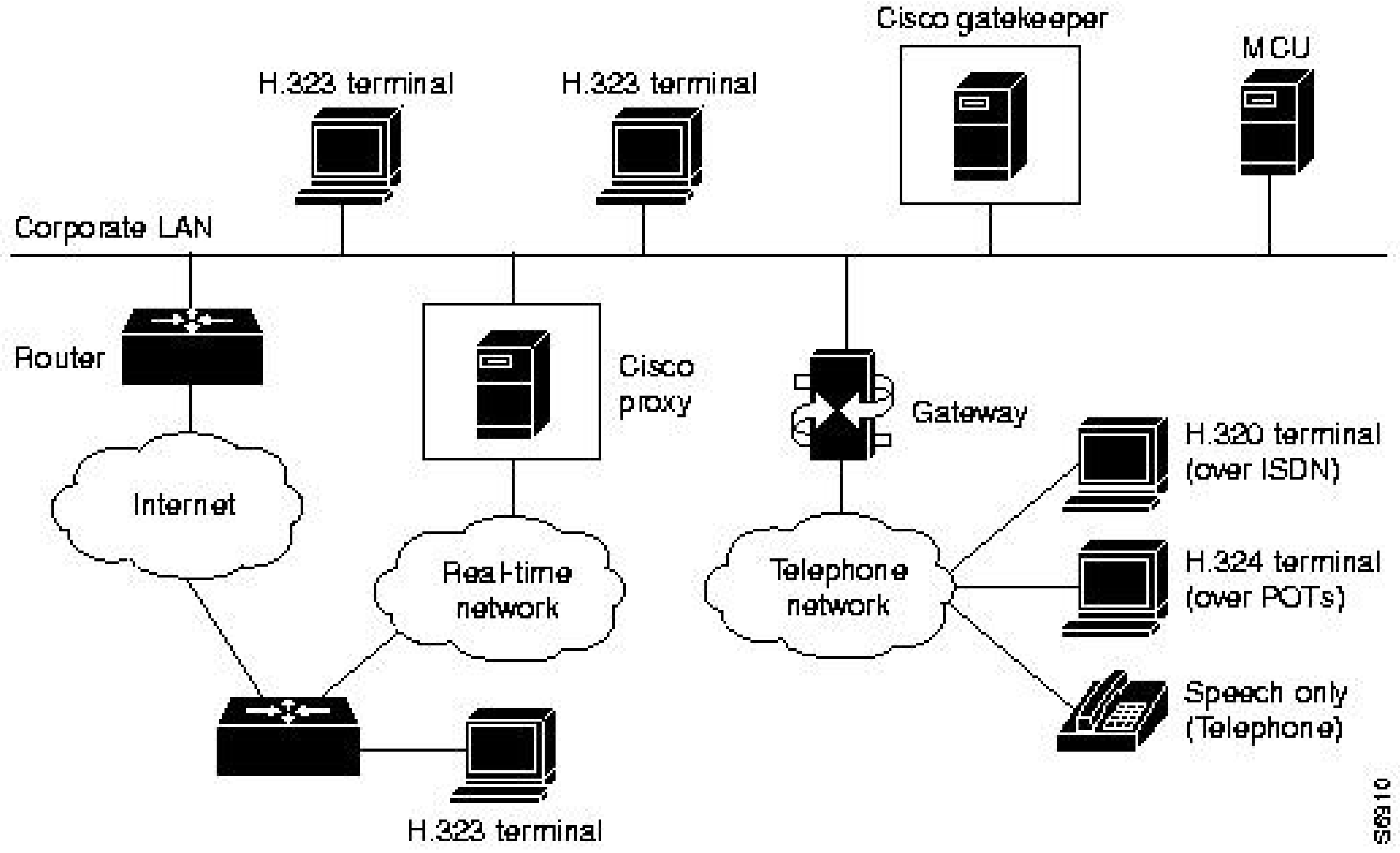
- Telecommunication Standardization Sector of ITU[a] (ITU-T): *Packet-based multimedia communications systems – ITU-T Recommondation H.323*
- working draft started in 1995
- approved in 1996
- actual version is 4 (H.323v4)
- philosophy: “H.323 was designed with a good understanding of the requirements for multimedia communication over IP networks, including audio, video, and data conferencing. It defines an entire, unified system for performing these functions, leveraging the strengths of the IETF and ITU-T protocols.”

[a] International Telecommunication Union

Devices

- Terminal
- Gateway
- Gatekeeper
- MultipointControlerUnit (MCU)

Example-Network



Terminal

description

- (user)-endpoint
- provides real-time two-way communications
- mandatory: voice streaming
- optional: video and data streaming

Terminal

supported protocols

- H.323 (of course ☺)
- H.245 (control channel usage & capabilities)
- Q.931 (call setup & signalling)
- RAS (for use with Gatekeepers / Registration/Admission/Status)
- RTP/RTCP (sequence audio & status video packets)

Gateway

task: translation

- $\leftrightarrow$ SCN[a] e.g. ISDN
- $\leftrightarrow$ PBM[b] e.g. LAN
- H.245 $\longrightarrow$ H.221 (ISDN-conferences)
- H.245 $\longrightarrow$ H.242 (audiovisual terminals)
- audio codecs
- video codecs

[a] Switched Circuit Network
[b] Packet Based Network

Gatekeeper

task: user information / “name server”

- it is optional but essential
- managing communications
- address translation
- call control
- routing services
- system management
- security policies

Gatekeeper

registered endpoints

- H.323-Terminals
- Gateways
- MCUs

MCU

task: maintain all audio, video data & control streams

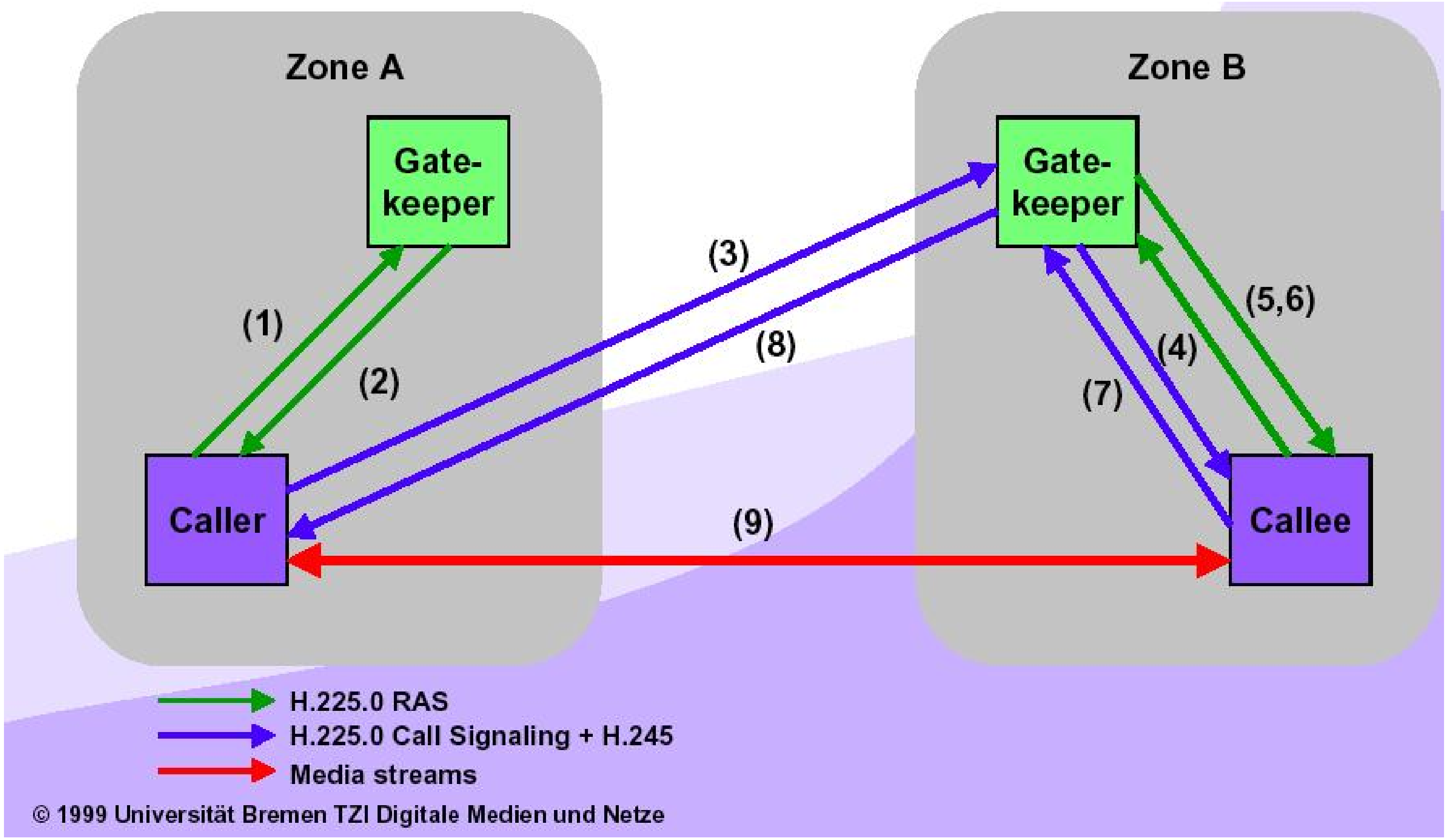
mandatory for conferences

an MCU is usually splitted into two devices:

- MC (Multipoint Controler)
- MP (Multipoint Processor)

usually they are located inside the Gateway or Gatekeeper

Call example



SIP

- Internet Engineering Task Force: *SIP: Session Initiation Protocol*, RFC 2543
- draft in 1996
- approved in 1999
- philosophy: “SIP was designed to setup a “session” between two points. It has a loose concept of a call (that being a “session” with media streams), has no support for multimedia conferencing, and the integration of sometimes disparate standards is largely left up to each vendor.”

Ancestors

- HTTP (basic request/response format, status codes, etc.)
- email/SMTP (addressing style)
- URL (addresses)
- TCP/UDP (message transport)
- RTP (voice session)

Client

Terminal act as UserAgentServer (UAS) or UserAgentClient (UAC); could be soft-phones or IP-phones

Gateways primary task: translation – see H.323-Gateway

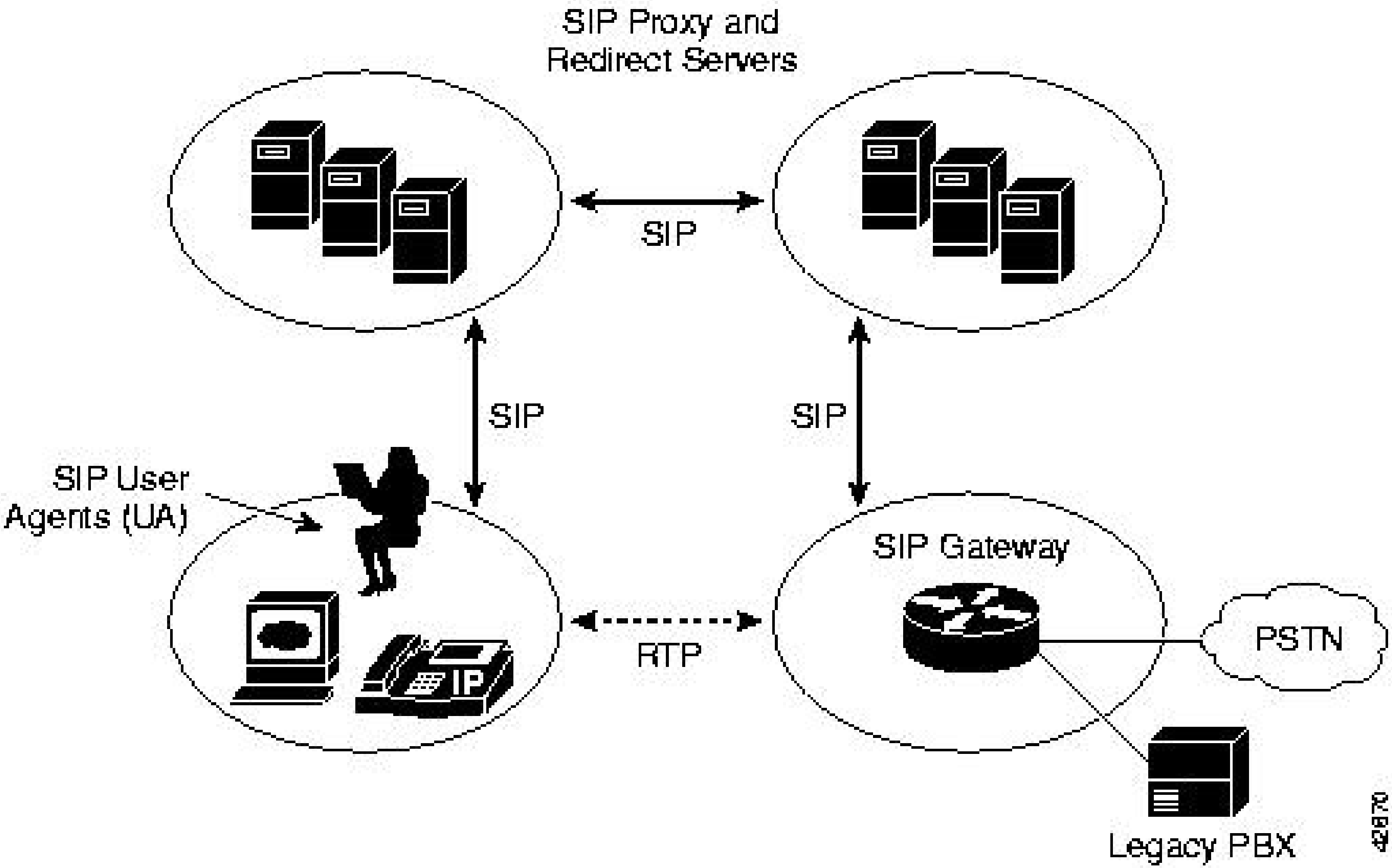
Server

Proxy server receives SIP messages and forward them to the next SIP server in the network

Redirect server provides information about the next hop

Registrar server requests current location for UAC registration; they are often co-located with a redirect or proxy server

Example-Network



Example for SIP messages

The commands that SIP uses are called “methods”:

SIP method	Description
`INVITE`	Invites a user to a call
`ACK`	Used to facilitate reliable message exchange for INVITEs
`BYE`	Terminates a connection between users or declines a call
`CANCEL`	Terminates a request or search for a user
`OPTIONS`	Requests information about server capabilities
`REGISTER`	Registers a user’s current location
`INFO`	Used for mid-session signalling

SIP server responses:

`1xx`	informational (e.g. 100 Trying; 180 Ringing)
`2xx`	successful (e.g. 200 OK; 202 Accepted)
`3xx`	redirection (e.g. 302 Moved temporarily)
`4xx`	request failure (e.g. 404 Not found; 482 Loop detected)
`5xx`	server failure (e.g. 501 Not implemented)
`6xx`	global failure (e.g. 600 Busy everywhere)

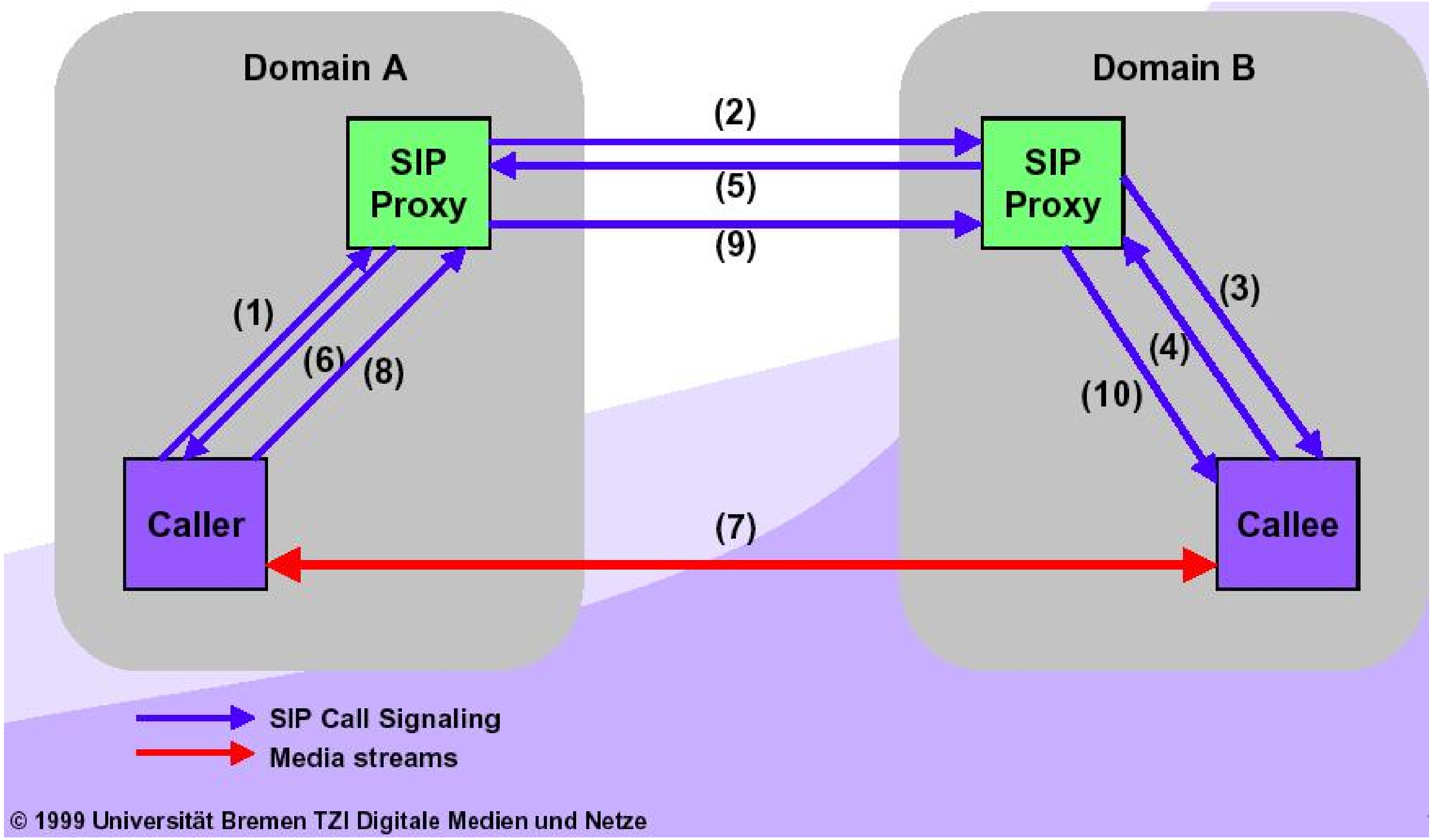
Example of a SIP message

```
INVITE sip:watson@boston.bell-telephone.com SIP/2.0
Via: SIP/2.0/UDP 169.130.12.5
Authorization: PGP version=5.0, signature=...
From: A. Bell <sip:a.g.bell@bell-telephone.com>;tag=7abm
To: T. A. Watson <sip:watson@bell-telephone.com>
Call-ID: 187602141351@worcester.bell-telephone.com
Subject: Mr. Watson, come here.
Content-Type: application/sdp
Content-Length: ...

v=0
o=bell 53655765 2353687637 IN IP4 128.3.4.5
s=Mr. Watson, come here.
t=0 0\\
c=IN IP4 135.180.144.94
m=audio 3456 RTP/AVP 0 3 4 5
```

taken out of RfC 2543

Call example



H.323 vs SIP – comparision

philosophy

H.323	*SIP*
“Old World”; complex; vertical	“New World”; a relative of Internet protocols; simple; open; horizontal
ITU-T	IETF

characteristics

H.323	*SIP*
H.323 specifies everything for the media	A simple toolkit upon which smart clients and applications can be built. It re-uses elements of the net (URL, MIME, DNS)
Assumes reliability of network	Leaves issues of reliability to underlying network
Binary format messages	SIP-Messages formatted as text
Addressing scheme does not scale well (E.164: *111567*; H.323-Identities: *lars*; URL; transport address etc.)	Hierarchical URL-style addressing scheme (*sip:lars@bart.dev.matav.net*)
Possibilities of delay (FastCall)	Minimal delay; simpliefied signalling scheme makes it faster

services

H.323	*SIP*
Can not fork calls	Ability to fork calls
Bewildering messages	Unified messages
Can not mix media within a session	Ability to mix media
Operability with SS7 is complex	Works smoothly with media gateways
Troubles in connecting calls to and from PSTN	Can act with a prior list for multiple protocols

status

H.323	*SIP*
Popularity due to the fact that it was the first set of agreed-upon standards	Industry supported
The majority of existing IP telephony products rely on the H.323 suite	Many vendors are developing products

Conclusion

H.323 is a fairly stable standard – you can buy a lot of components, there are a lot of PC-clients (like NetMeeting), some networks are deployed and scalability is implemented. But always remember that it is a conversion of PSTN to IP-networks. Due to the fact that SIP depends on the actual internet applications, you can call it really Voice over IP. The current developments of SIP show by now that this protocol will give you more freedom to implement features for the work, in the household or during the leisure time, with the machine, you originally called telephone.

Summary Part I

- Wherefore Voice over IP?
- MGCP and MEGACO/H.248
- H.323
- SIP
- Comparision H.323 vs. SIP

(Short) Break?

Experiences with H.323-applications

Part II

Table of contents Part II

- Terminals
- OpenH323-Gatekeeper
- isdn2h323-Gateway
- Example network

Terminals

- NetMeeting
- GnomeMeeting
- OhPhone

NetMeeting

- *The* VoIP Terminal
- Delivered with MS-Windows
- Can access both gatekeepers and gateways
- A lot of non-H.323-features (chat, file-transfer)
- Incompatible to some other terminals
- GSM-codec not implemented

GnomeMeeting

- A GNOME[a] application
- Free software on Linux
- Easy configuration with a lot of parameters
- Depends on the Portable Windows- and OpenH323-Library
- Works fine with gatekeepers and gateways

[a] The GNU Network Object Model Environment

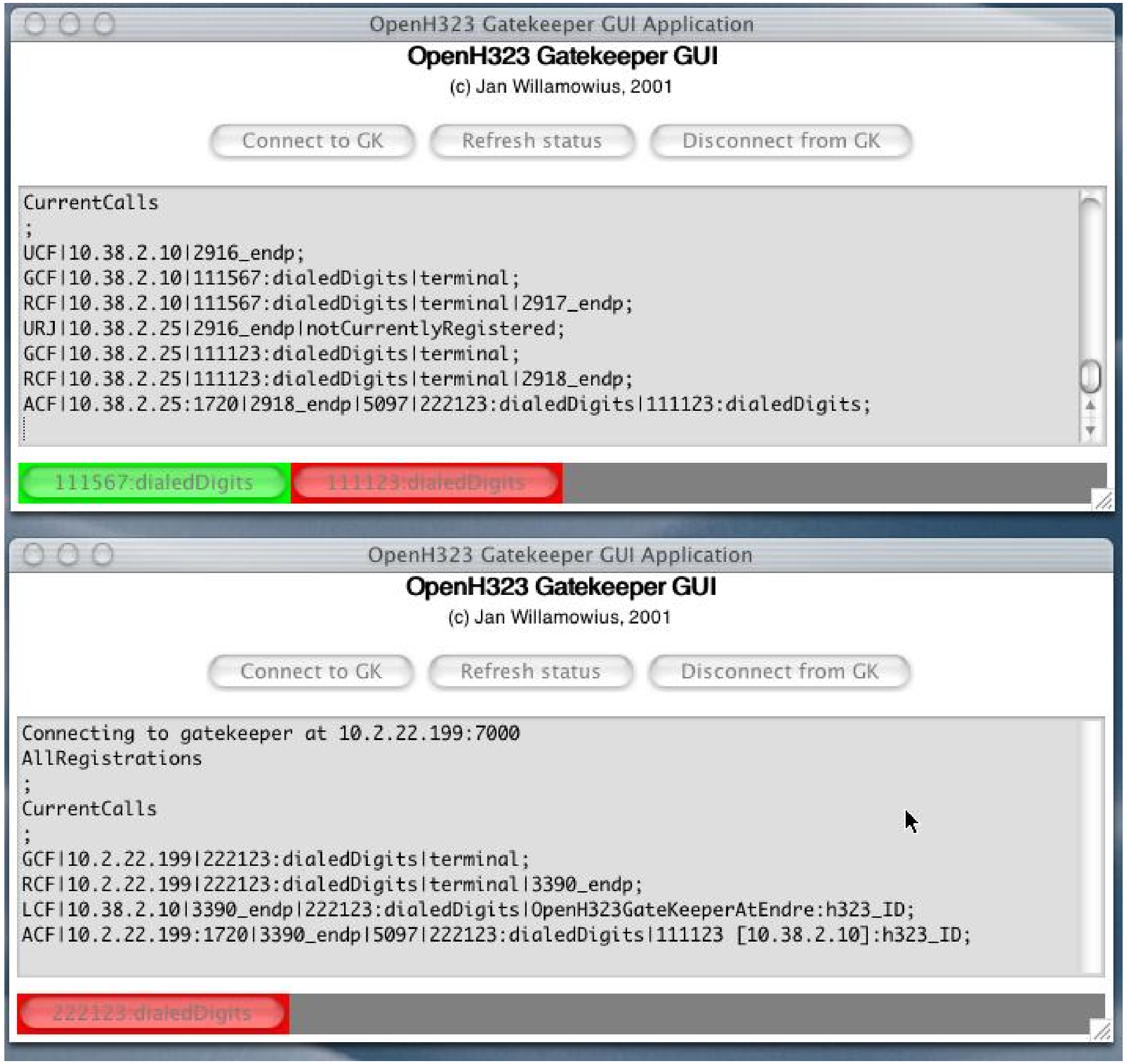
OhPhone

- Console application with video support
- Runs on at least on Linux and Windows
- Depends on the PWLib and OpenH323-Library
- Developed by the *Equivalence Ltd. Pty.* (http://www.equival.com.au/)
- Fast but a little bit tricky to use
- Interoperates with almost every other terminal-application

OpenH323-Gatekeeper

- Part of the *OpenH323-Project* (http://www.openh323.org/)
- Written by Jan Willamowius (http://www.willamowius.com/openh323gk.html)
- It is free software depending on the PWLib and OpenH323-Library
- Highly configurable with an ini-file
- Graphical user interface (GkGUI) for accessing the status port

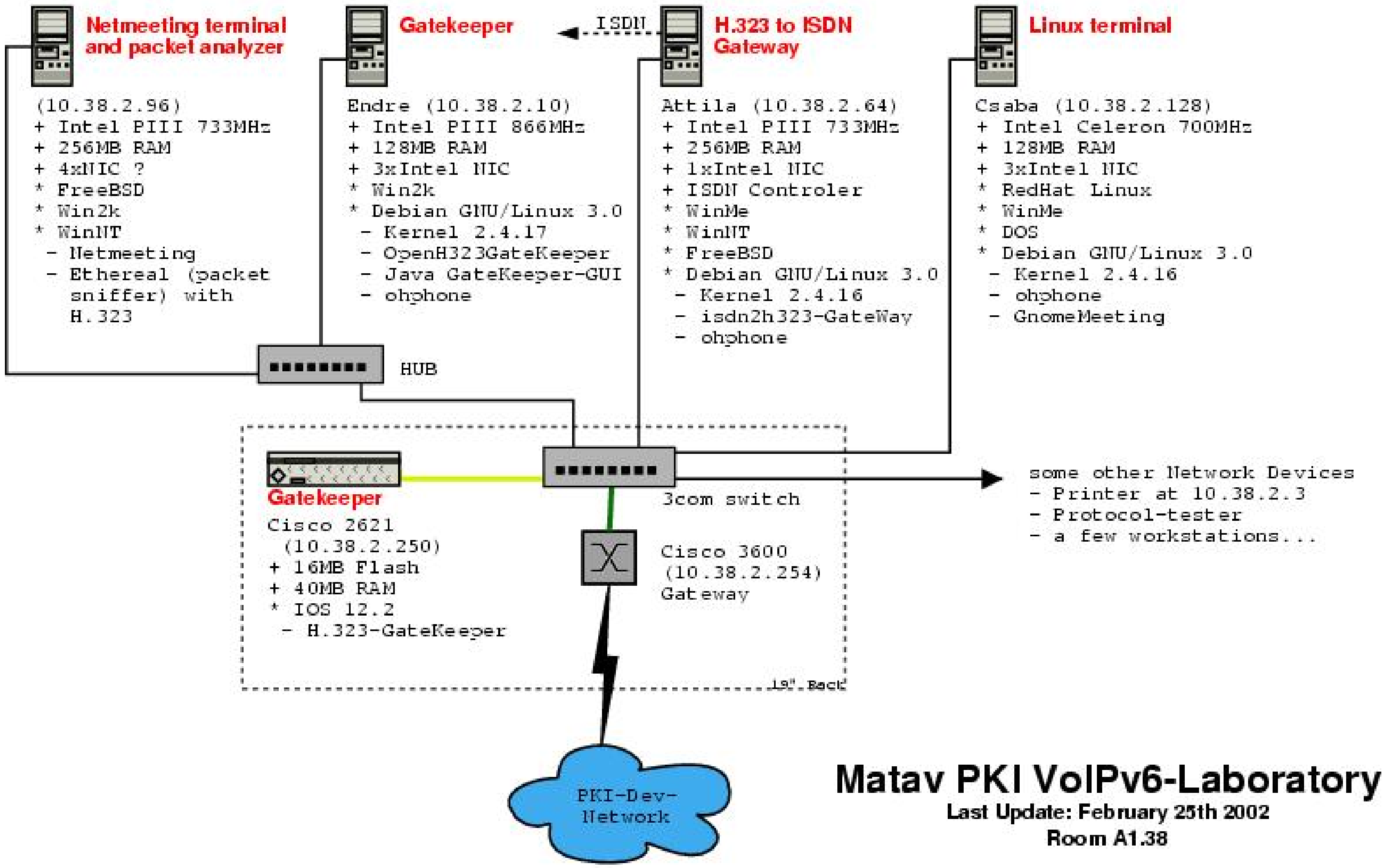
Gatekeeper-GUI



isdn2h323-Gateway

- Allows routing calls from the H.323-LAN to the ISDN and vice versa (with forwarding MSN to defined terminals)
- Distributed by *telos EDV Systementwicklung GmbH* (http://telos.de)
- Free Software using the PWLib and OpenH323-Library
- Good cooperation with the OpenH323-Gatekeeper and GnomeMeeting

Example network



Conclusion

All told it is possible building up a fine working H.323-network with mostly usual Personal Computers which results in a devolvement of a todays PSTN with some more features.

Summary Part II

- Terminals
- OpenH323-Gatekeeper
- isdn2h323-Gateway

Questions?

Or write me an email: `pylon@chaosdorf.de`

This presentation on my homepage:

`http://www.chaosdorf.de/˜pylon/VoIP-chaosdorf.pdf`

Thank you!