Communication Networks II

Multimedia Communications /QoS

Specific Topics:

Internet and Intranet Telephony Protocols (H.323, SIP, MGCP)

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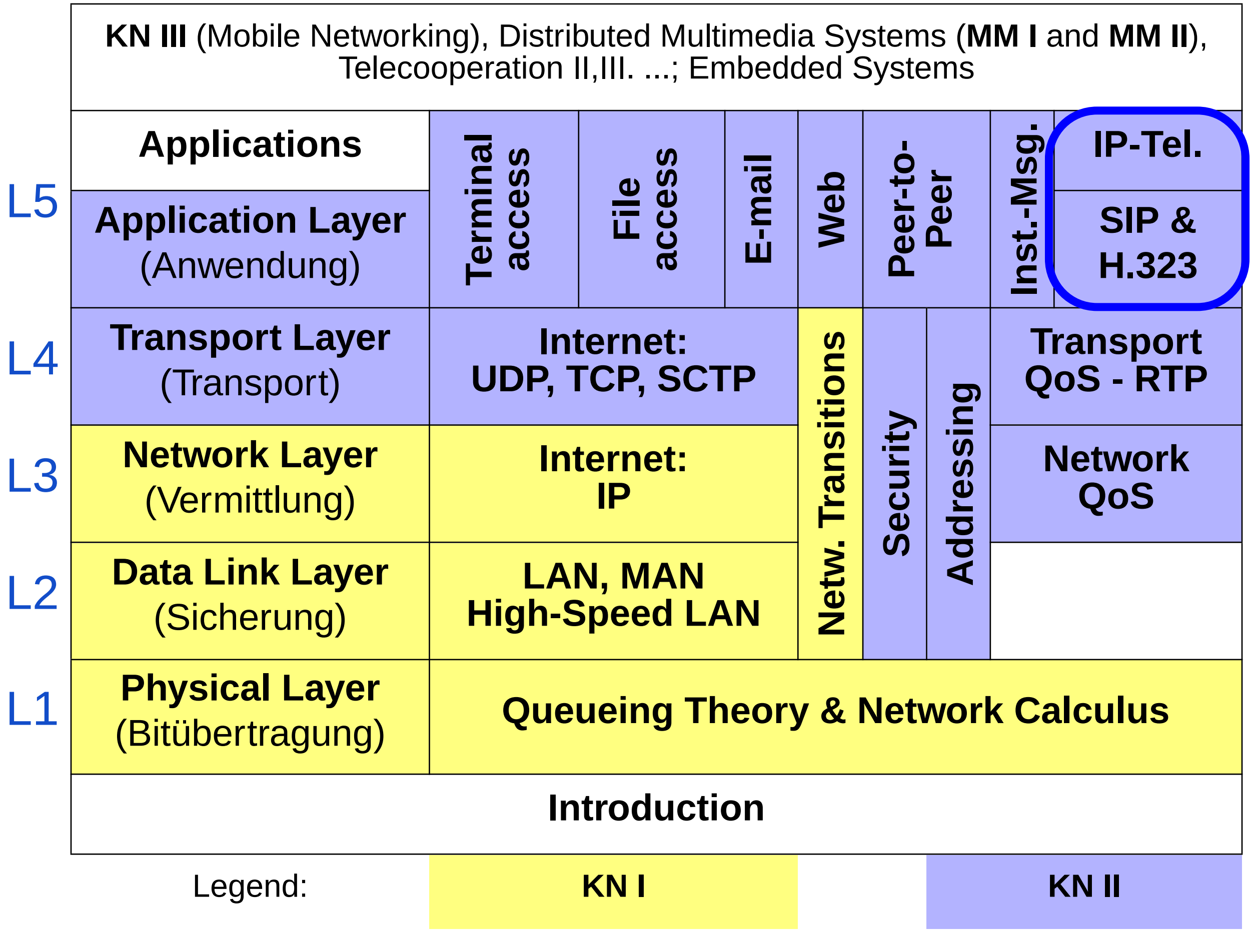
Scope


KN III (Mobile Networking), Distributed Multimedia Systems (MM I and MM II), Telecooperation II,III. ...; Embedded Systems
L5
Applications
Application Layer (Anwendung)
Terminal access
File access
E-mail
Web
Peer-to-Peer
Inst.-Msg.
IP-Tel.
SIP & H.323
L4
Transport Layer (Transport)
Internet: UDP, TCP, SCTP
Transport QoS - RTP
L3
Network Layer (Vermittlung)
Internet: IP
Network QoS
L2
Data Link Layer (Sicherung)
LAN, MAN High-Speed LAN
Netw. Transitions
Security
Addressing
L1
Physical Layer (Bitübertragung)
Queueing Theory & Network Calculus
Introduction
Legend:
KN I
KN II

Overview

Overview (2)

3. Signaling: H.323

3.1 H.323 - Architecture
3.2 H.323 - Media Coding
3.3 H.323 - Basic Elements
3.4 H.323 - Functions
3.5 Phases
3.6 H.323 - A Scenario

4. Signaling: Session Initiation Protocol - SIP

4.1 Session Description Protocol - SDP
4.2 SIP - Components
4.3 SIP - Example - Control and Data Flow
4.4 SIP - Requests/Methods, Responses, Header e.g.

5. H.323 + SIP

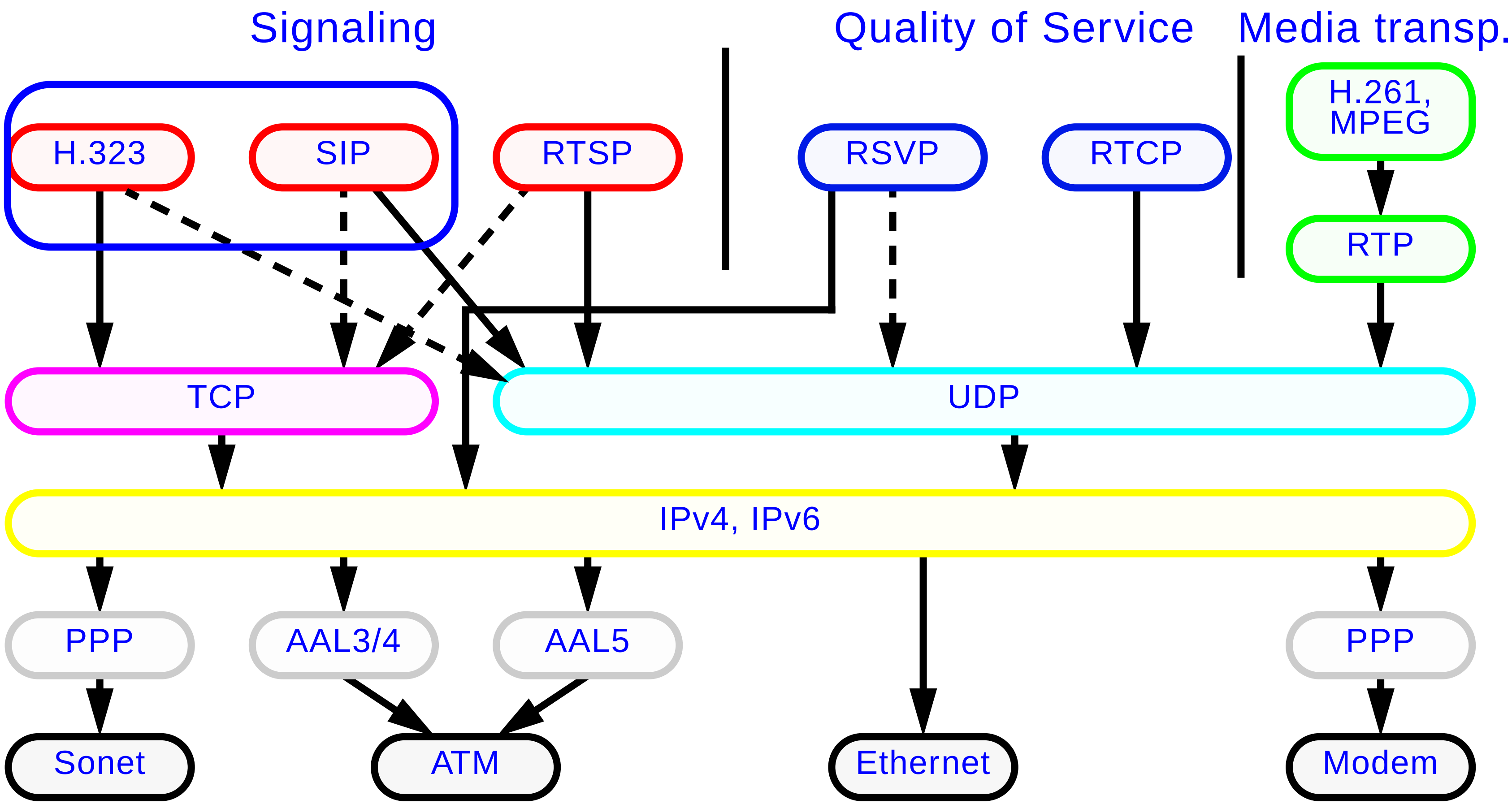
6. Media Gateway Control Protocol (MGCP)

Overview (3)

Overview (4)

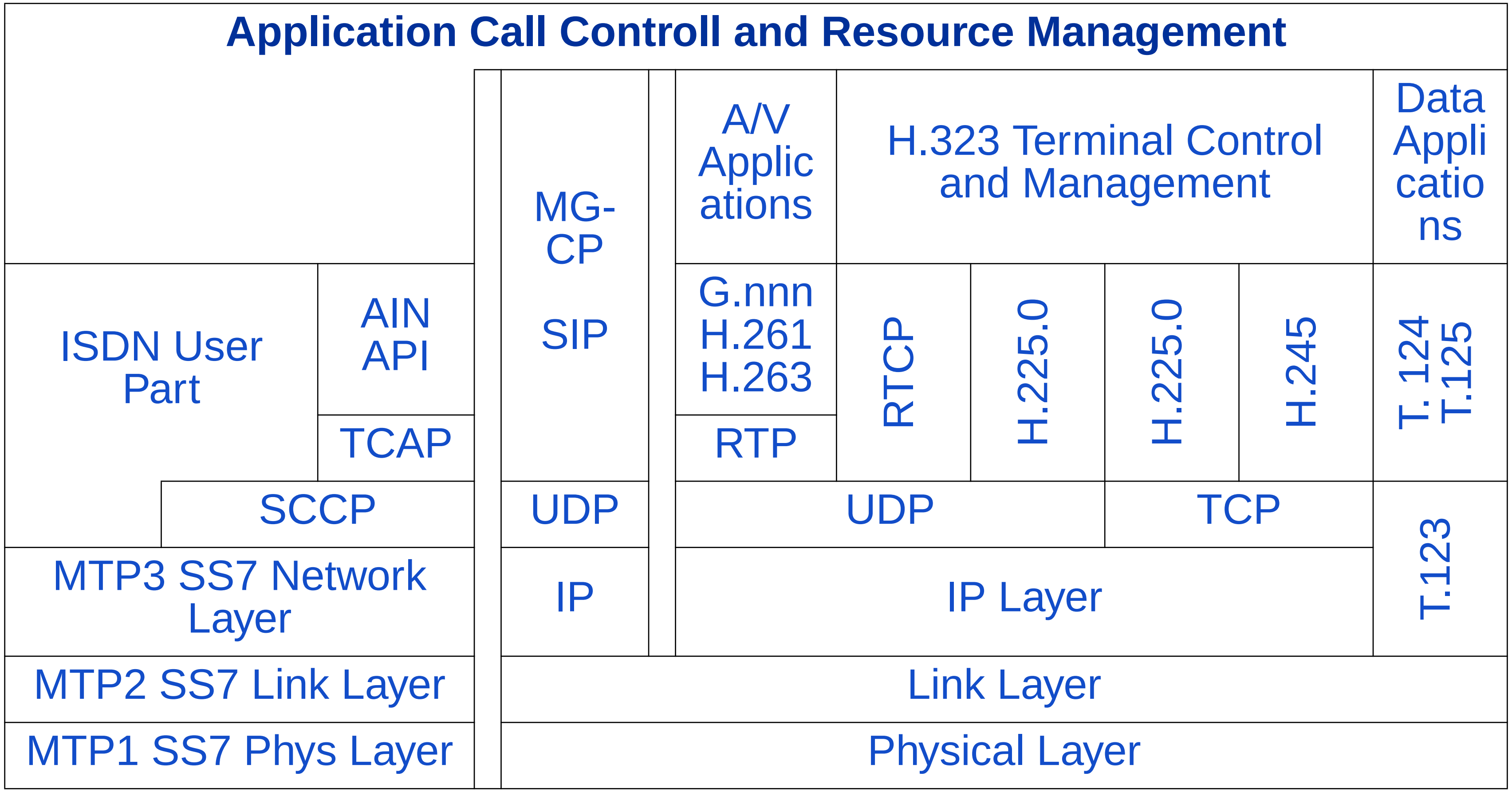
1. Introduction

Based on Internet Real-time and Multimedia Protocols

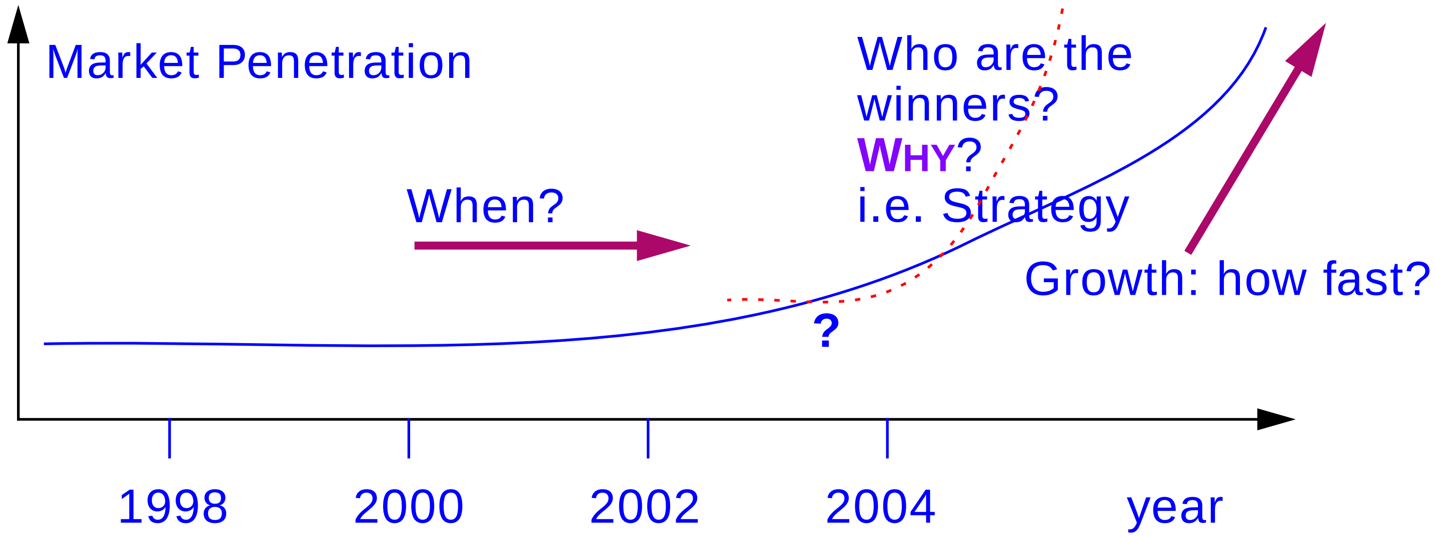


Introduction (2)

And in some more detail



1.1 Introduction: Motivation and Expectations



Trend:

- **shift towards dominance of data traffic (telephony volume << data vol.)**
- **convergence of voice/telecommunications and data/Internet world**

interoperating terminals

IP Telephony in and between **INTRANETS**

Interoperable and interacting gateways

IP Telephony enhanced / based PBXes

IP Telephony as common service in the **INTERNET**

new services and related applications

time

?

1.2 Introduction: Origin

IP-Telephony evolved

- **over many years from**
  - early voice over networks, e.g. Etherphone
    - early experiments at Xerox with audio over Ethernet
  - Computer Telephony Integration CTI
    - control of hardphones or Modem/ISDN-cards in a PC
- **and more recently from**
  - Video Conferencing
    - as a special case (1 to 1 communications)
  - Mbone
    - multicasting audio data over the Internet
  - Audio in many PCs
    - Softphones

1.3 Internet+Intranet Telephony: What do we mean?

What we do NOT mean:

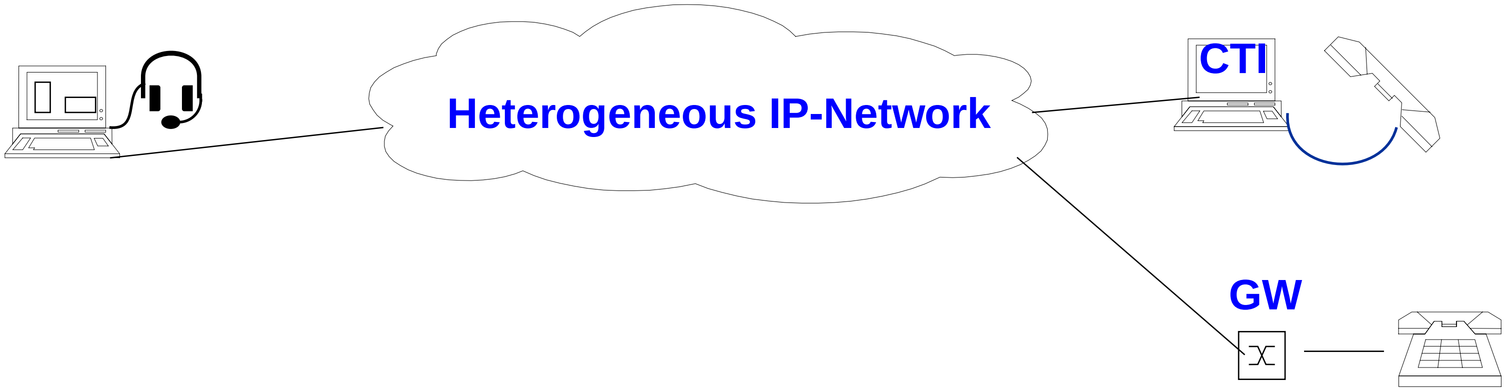
- **just the simplified user's point of view (out of date)**
  - like user interfaces to phone on a PC, soundcards & softphones
  - i.e. just to surf and to talk
- **VoIP: Voice over IP**
  - just voice packets over an IP protocol
  - i.e. no other media (as voice allowed)
- **Internet telephony (as known in massmedia)**
  - only as a means for cheap phoning
  - i.e. it is much more

What we do mean:

- **"Professional" view**
  - integration of
    - telecommunications and data networks,
    - related services and applications
  - simplified management and operation,
  - PBX and data network enhancements or replacements
  - Value Added Services

IP-Telephony: What do we mean?

What we do NOT only mean: The “well-known point of view”



Characteristics

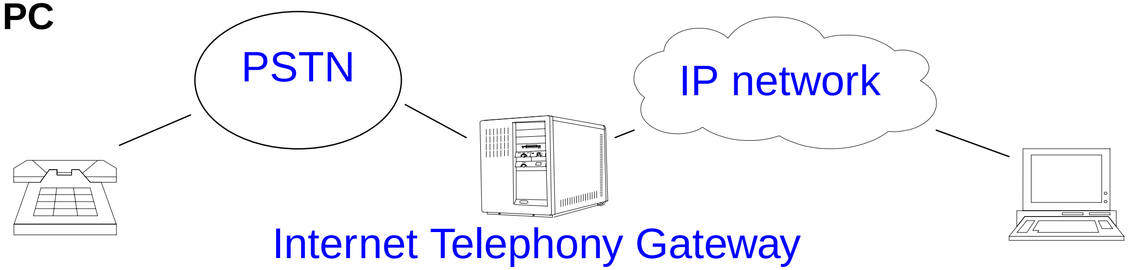
- **cost saving**
  - cheaper call ⇒ flatrate is needed
  - (hardly any) hardware costs ⇒ just a soundcard and a headset
  - only one line/cabling is needed (surf&talk)
- **quality**
  - low quality is accepted if its for free
- **services**
  - participants lists ⇒ my telephone book
  - configuration (skins) for Softphones

Internet+Intranet Telephony: What do we mean?

The “professional view”: Scenarios (apart from PC to PC)

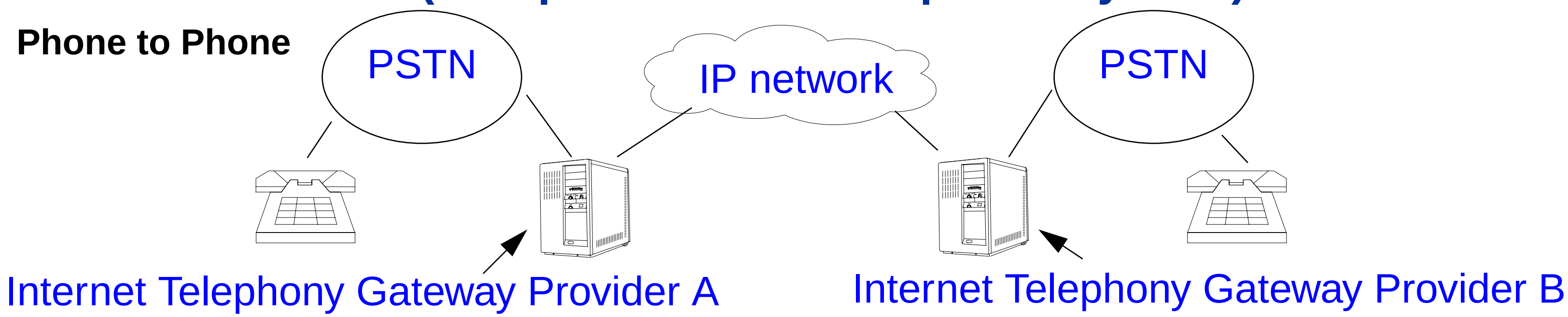
- **deployment of dial-in and dial-out facilities**

Phone to PC



- **IP-based backbone (as a part of future telephone system)**

Phone to Phone



Characteristics

- **cost saving**
- **quality**
- **services**
- **convergence**

Traditional Telephony vs. IP-Telephony

Circuit switched

- **analog and digital networks**
- **internal sophisticated SS7 protocol**

Well known basic call model:

- **finite state machine:**
  - off-hook, tone, ring, ..

Endsystem

- **easy and cheap**
- **Q.931 protocol to access switch**

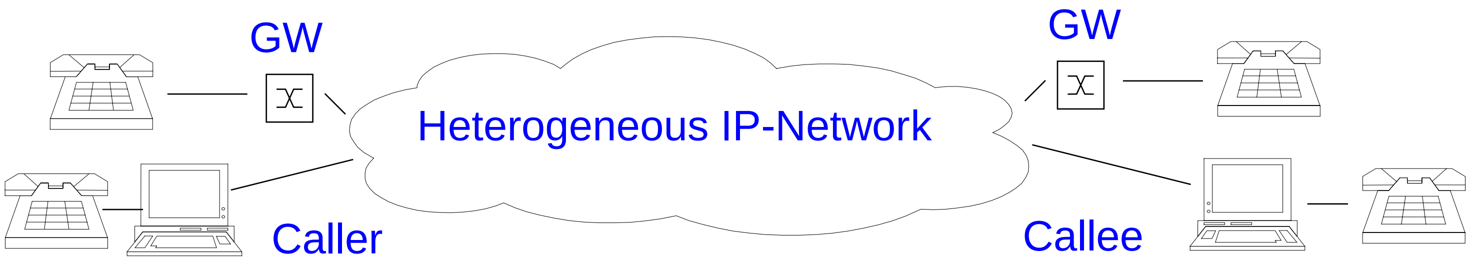
High reliability of individual switches

- **99.9999% (less than 5 min/year)**

Centralized network intelligence

- **centralized service control points SCP for services**
- **access through add-on to switches: service switching points SSP**

2. Scenarios, Basics & Principles, Building Blocks



Calling Party

- **to contact partner using**
  - direct IP-addressing, personal telephone book or directory service
- **to use “Internet telephony service provider” gateway**
  - in order to reach phone partners on POTS (plain old telephone system)
- **to receive call processing information (ring, busy)**
- **to perform full- or half-duplex conversation**

Called Party

- **possible registration with variety of directories**
- **application (standalone / applet) or start-up mechanism listening on (proprietary / “well-known”) port**
- **“ring” - predetermines handling**
  - audible / visible information,
  - accept, reject, store, forward, answering machine ...

2.1 Enhanced Scenarios: IP-Telephony Gateways

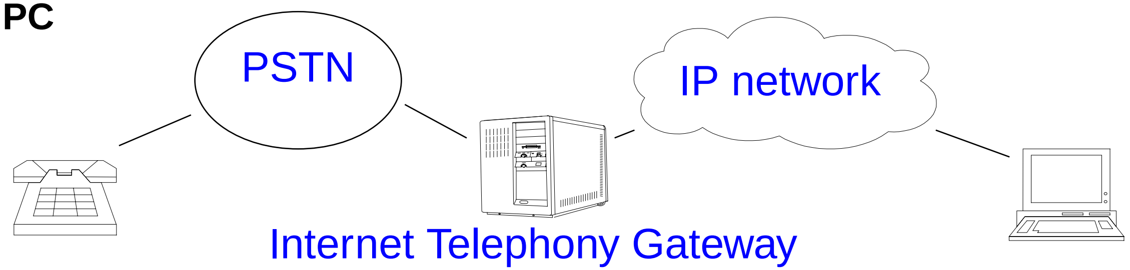
Goals

- **to extend the range of "reachable" users**
- **to have means for**
  - gathering experience & gaining widespread acceptance
  - transition period

Scenarios (apart from PC to PC)

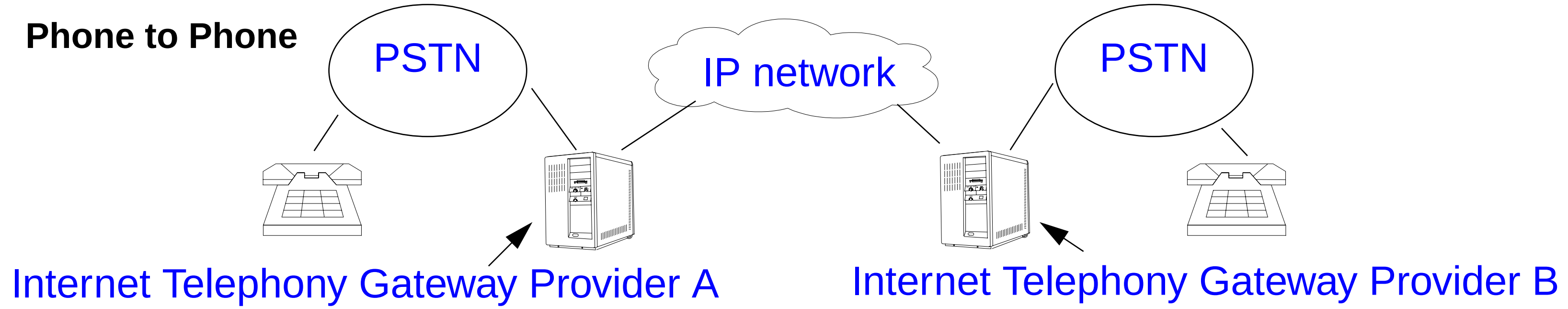
- **deployment of dial-in and dial-out facilities**

Phone to PC

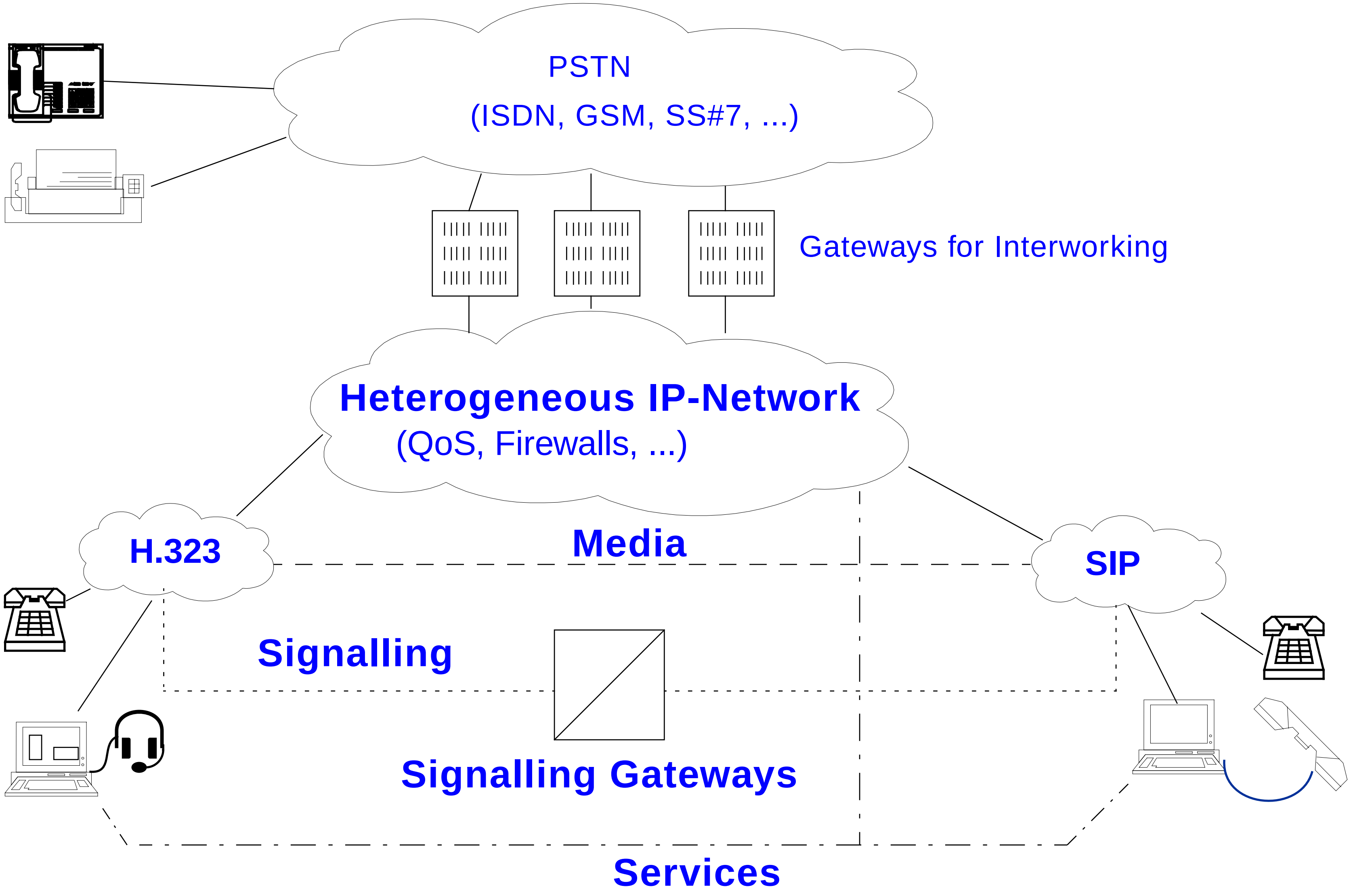


- **IP-based backbone (as a part of future telephone system)**

Phone to Phone



IP Telephony - Comprehensive View



2.2 Building Blocks

IP Telephony
- demands
- expectations
- limitations

Basic Connectivity
- bandwidth
- technology (IP over/vs. ATM)
- tetwork planning
- QoS support
- ...

Value Added Services
- new applications
- gateways
- surveillance
- unified messaging

Signaling (telephony related)
- to end systems
- in the backbone

Operation
- migration strategy management
- maintenance
- security infrastructure
- ...

Building Blocks (2)

Basic Connectivity

- **interoperability of end systems**
- **adaptive encoding**

Signaling

- **standardized infrastructure**
  - to locate and address communication partners
  - find out about their specifics
- **locating (ingress / egress) gateways towards POTS**
- **efficient setup and use of multipoint communication**
- **mobility (identity attached to person, not to a “line jack”)**

- **approaches:**
  - H.323 and H.450.x for supplementarys services
  - Session Initiation Protocol (SIP)
  - Media Gateway Control Protocol (MGCP) / Megaco

⇒ **Trend towards “umbrella standard architecture”**

Building Blocks (3)

Additional Functions for Operating and Maintenance

- **network and facility management**
- **pricing, billing, charging (in the Internet)**
- **security issues like**
  - privacy for end users
  - authentication and protection for infrastructure
  - interaction with other firewalls

Value Added Services

- **vision:**
  - Ubiquitous Seamless Communication
- **integration with other services**
  - e.g. MBone, Unified Messaging
  - Workflow, conferencing, Computer Supported Cooperative Work
- **E-commerce**
- **surveillance**
- **entertainment**
- **distributed games**

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2.3 Driving Forces and Related Protocol Families

IETF Working Group and Mailing Lists

IPTel, Pint

ECTF, IMTC, ITC

SIP

MGCP

„Internet“

- **experience gathered from e.g. multimedia conferencing**
- **combination and extension of protocols**
- **more “lightweight” and adaptable**

Driving Forces

Agreement on data transport but difference on how to deal with control

SS#7

H.3xx

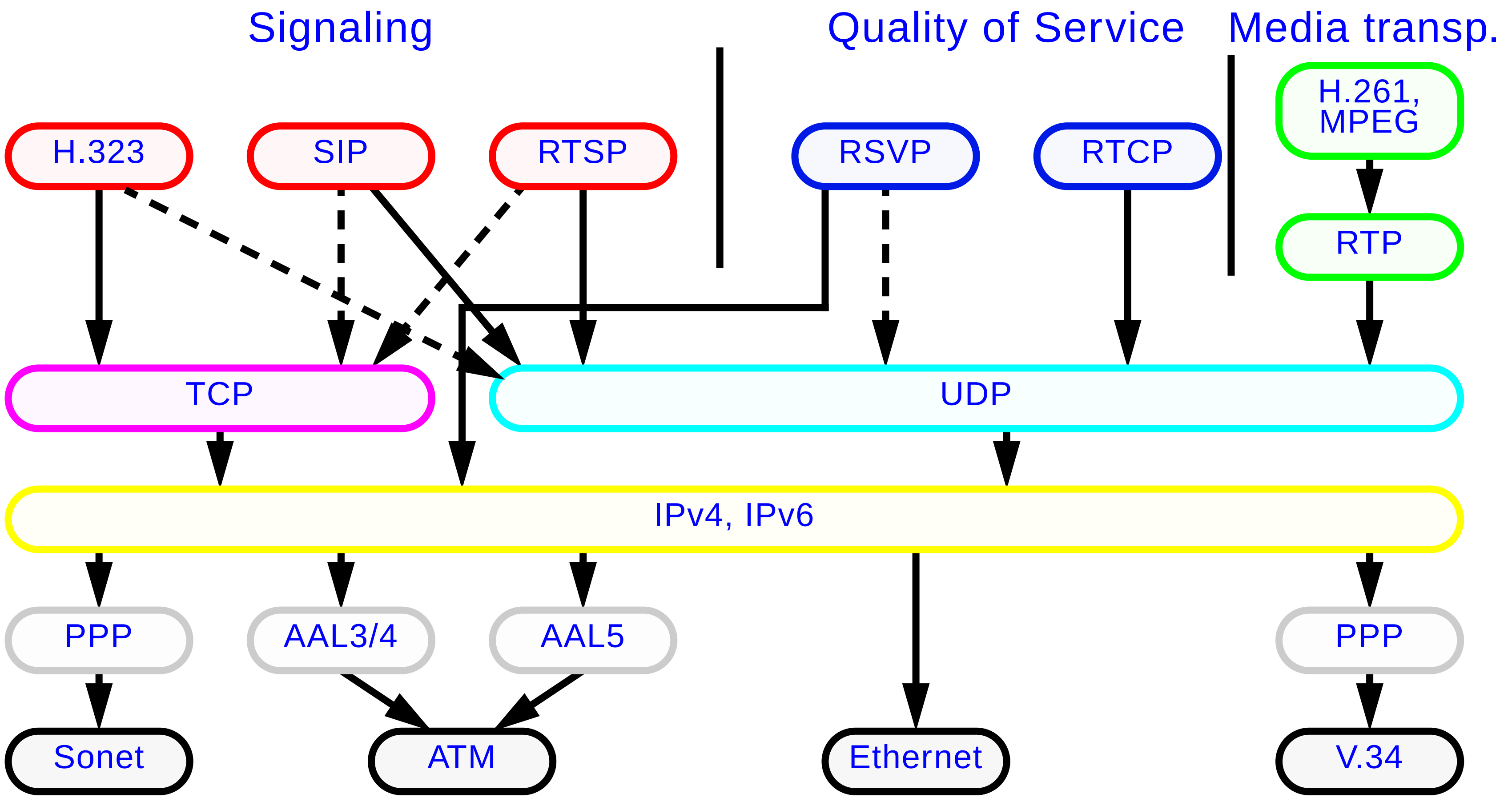
ETSI- Tiphon

ITU-T SG 16

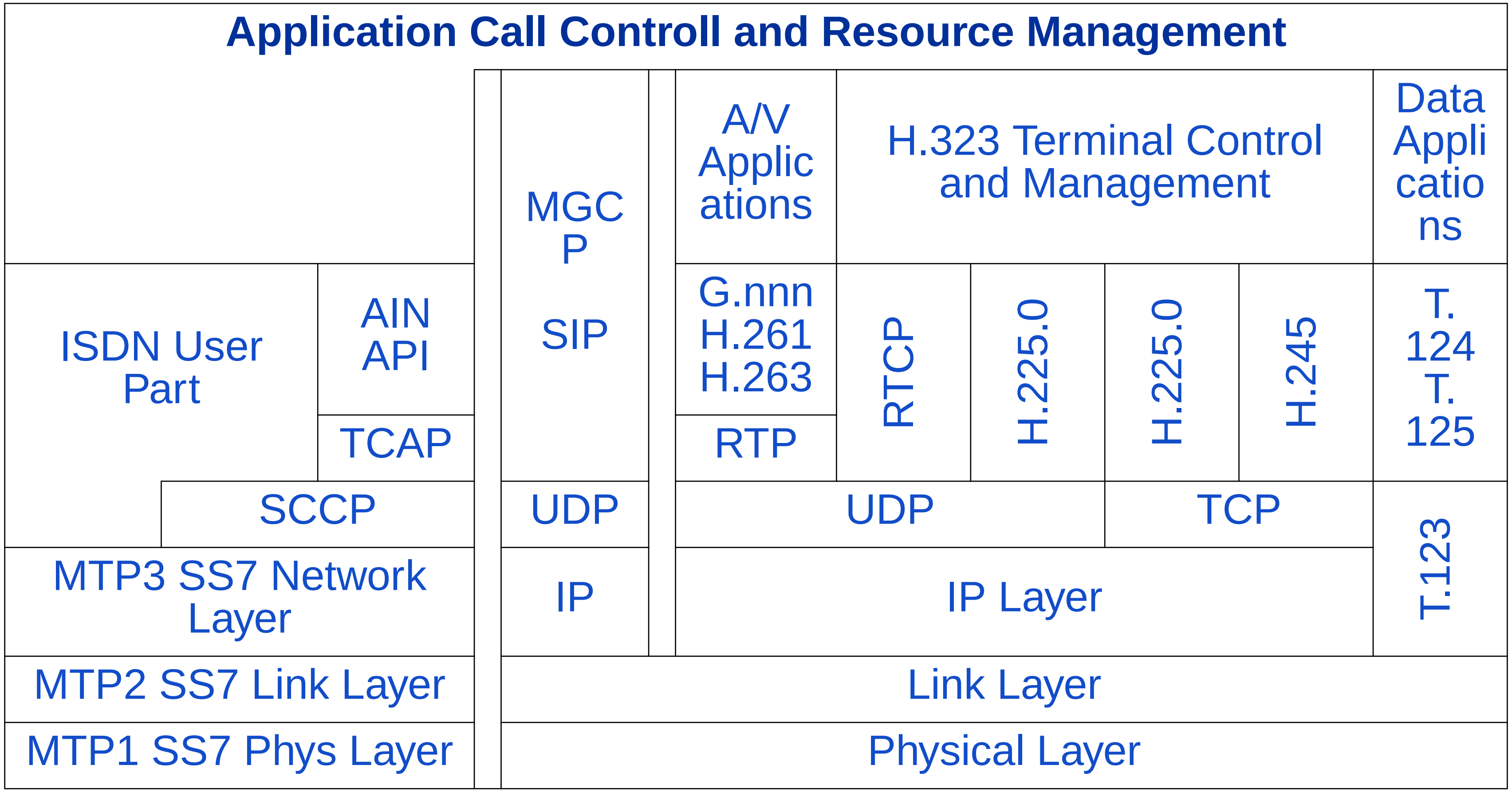
- **H.323 specification**
- **extensive and complex standards**
- **POTS & ISDN know-how**
- **industry acceptance**
- **existence of reference implementations**

„Telecommunications“

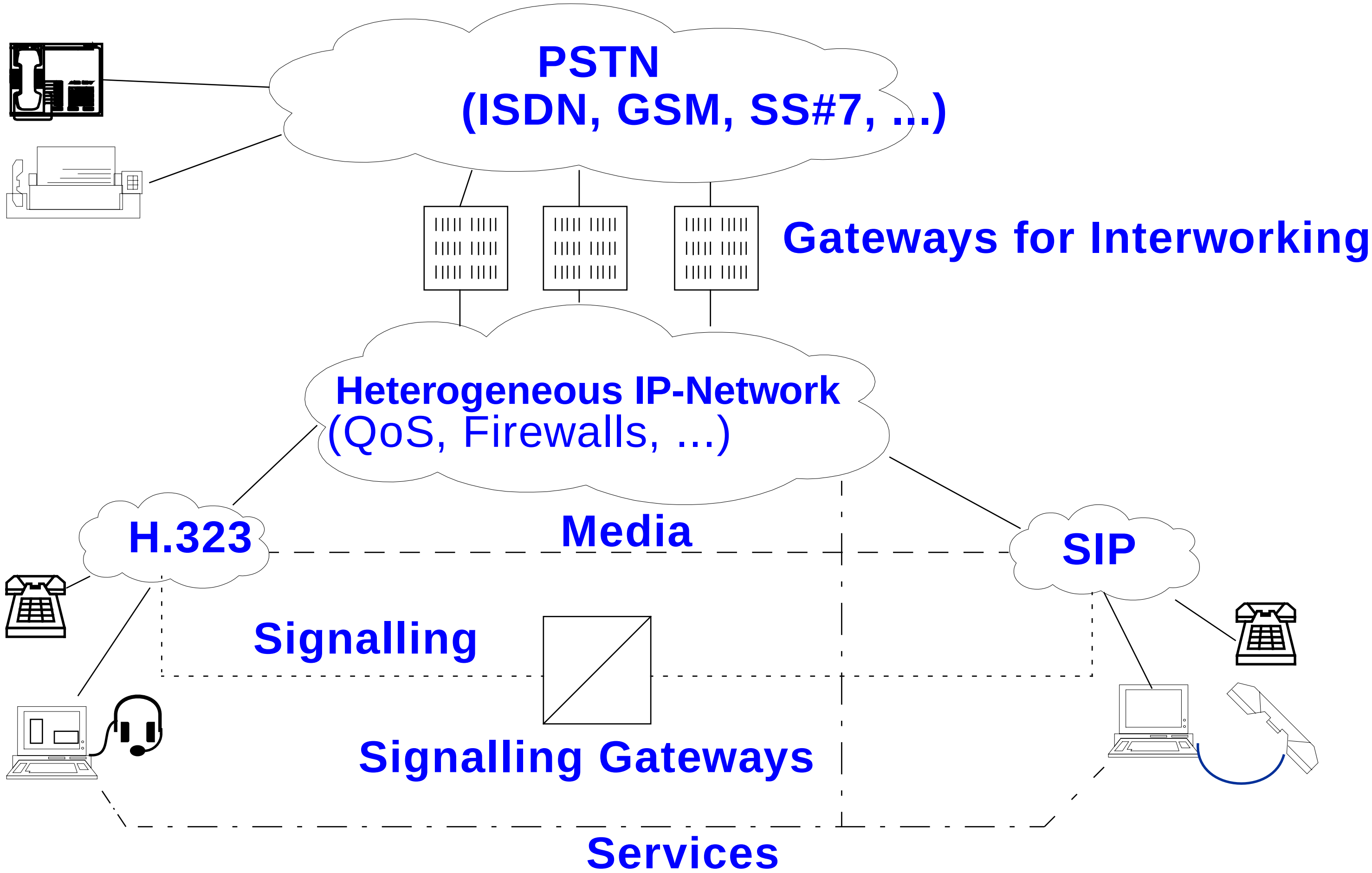
2.4 IP Telephony - Interaction of Protocols



PSTN/IP Network



Signaling Protocols

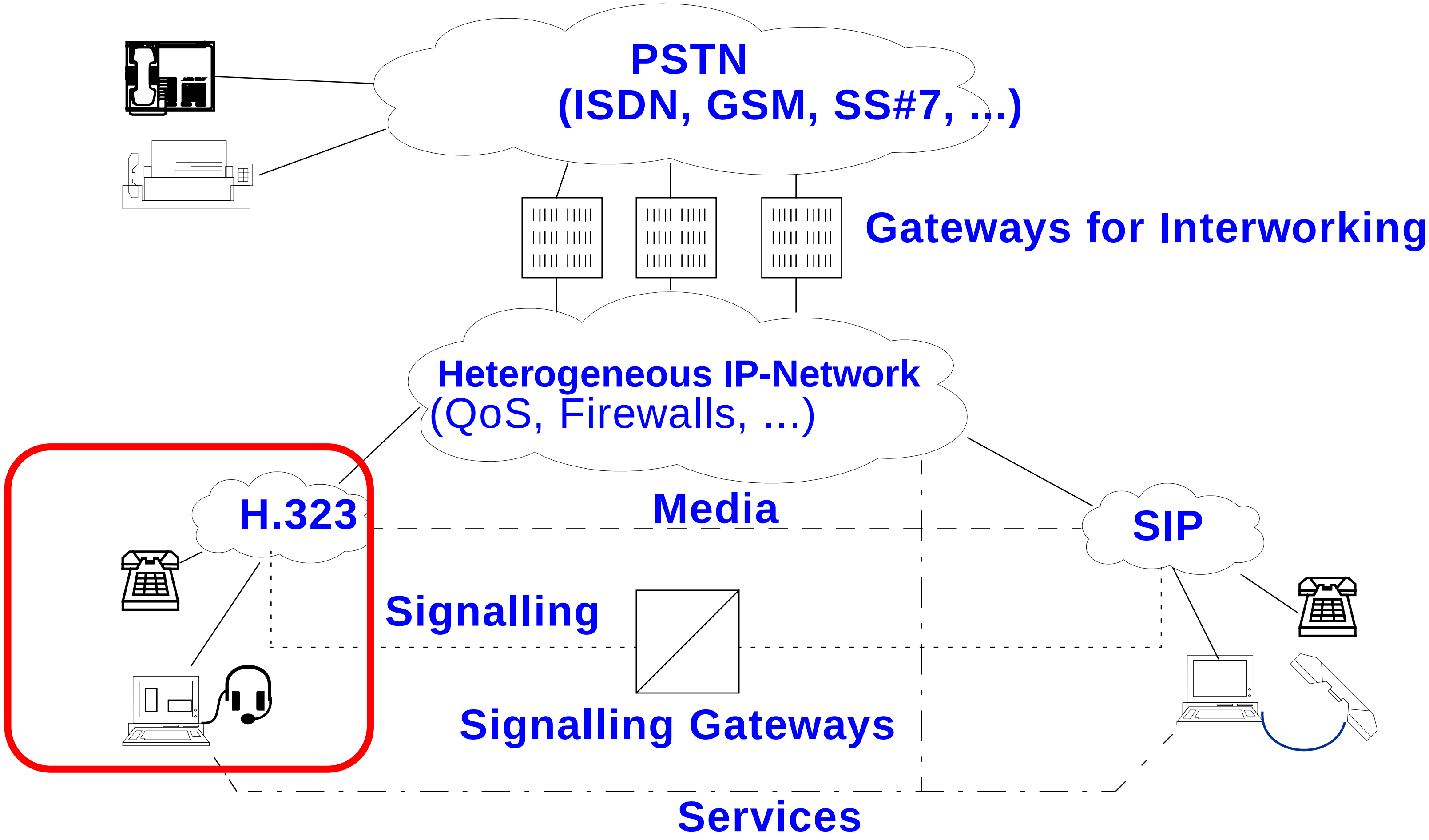


Candidates - Tasks - Basic Principles

- ITU H.323
- Session Initiation Protocol SIP
- S/MGCP and MeGaCo

3. Signaling: H.323

Signaling Protocols Overview



Signaling: H.323 (2)

ITU-T Recommendation

- **industry driven**
- **packet-based multimedia conferencing over LAN**
- **no inherent quality-of-service guarantees**

Properties

- **stateful**
- **binary**
- **session as ASN.1 notation: Packet encoding rules**

Protocol suite

- **H.225.0 / Q.931**
  - registration, authentication, status (RAS)
  - call control
- **H.245**
  - logical channel capabilities negotiation
  - dynamic port negotiation ($\Rightarrow$ problem with firewalls)
- **H.450.x - H.323 Supplementary Services**
  - supplementary services

H.323 as Part of H.320, the H.32x Family

H.320 specifies (as overview) videophone for ISDN

H.310
- adapt MPEG 2 for communication over B-ISDN (ATM)

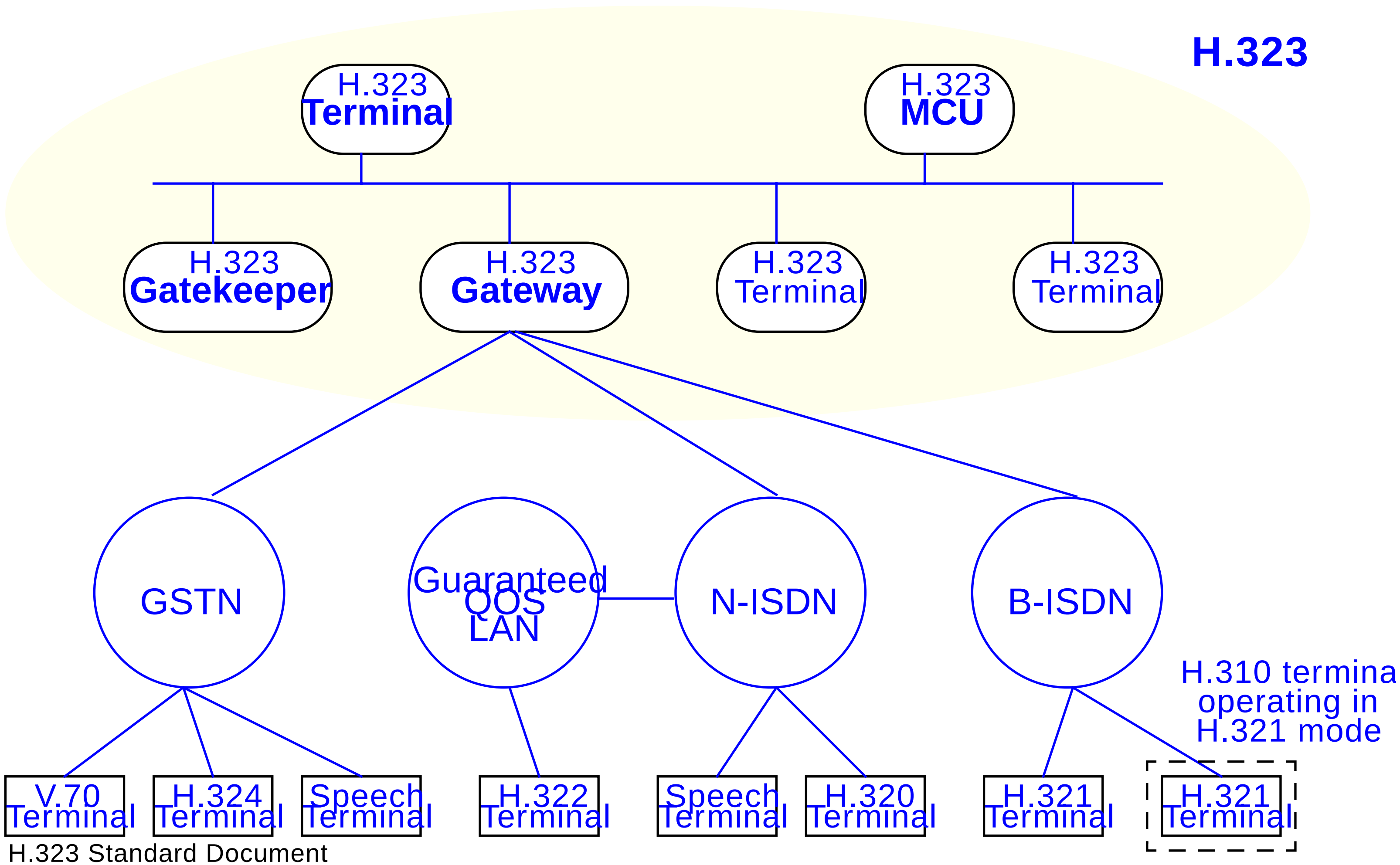
H.321
- define videoconferencing terminal for B-ISDN (instead of N-ISDN)

H.322
- adapts H.320 for guaranteed QoS LANs (like ISO-Ethernet)

H.323
- original: videoconferencing over non-guaranteed LANs

H.324
- terminal for low bit rate communication (over V.34 Modems)

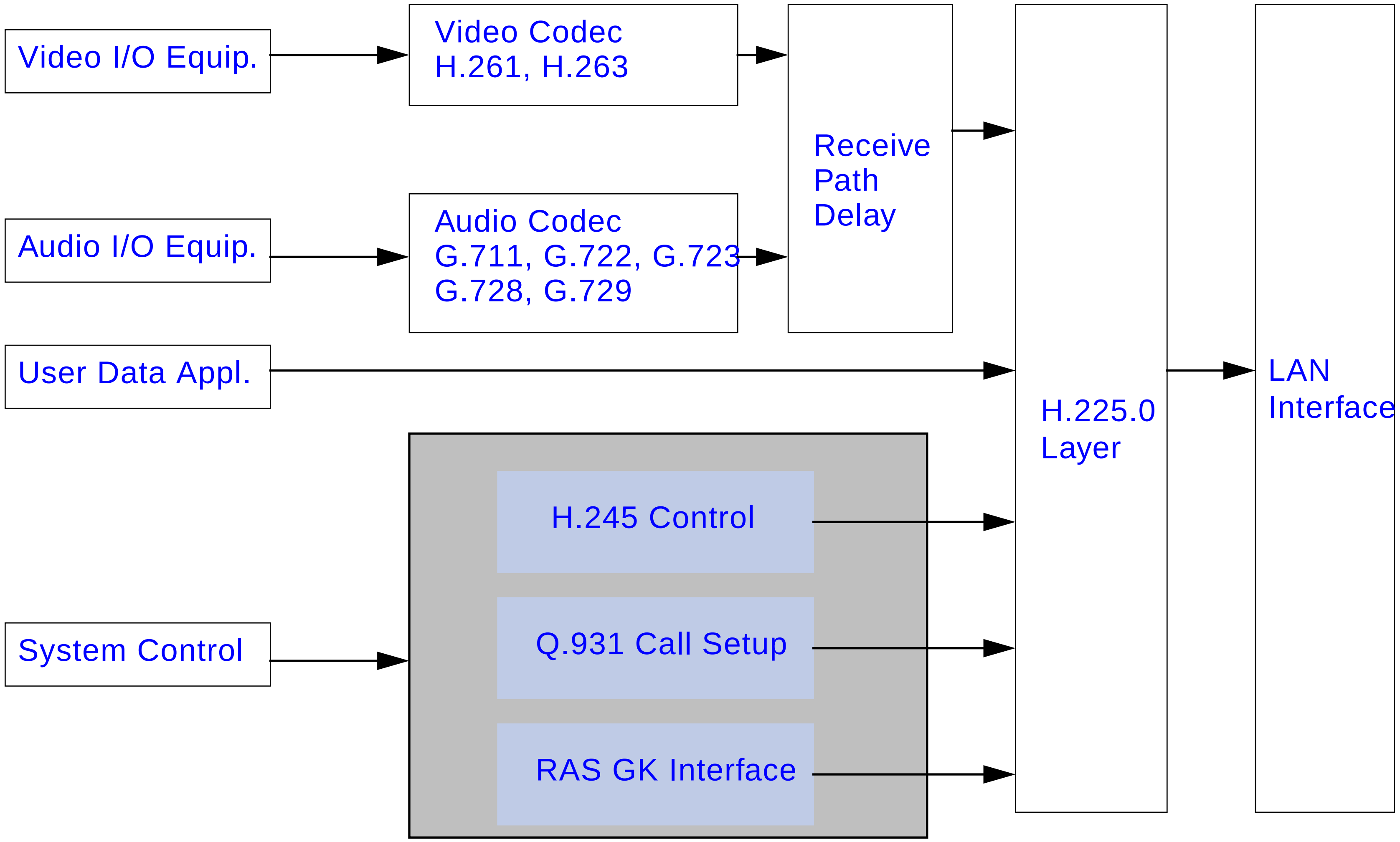
3.1 H.323 - Architecture



Source: H.323 Standard Document

- **interoperable and “industry standard”**
- **may form building block for wide-spread deployment**
- **somehow complex and developed mainly for local environments**

H.323 - Architecture (2)



3.2 H.323 - Media Coding

Specified Audio Coding

ITU-Rec.	Description
G.711	Audio codec, 3.1 KHz at 48, 56, and 64 Kbps (normal telephony)
G.722	Audio Codec, 7 KHz at 48, 56, and 64 Kbps
G.728	Audio Codec, 3.1 KHz at 16 Kbps
G.723	Audio Codec, for 5.3 and 6.3 Kbps modes
G.729	Audio Codec, 8 kbps audio codec

Specified Video Coding

- **H.261: Video codec for audiovisual services at p x 64kbit/s**
  - CCITT standard from 1990
  - for ISDN with p=1,..., 30
  - technical issues:
    - real-time encoding/decoding
    - max. signal delay of 150ms
    - constant data rate
    - implementation in hardware (main goal) and software

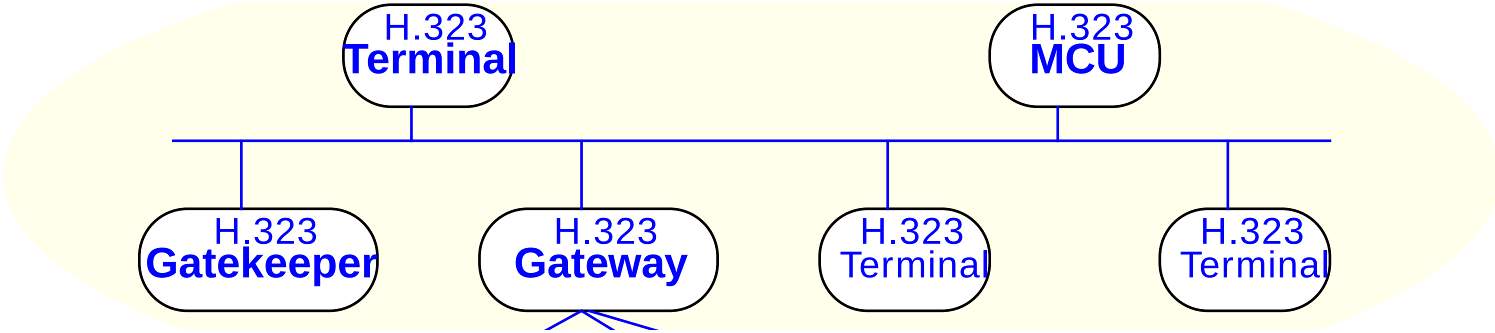
H.323 - Media Coding (2)

Specified Video Coding

- **H.263**
  - extension to H.261
  - max. bitrate: H.263 approx. 2.5 x H.261
- **H.261 and H.263 source image formats**

Format	Pixels	H.261		H.263	
		Encoder	Decoder	Encoder	Decoder
SQCIF	128 x 96	optional		required	
QCIF	176 x 144	required		required	
CIF	352 x 144	optional		optional	
4CIF	704 x 576	not defined		optional	
16CIF	1408 x 1152				

3.3 H.323 - Basic Elements



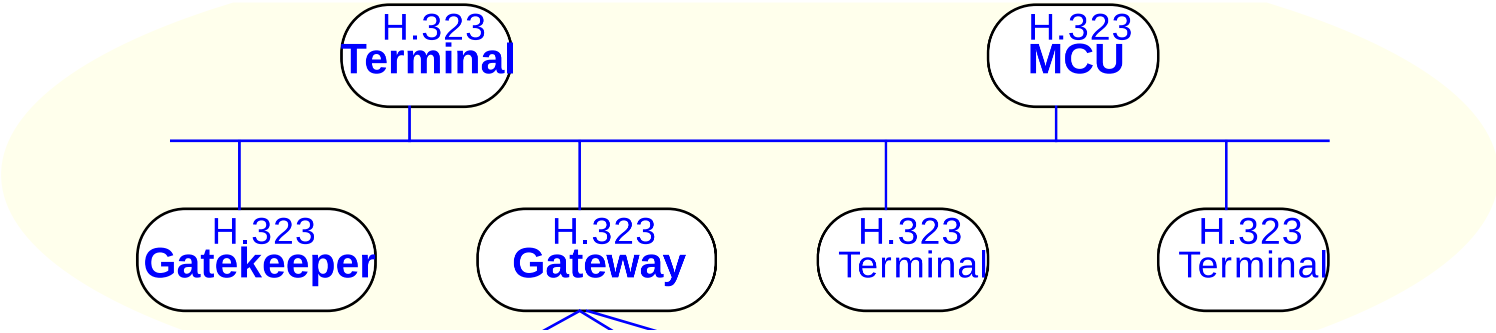
Terminal

- **client endpoints**
- **provide real-time two-way communication**

Gatekeepers (GK)

- **“brain” of a H.323-zone**
  - but optional
- **zone = logical grouping of devices**
  - only one GK per zone
  - no GK means, i.e. no zone exists
- **provides**
  - address translation
  - assigns bandwidth to connections

H.323 - Basic Elements (2)



Multipoint Control Units (MCU)

- **support for multipoint conferences**
- **resource management**
- **codec negotiation**

Gateways (GW)

- **translation functions between**
  - transmission formats, AV codecs
  - communication procedures

3.4 H.323 - Functions

Call processing divided into

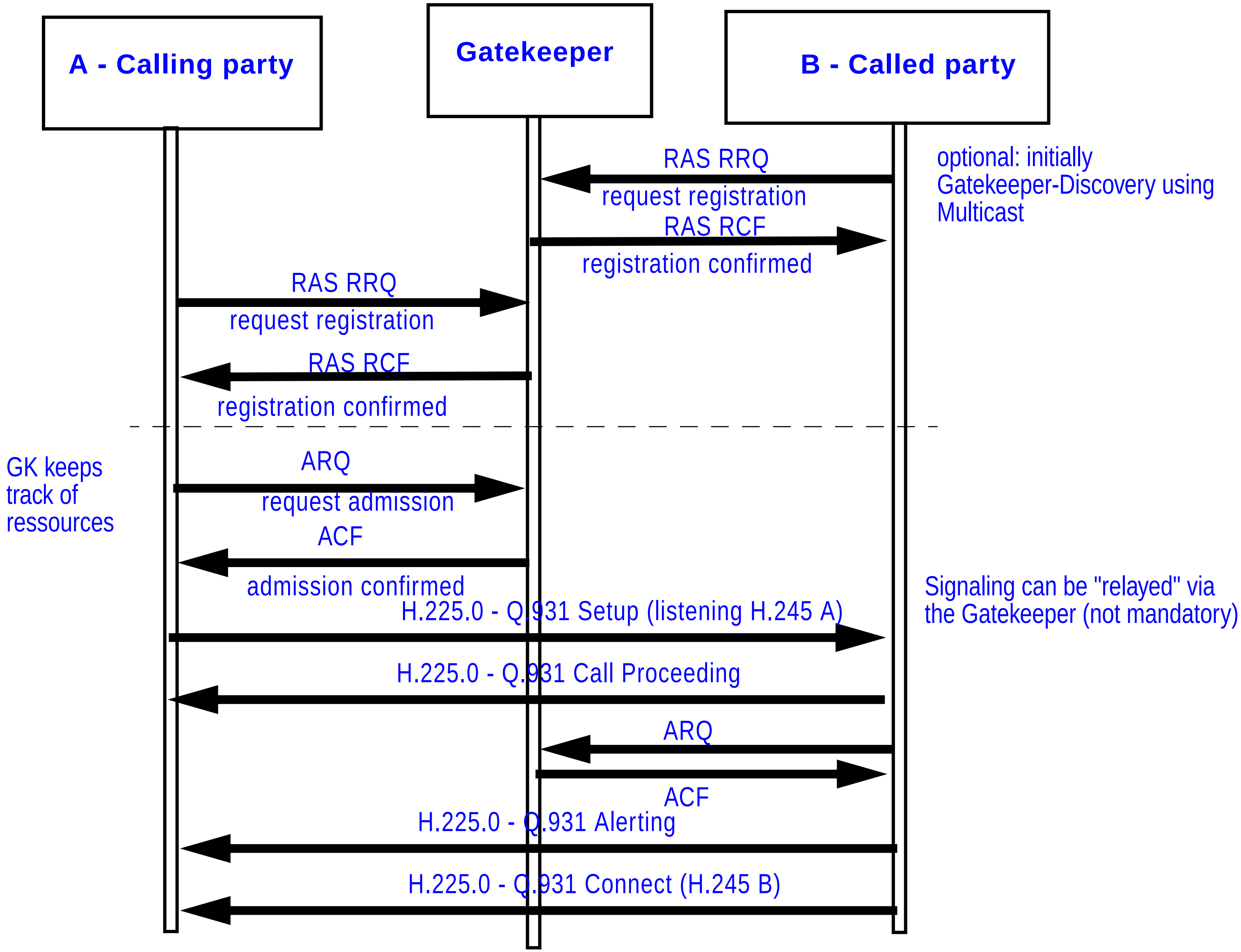
- **registration with the gatekeeper**
  - using **Registration Authentication and Status protocol** RAS
- **call signaling**
  - using H.225.0
  - may be
    - gatekeeper routed - and then H.245 gatekeeper routed
    - gatekeeper routed - and then H.245 directly routed
    - direct i.e. H.245 directly routed
- **media capability exchange**
  - using H.245
  - usually end-to-end

3.5 Phases

1. **Call set-up**
   - e.g. point-to-point or multipoint request
2. **Initial comm. between endpoint & capability exchange**
   - master/slave relation is defined
     - one terminal is defined to be the master
     - with the "highest" media capabilities
     - resolves conflicts of media capability exchange
3. **Establishment of audio/visual communication between endpoints**
4. **Media data flow**
5. **Request and negotiation of Call Services**
   - e.g. request for additional bandwidth
   - may occur any time after having established audio/visual communications
6. **Call termination**

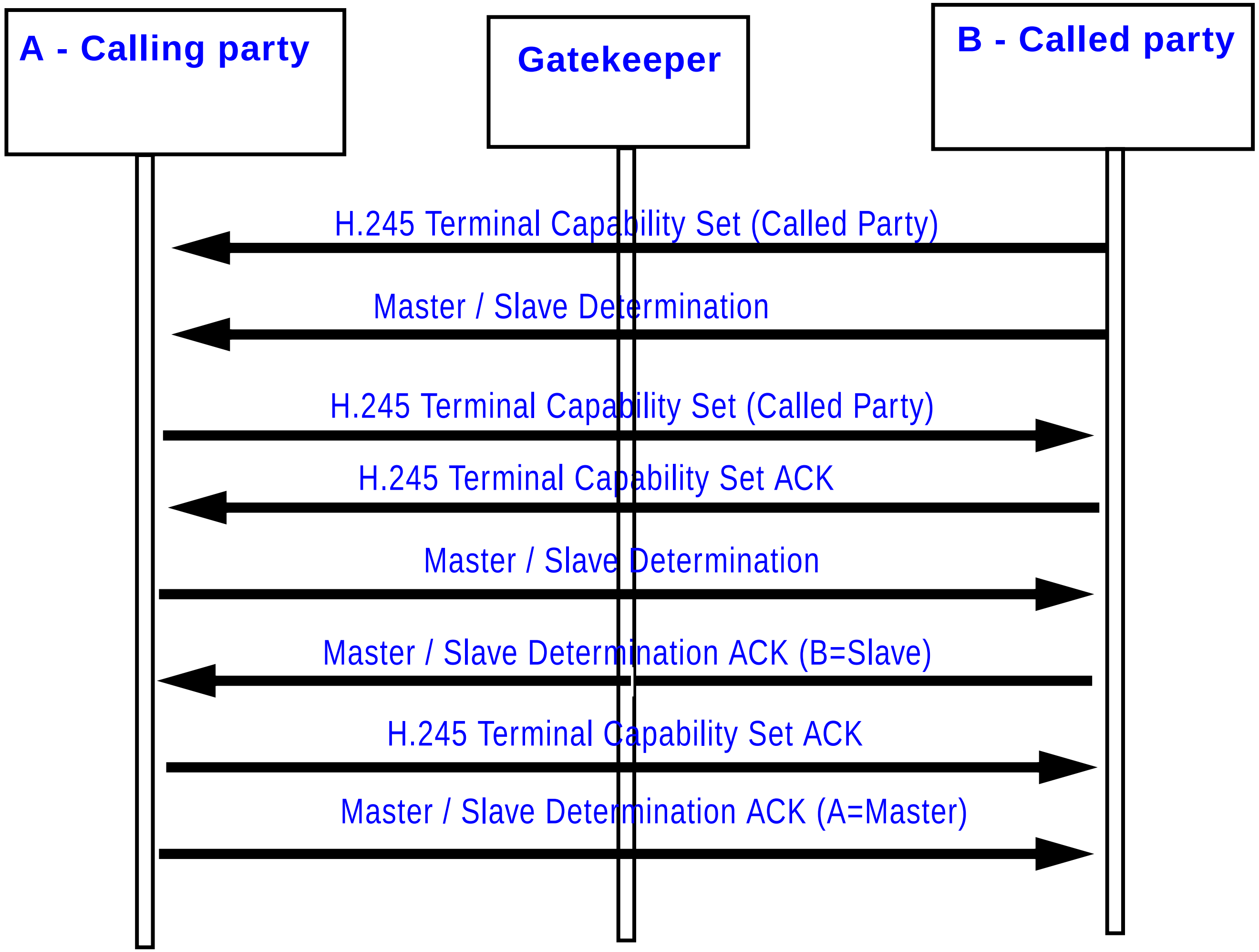
Example of the Phases

1. H323-CallSetUp-1-init



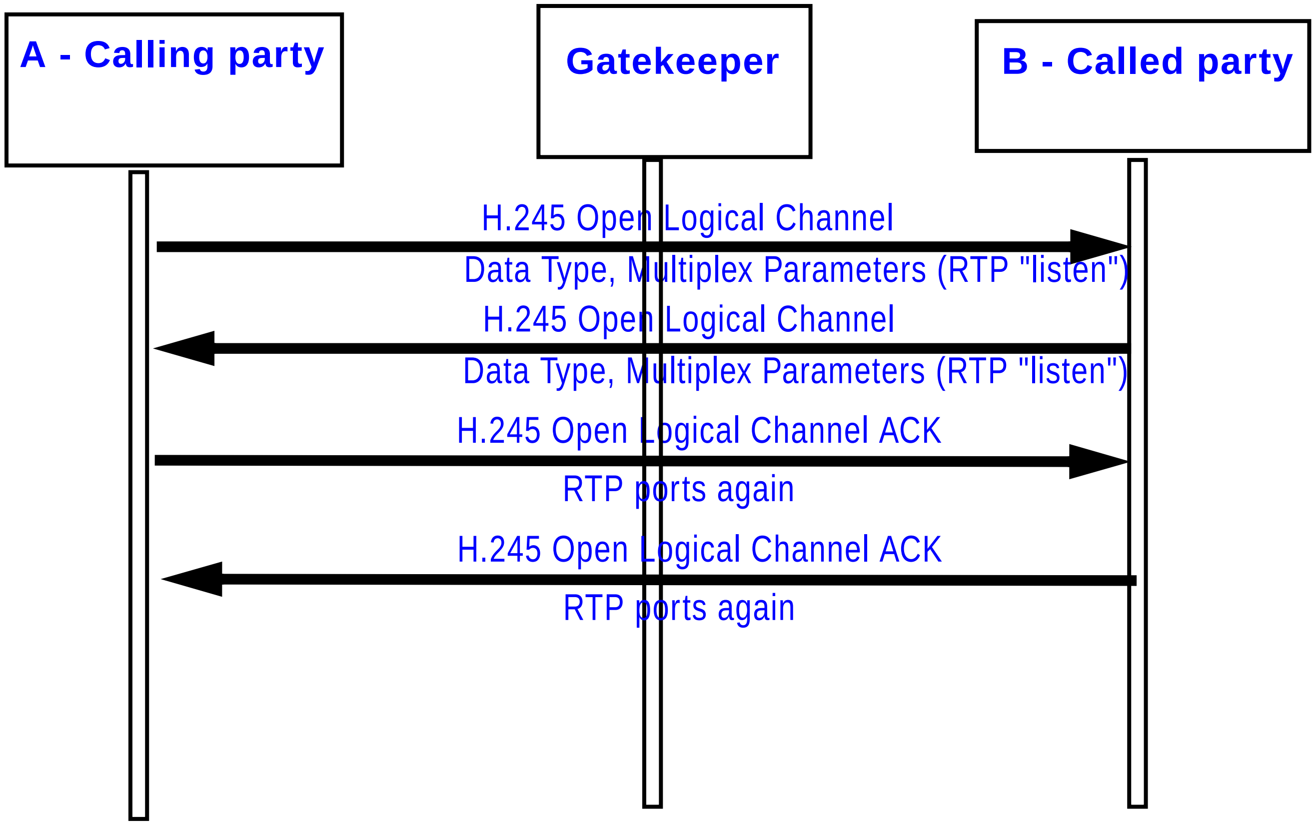
Example of the Phases (2)

2. H323-CallSetUp-2-capability



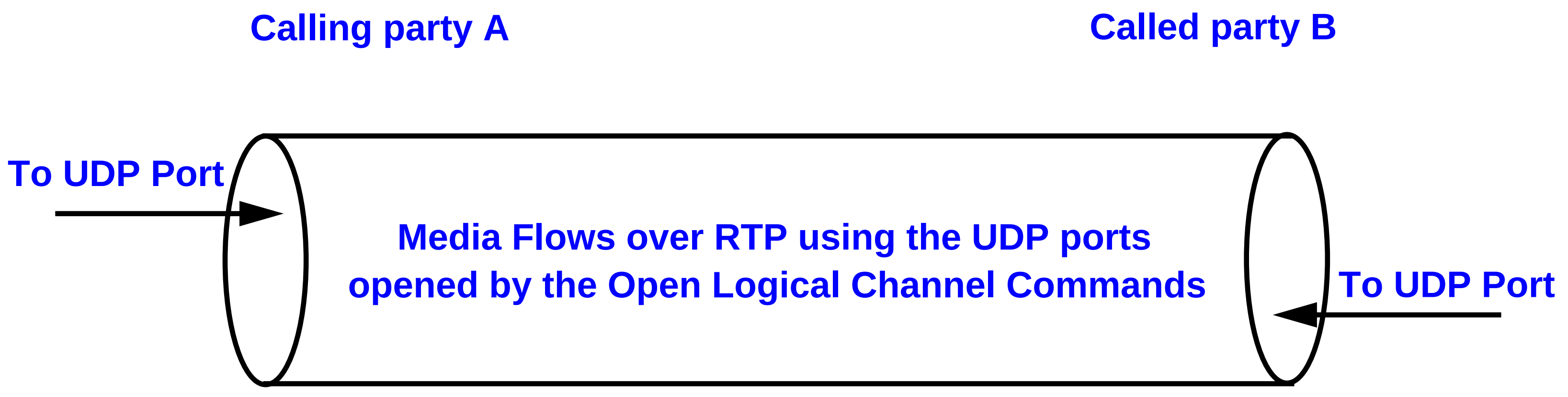
Example of the Phases (3)

3. H323-CallSetUp-3-avcomm



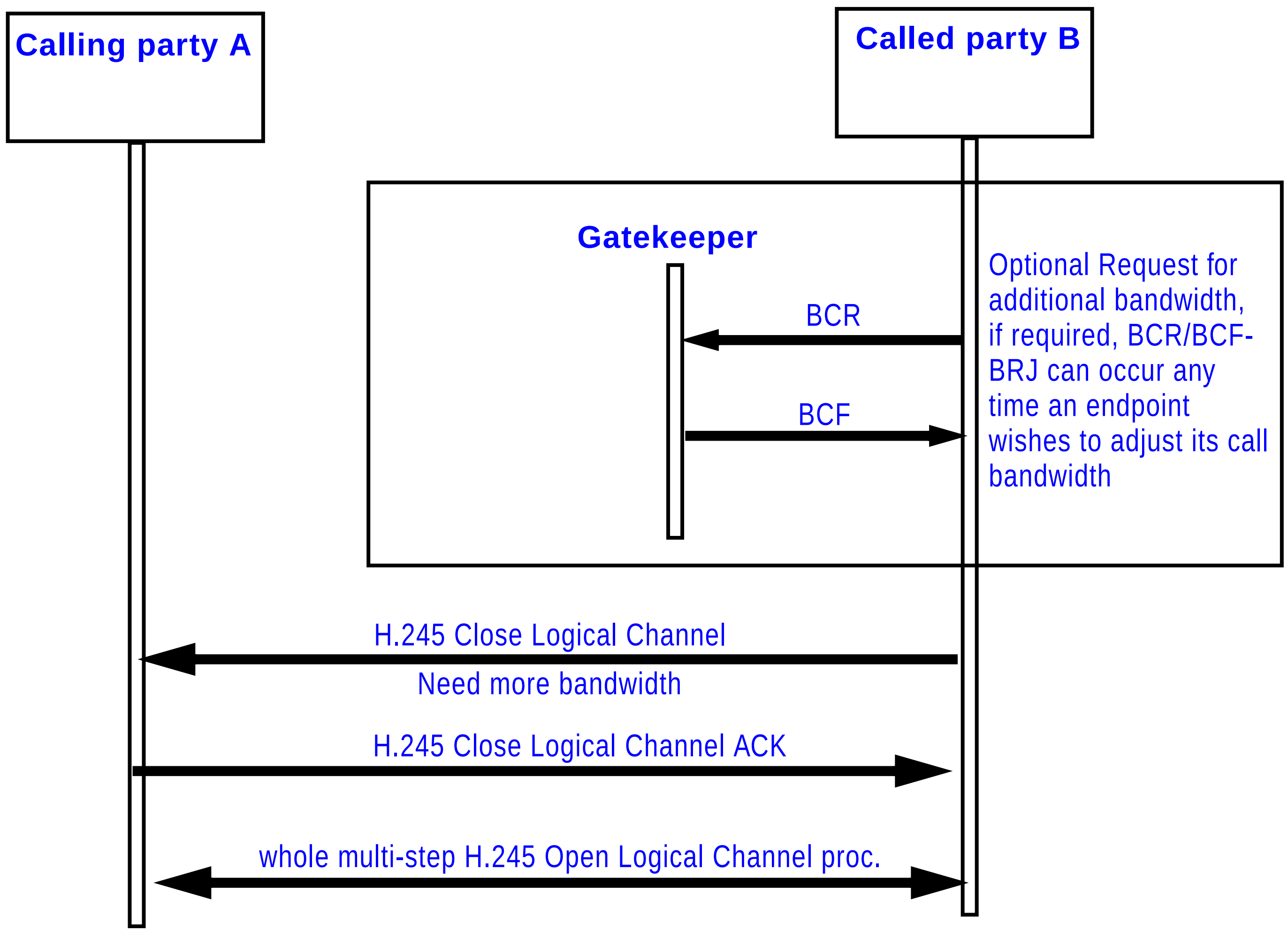
Example of the Phases (4)

4. H323-CallSetUp-4-flow



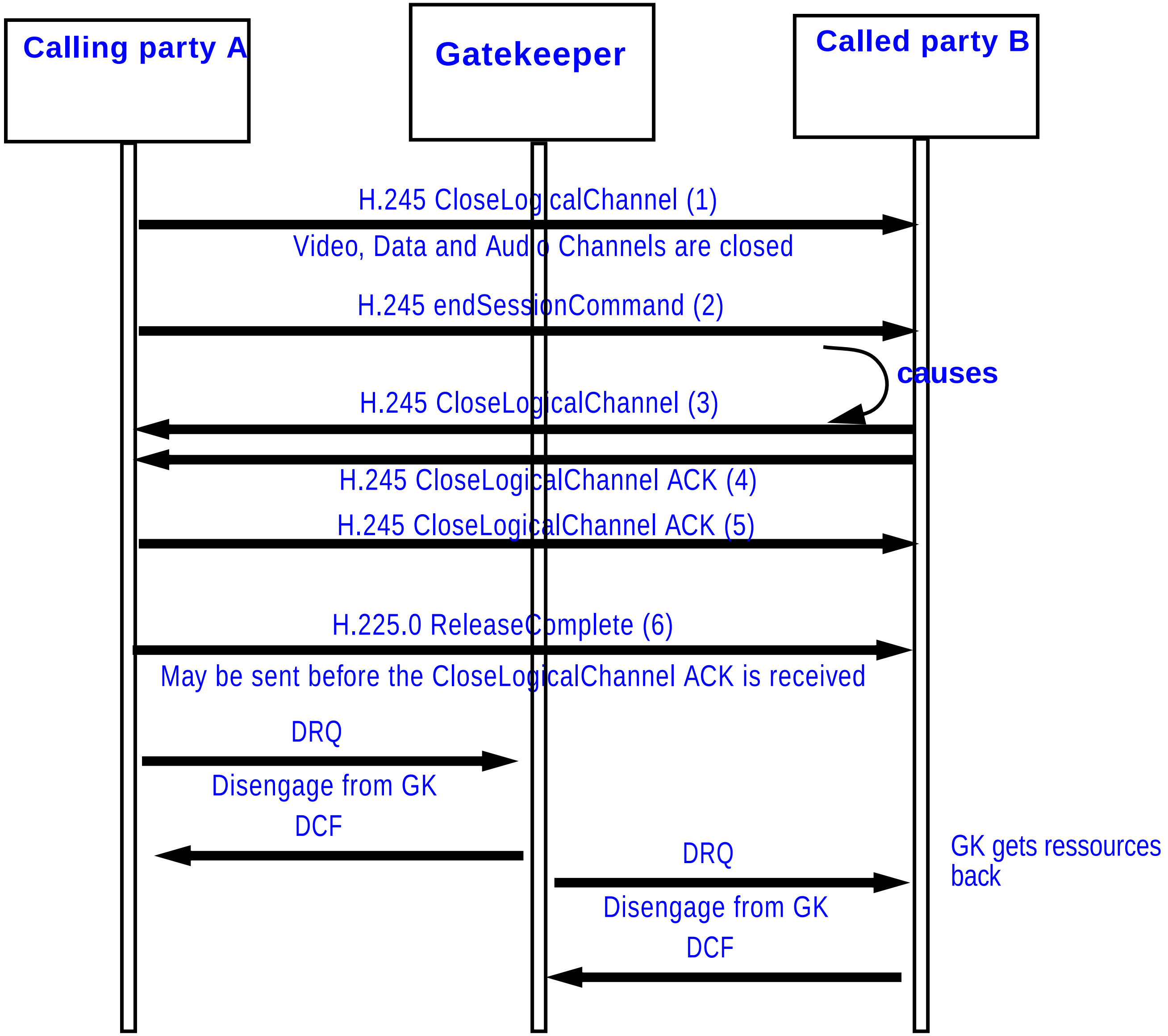
Example of the Phases (5)

5. H323-CallSetUp-5-renegotiate

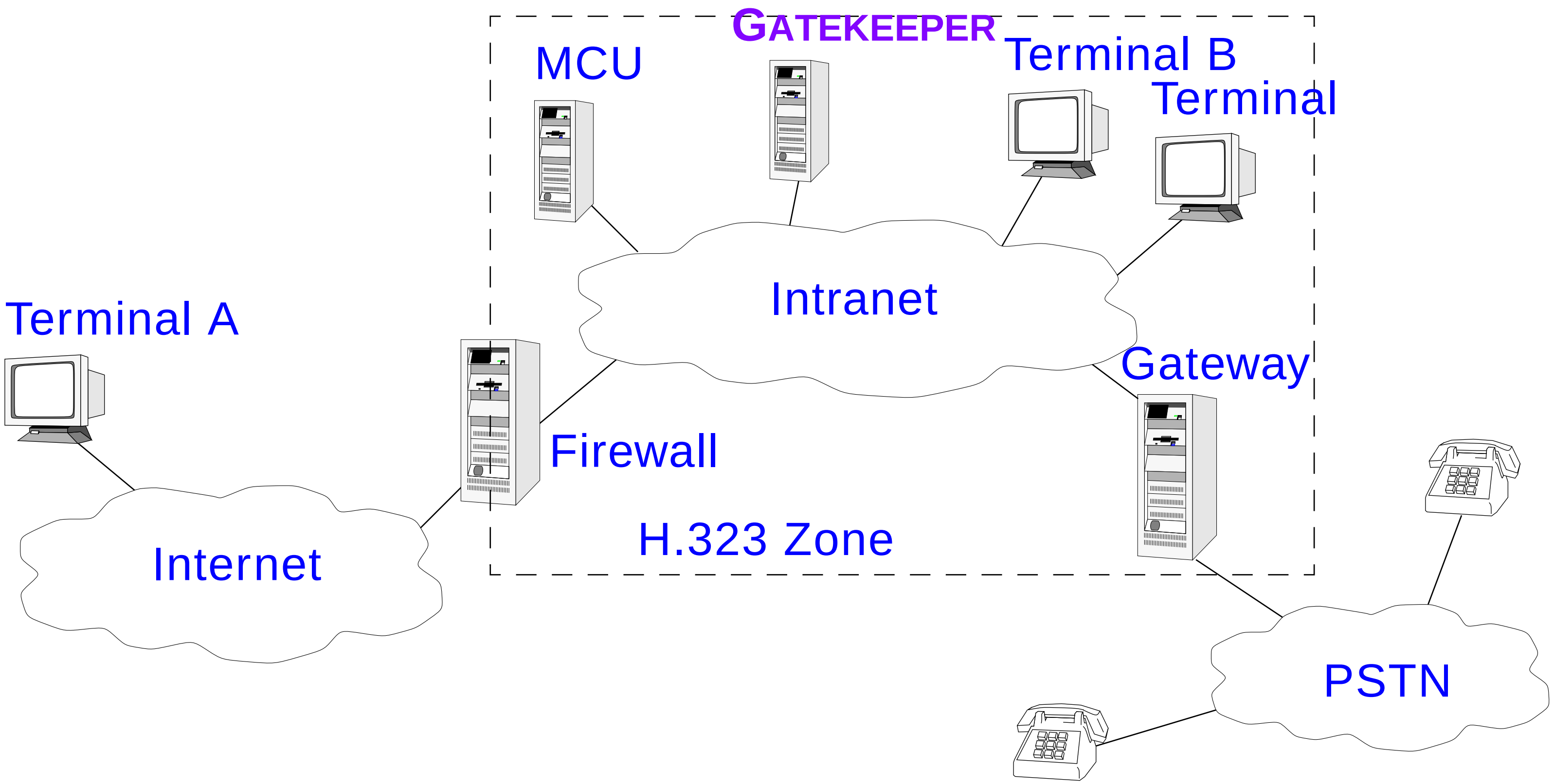


Example of the Phases (6)

6. H323-CallSetUp-6-terminate



3.6 H.323 - A Scenario



i.e. has been in focus of the industry

- **with its extensions for supplementary services**

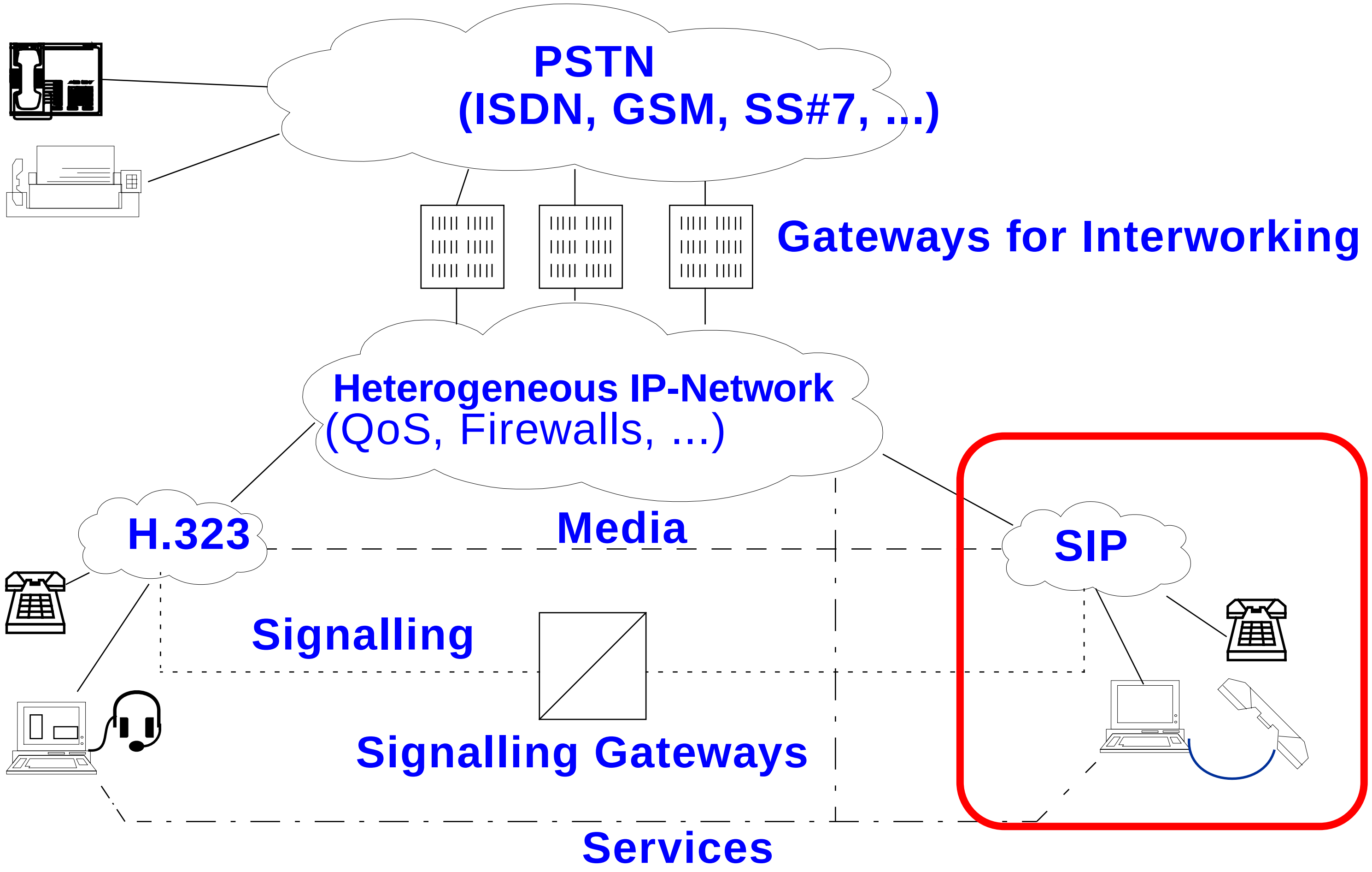
H.323 Supplementary Services - H.450.x

H.450 Protocols

	Description
H.450.1	Generic functional protocol for the support of supplementary services in H.323
H.450.2	Call transfer supplementary service for H.323
H.450.3	Call diversion supplementary service for H.323
H.450.4	Call hold supplementary service for H.323
H.450.5	Call park and call pickup supplementary services for H.323
H.450.6	Call waiting supplementary service for H.323
H.450.7	Message waiting indication supplementary service for H.323
H.450.8	Name identification supplementary service for H.323
H.450.9	Call Completion Supplementary Service
H.450.10	Call Offer Supplementary Service
H.450.11	Call Intrusion Supplementary Service

4. Signaling: Session Initiation Protocol - SIP

Signaling Protocols Overview

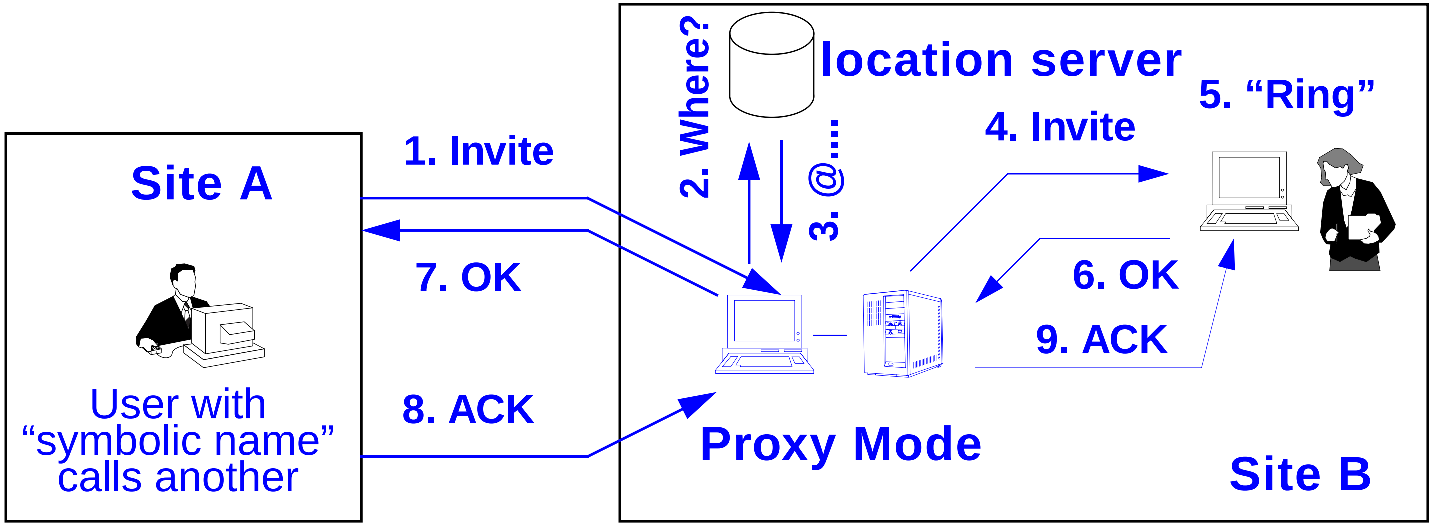


Signaling: Session Initiation Protocol - SIP (2)

Properties

- **IETF driven peer-to-peer signaling protocol**
  - RFC 2543
- **text-based signaling protocol**
  - machine independence
  - header / body comparable to HTTP
- **Based (most often) on UDP, but TCP allowed as well**
  - simplicity, speed

Session



Signaling: Session Initiation Protocol - SIP (2)

Characteristics

- **ASCII (readable) protocol - HTTP vs. SIP**
  - similarities
    - request/response, proxies ...
  - differences
    - H323: server state, server may initiate actions ..
- **control, location and media description**
  - via **SESSION DESCRIPTION PROTOCOL** SDP
- **supports personal mobility**
  - through
    - proxies or
    - redirection
  - email-style address (sip:[user:pw@]host:[port])
- **does not provide full-fledged functionality (i.e. extensible towards)**
  - usage for IP-IP, POTS-IP environments
  - inter-gateway communication
  - interaction with firewalls
  - billing system, ...

$\Rightarrow$ **principle: fast in the core, smart at the edge**

4.1 Session Description Protocol - SDP

intention: to describe multimedia sessions

- session announcement, session invitation

properties: abbreviated message format as ASCII-text

SDP in SIP messages includes description of:

- media to use (codec, sampling rate)
- media destination, IP address and port number
- session name and purpose
- time / period the session is active
- contact information

example (SIP)

```
v=0
o=mgoertz 28944526 2890842807 IN IP4 130.139.83.68
s=ICW SIP
i=A session for voice transmission using SIP
u=http://www.kom.e-technik.tu-darmstadt.de/test.pdf
e=mgoertz@KOM.tu-darmstadt.de (Manuel Goertz)
c=IN IP4 130.233.154.68
t=2873397496 2873404696
m=audio 49170 RTP/AVP 0
m=video 51372 RTP/AVP 31
```

4.2 SIP - Components

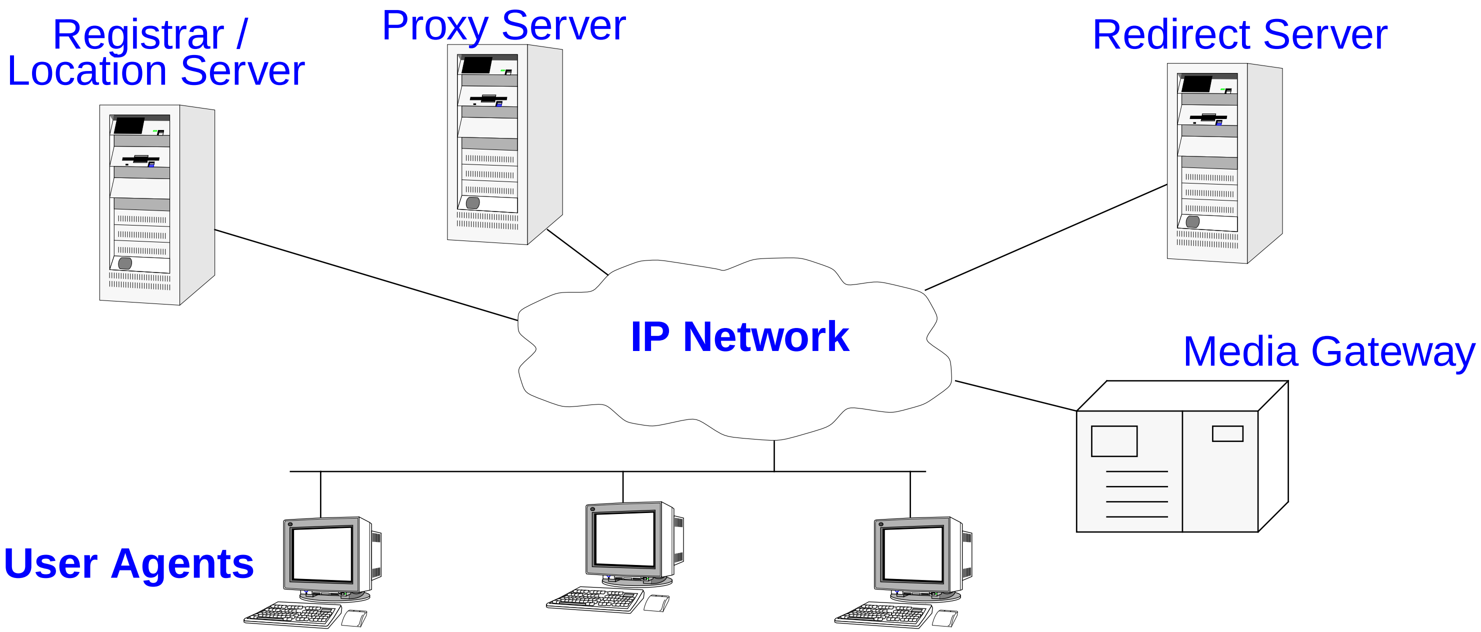
UAC: User-agent client

UAS: User-agent server

redirect server: to redirect requests

proxy server: to fork requests to multiple servers

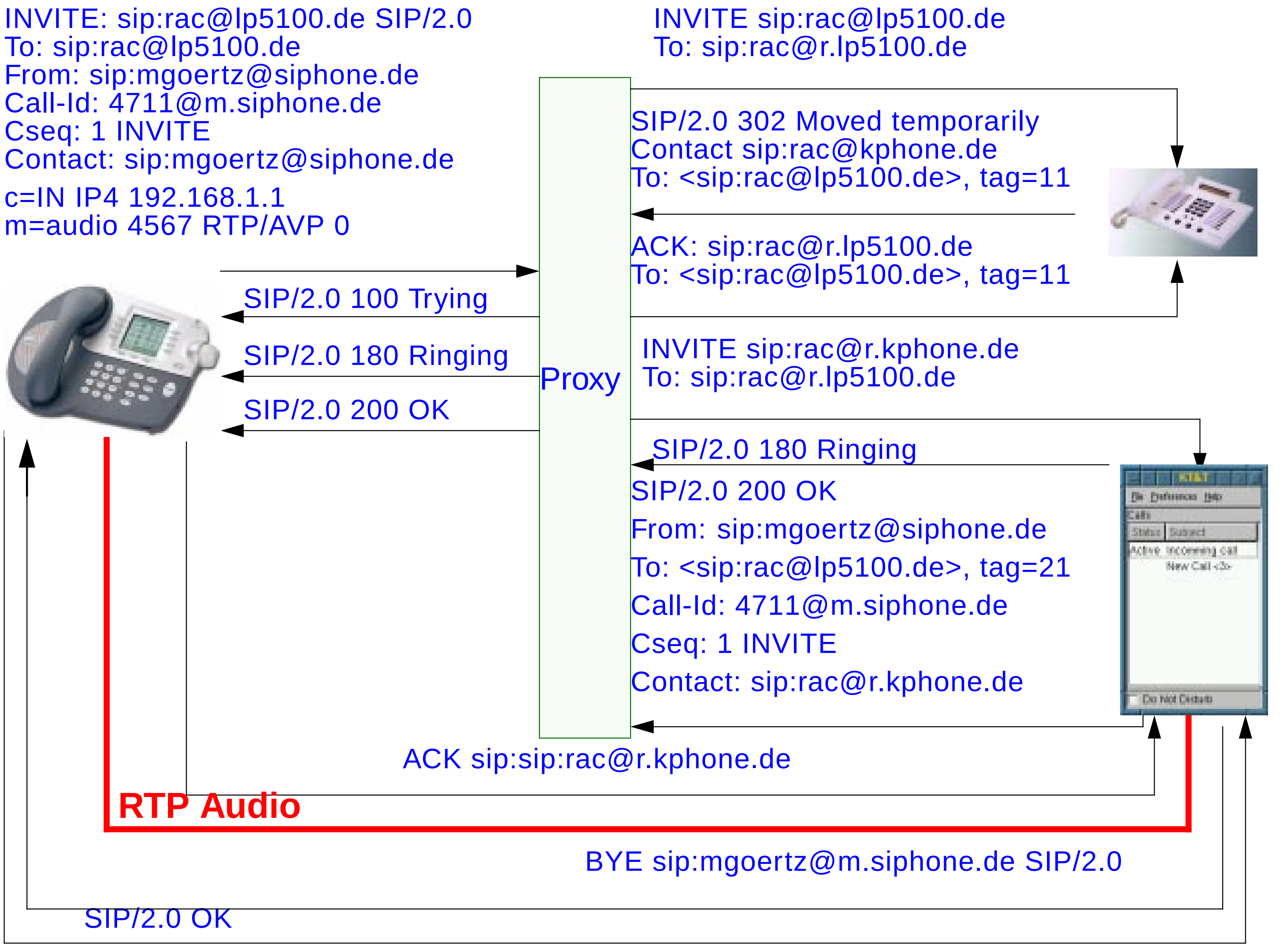
registrar: to track user locations



Usally: user agent = UAC + UAS

Often: registrar and proxy or redirect server combined

4.3 SIP - Example - Control and Data Flow



4.4 SIP - Requests/Methods, Responses, Header e.g.

Requests/Methods

- **REGISTER**
  - registers a user address at a registrar
  - binds a permanent address to the current location
  - registrations can timeout => refresh
- **INVITE**
  - usually begin of a session
  - body contains session description
  - re-Invite to change session state
- **OPTIONS**
  - returns user capability to the requestor
- **ACK**
  - only used in session initiation process
  - end of 3-way handshake
- **CANCEL**
  - terminates a uncomplete request
- **BYE**
  - terminates an open session

SIP - Responses

Characteristics

- **Similar to HTTP - syntax**
  - xxx explanatory text
- **Codes**
  - response codes from HTTP/1.1
  - extended with new codes

Numbering

- **1xx : Informational**
  - 100 Trying
  - 180 Ringing
- **2xx : Success**
  - 200 OK
- **3xx : Redirect**
  - 302 Moved temporarily
- **4xx : Client error**
  - 400 Bad Request
- **5xx : Server error**
- **6xx : Global failure**

SIP - Header

E.g. Invite Request

INVITE sip:mgoertz@sip.de SIP/2.0

Via: SIP/2.0/UDP gw.proxy.com
From: Ralf <sip:rac@phone.de>
To: Manu<sip:mgoertz@sip.de>
Call-ID: 1234@gw.proxy.de
CSeq: 1 INVITE

Subject: IP Telephony Tutorial at ACM MM 2001

Content-Type: application/sdp
Content-Length: 400

blank

v=0
o= gw 881300437 IN IP4 192.168.0.2
c=IN IP4 gw.proxy.de
m=audio 3456 RTP/AVP 0 3 4 5
...

Start Line

SIP Header

Request Header

Entity Header

Header

SDP Data

Body

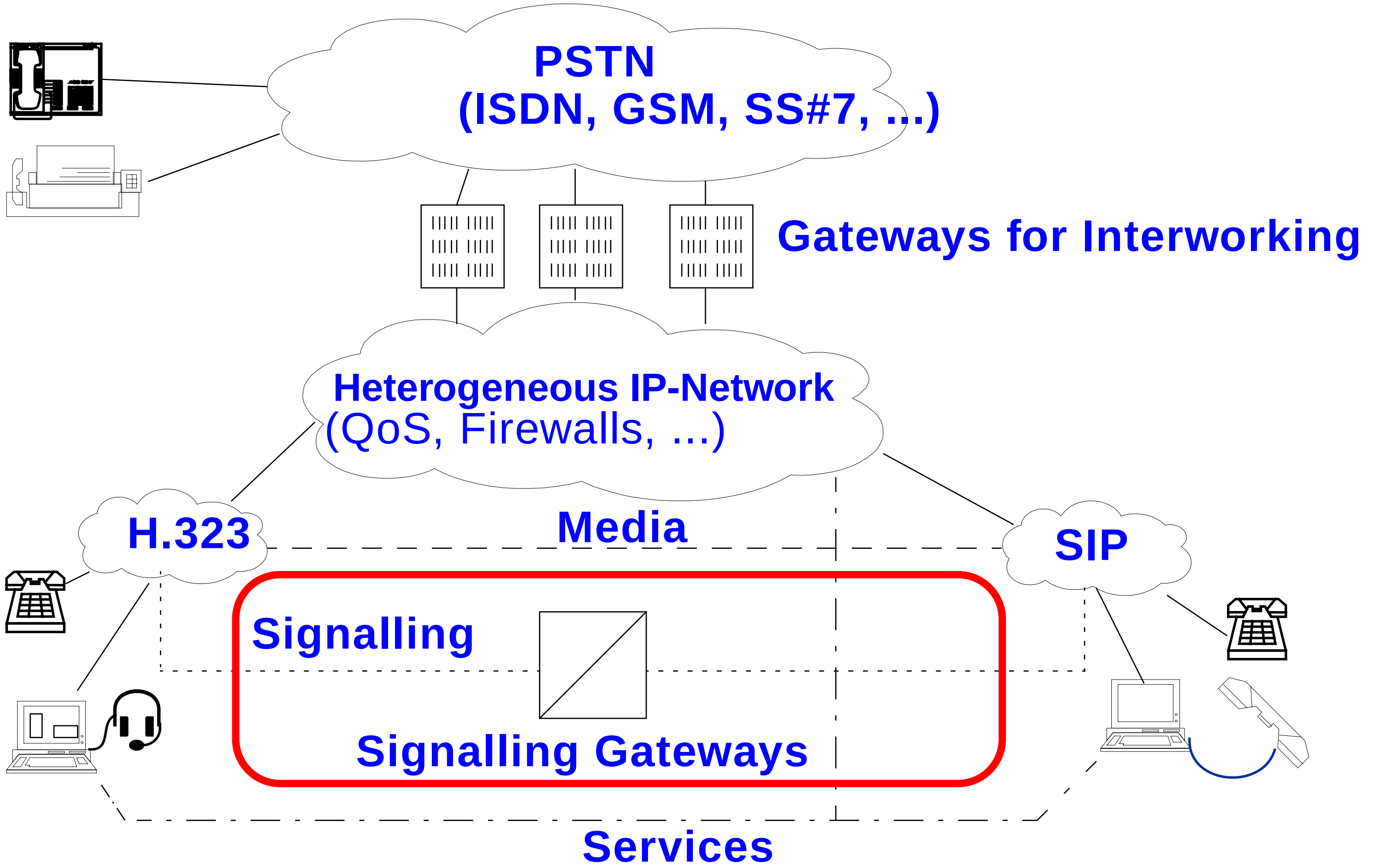
SDP - Example within SIP Header

Structure of a SDP message, e.g.

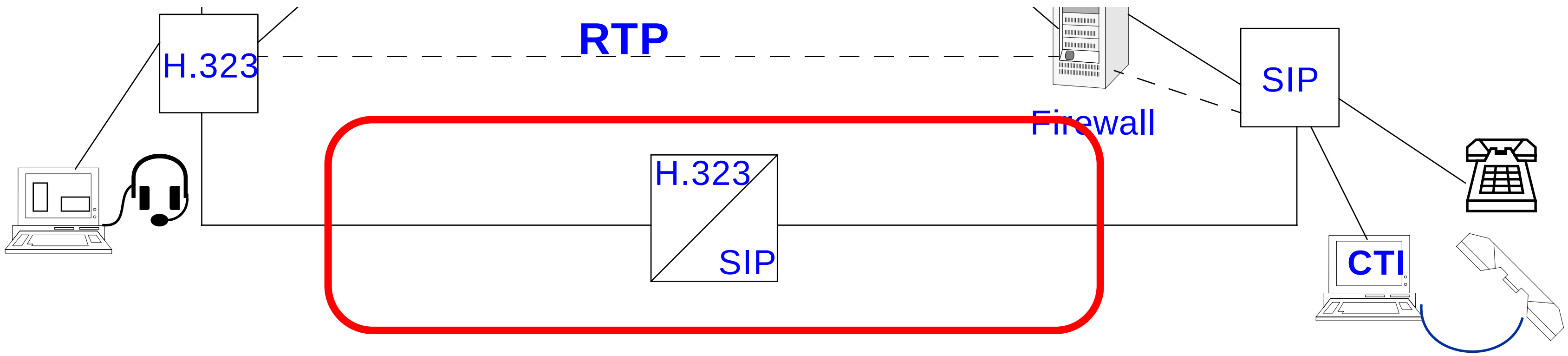
```
v=0
o=rac 28944526 2890842807 IN IP4 130.139.83.68 s=test
i=A session for voice transmission using SIP
u=http://www.kom.e-technik.tu-darmstadt.de/test.pdf
e=rac@KOM.tu-darmstadt.de (Ralf Ackermann)
c=IN IP4 130.233.154.68
t=2873397496 2873404696
m=audio 49170 RTP/AVP 0
m=video 51372 RTP/AVP 31
```

5. H.323 + SIP

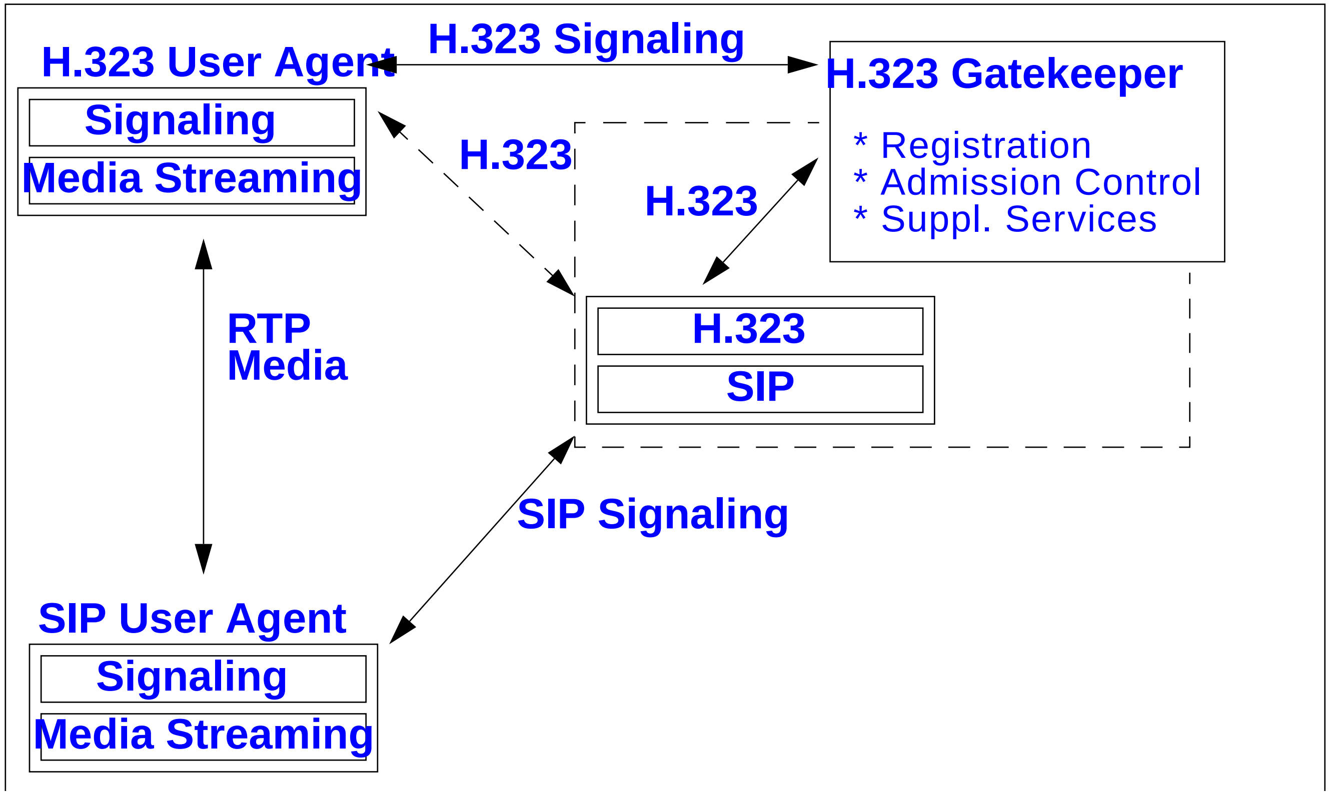
Signaling Protocols Overview



H.323 + SIP (2)



Mapping and Gatewaying: H.323/SIP Gateway



H.323 + SIP: Cases (2)

Case: H.323 Participant calls SIP participant

- **using Gatekeeper telephone number or alias (e.g. 100)**
- **mapping to SIP URL is done**
  - at the gateway based on a loadable mapping / registration list

Case: SIP Participant calls H.323 participant

- **has to address the Gateway at port 5060**
  - physically by means of
    - IP address or symbolic name
- **uses SIP URL:**
  - sip:<number or alias>@<siph323gw>:5060
- **mapping to H323 participant by using the heading part of the SIP URL**
  - <name or alias>
  - other mappings could be integrated

H.323 + SIP: Some Experiences (3)

Background

- **based on own implementation**

Limitations, i.e. to be considered ..

- **capability set limited to G.711 64k μlaw, due to**
  - limited mapping of H.323 capabilities to SIP AVP profiles and
  - limited capabilities of example SIP client
- **SIP INVITE for H.323 originated calls based on**
  - empty (concerning description of endpoint IP address and RTP ports) initial INVITE
  - transmission of this information as part of the ACK
    - after adequate information has been received from the H.323 side (as described in Gateway Draft)
- **does not actively register with a SIP server**
- **SIP and H.323 messaging characteristics**
  - based on internal (OpenH323, Vovida)

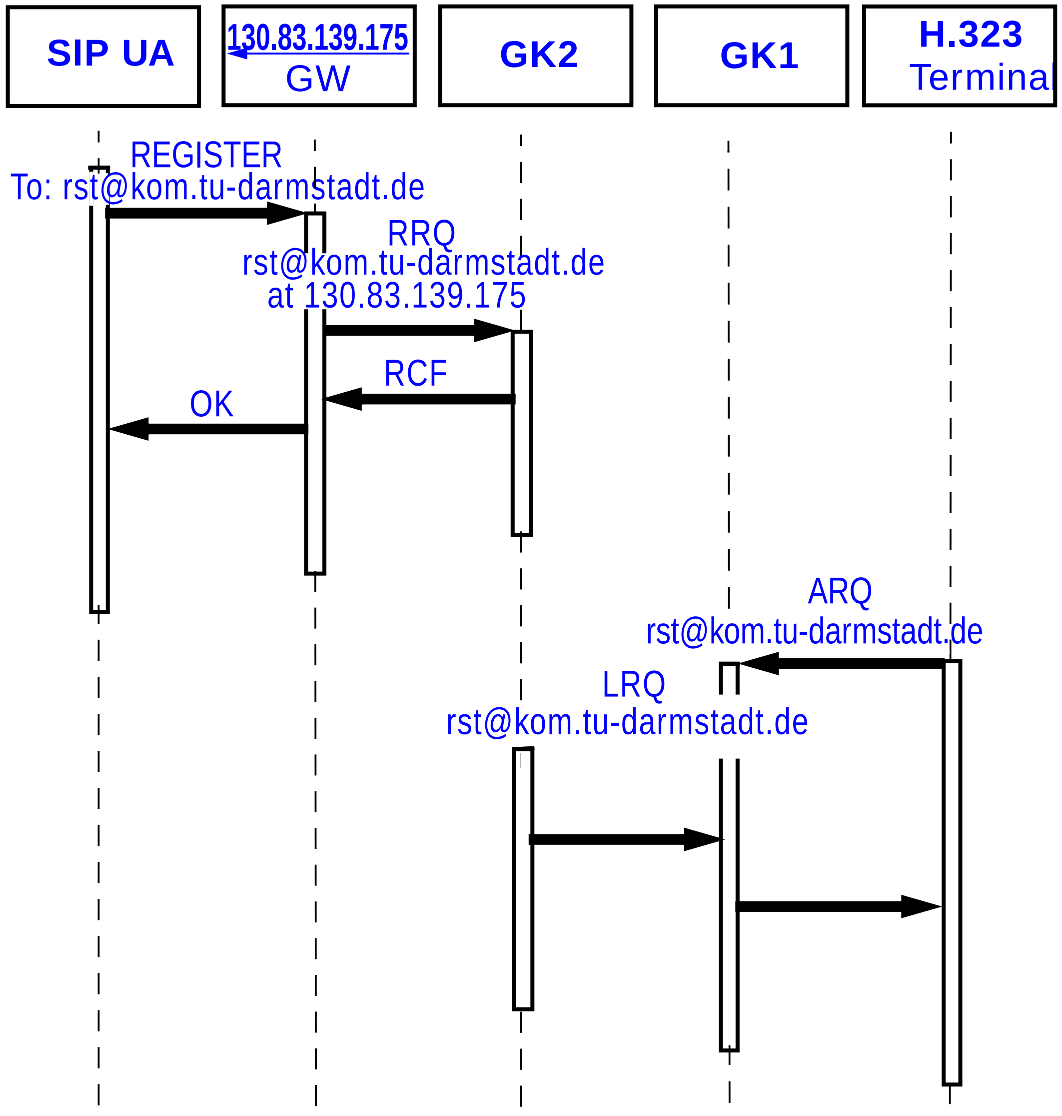
Gateways between Signaling Protocols - SIP/H323

Work in progress

- **SIP integration**
- **SIP - H.323 interaction**

i.e

- **building SIP-terminals**
- **integrating**
  - SIP with
  - conventional PABXs
- **bridging towards H.323**
- **integrating**
  - comprehensive signaling (QoS, security) into the process



6. Media Gateway Control Protocol (MGCP)

Goal

- **to bridge current circuit-based PSTN with IP-based protocols**

Characteristics

- **stateless**
- **functions in form of sequences of commands**
- **header and session descriptor**
- **session descriptor**
  - set of parameters
  - set up endpoints to produce or recognize media formats
  - uses SDP (like SIP)

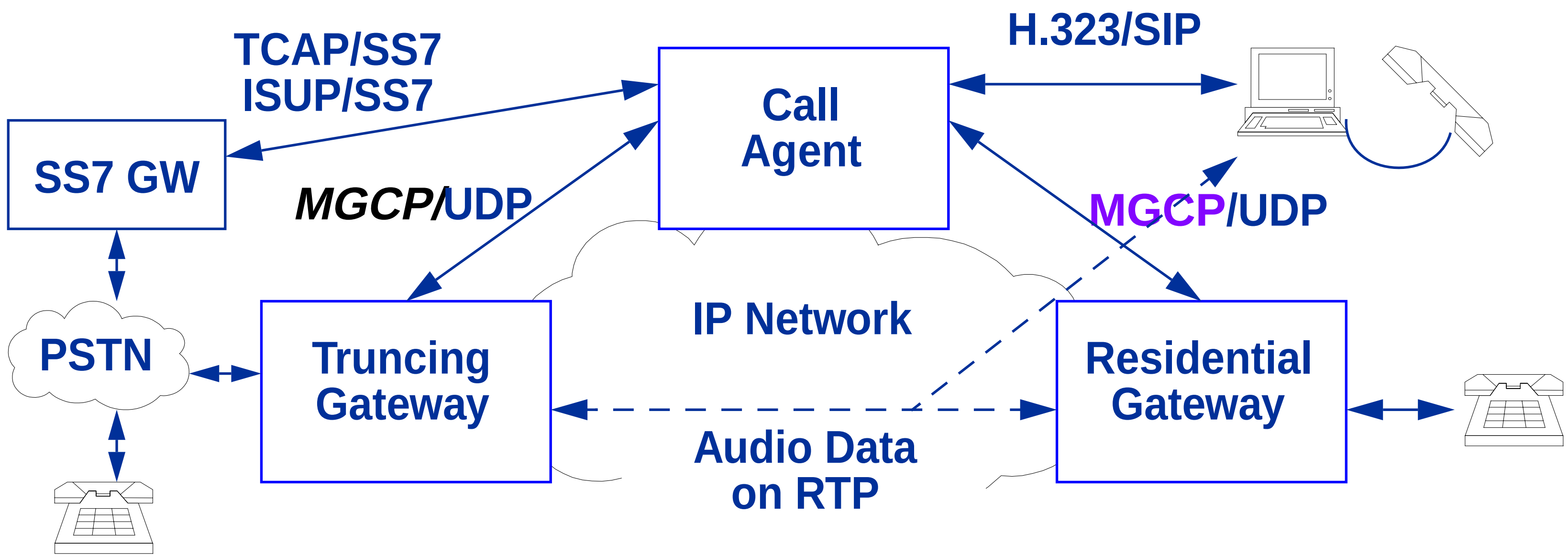
example: RQNT packet format

IP Header
UDP Header
MGCP Header Packet Type = Command Command = Notification Request [RQNT] Transaction Identifier = 68 Endpoint = aaln/2@192.168.12.2 Version = MGCP 0.1 Parameter = Request Events [R:]

Media Gateway Control Protocol (MGCP) (2)

Approach: to centralize intelligence

- in a call agent
- not in the Truncing (TGW) or Residential (RGW) Gateways



Source: MeGaCo Working Group

Features

- (simple) UDP based protocol (evolved from SGCP) for managing
  - endpoints
  - connections (which are described using SDP) between endpoint
  - new services may be implemented by augmenting the Call Agent

Media Gateway Control Protocol (MGCP) (3)

Megaco/H.248

- **joint effort by IETF and ITU-T**
- **media gateway control protocol**
- **for use in distributed switching environments**
- **set up media (for example, voice traffic) paths through the distributed network.**

Properties

- **signaling logic is located on Media Gateway Controllers (MGCs)**
- **media logic is located on Media Gateways (MGs)**

7. Operation

Security Issues

Confidentiality	• ensures privacy	(partial) encryption of control and media streams
Integrity	• ensures correctness of control information • e.g. alerting, call routing, bill • especially necessary for signaling, control traffic	cryptographic protection of message diggest
Authentication	• identity of com. partners • e.g. both, subscribers & service providers • prevents “spoofing”	(e.g. PIN, TAN) cryptographic protocols, Public Key Infrastructures (PKI)
Access Control	• who is allowed to access what part(s) of a system	system specific mechanisms, (media) firewalls
Non Reputation of Origin	• basic mechanisms for provider/customer relationships	see above

Security Issues (2)

Customer requirements

- **accurate and verifiable accounting**
- **confidentiality and privacy**
- **protection of anonymity**

Threats through

- **one network for**
  - (in-band) signaling
  - voice data
  - management of network components
- **signaling and service provided by decentral systems**

Security Issues (3)

Service provider requirements

- **mainly protection against corruption and fraud**

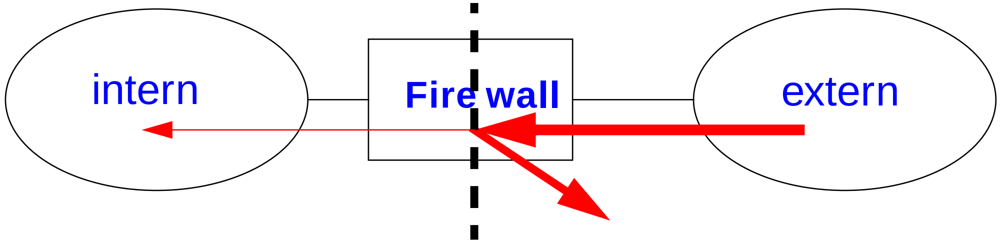
⇒ **Necessity for:**

- IP-telephony specific security
- security for services

Standardization Effort

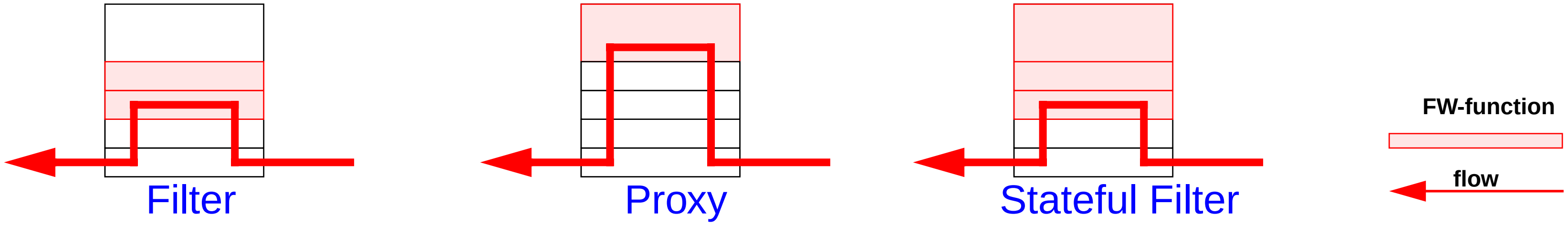
- **confidentiality**
  - H.3xx / H.235
    - RTP data encryption (DES, Triple DES, RC2)
    - key exchange in signaling messages (H.245)
  - IETF / SIP
    - RTP data encryption (DES, Triple DES, RC2)
    - key exchange is part of the SIP message bodies
- **authentication**
  - H.3xx / H.235
    - TLS or IPSec
  - SIP
    - authorization header field - digital signature

7.1 Specific Problem - Firewall Interaction



Firewall Functions

- permit/deny dedicated data flows
- assignment of dedicated data flows to users/systems
- hiding internal structures
- monitoring, logging and alerting



Firewall components and architectures

- components: filters, proxies, stateful filters, ...
- architectures: DMZ, inbound filters, dual homed gateway, ...

⇒ **Well defined functions, but many possible architectures**

Firewall Interaction

Facts

- (all) organisations use firewalls
- IP-telephony applications/protocols differ from “standard applications”
- firewalls have problems to support IP-telephony applications/protocols
- IP-telephony is emerging to a carrier grade service

⇒ **Firewalls have to support IP-telephony to allow the deployment**

⇒ **Failure prevents deployment**

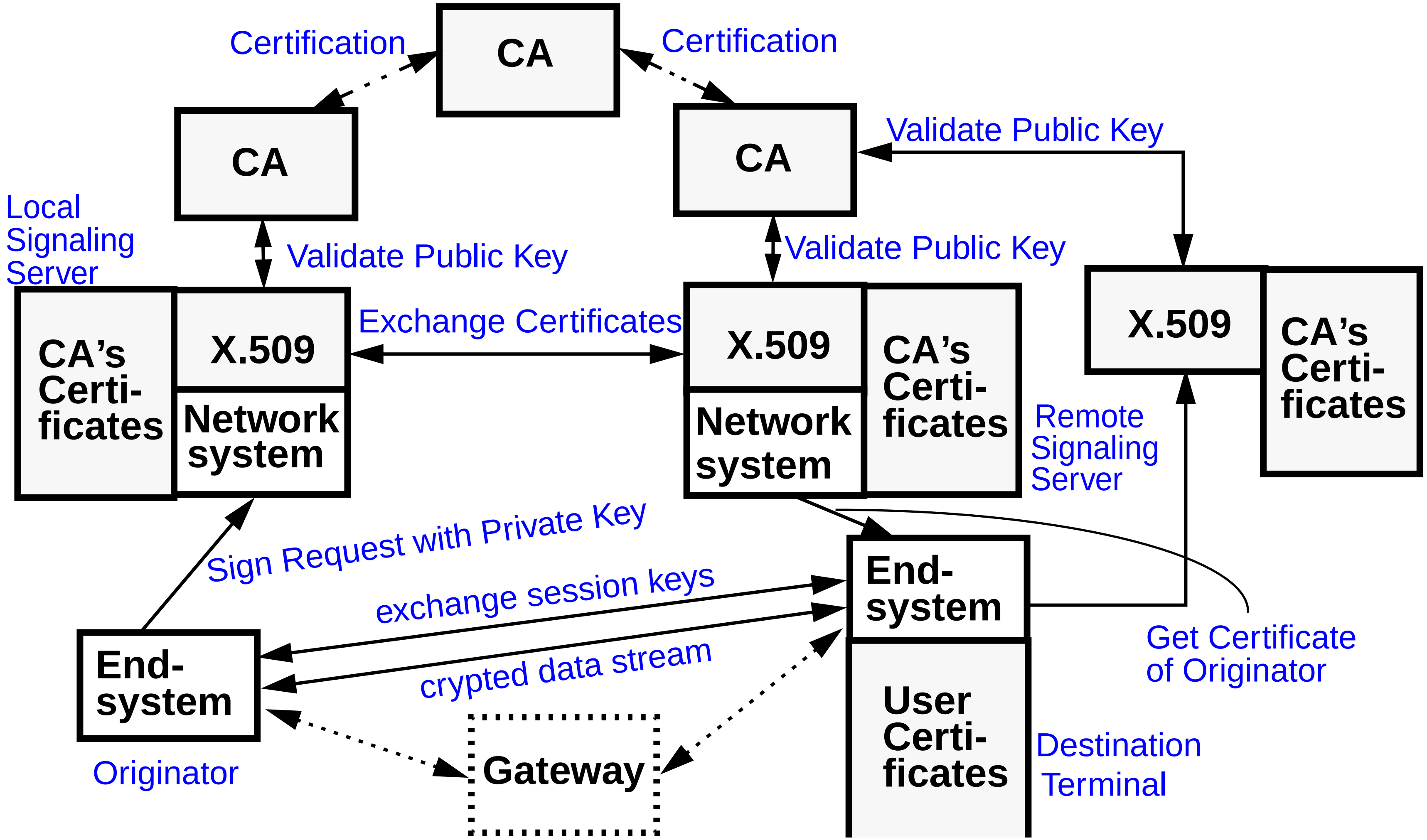
IP-Telephony Firewall Interaction

- **SIP (draft-rosenberg-sip-firewalls-00.txt)**
  - problem domain
    - getting SIP and RTP through firewalls
    - IP address for the media streams in SIP bodies
  - solution suggestions
    - application layer firewall that understands SIP
    - packetfilter with SIP proxy
- **H.323 (draft-shore-h323-firewalls-00.txt)**
  - problem domain
    - dynamic port negotiation
    - very complex structure due to ASN.1 and PER
  - solution suggestions
    - realm specific IP (RSIP)
    - packetfilter with H.23. parser
- **vendor specific solutions/proposals**
  - complete solutions are missing
- **IETF**
  - foglamps
  - put intelligence where the knowledge about the used protocols is:
    - in the endsystems

7.2 Specific Problem: Privacy & Authentication Support

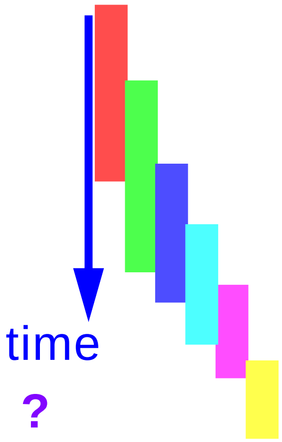
Deploying security services based on Public Key Infrastructures

- **from "Trust by Wire"**
- **to "Trust by Authentication"**



8. Value Added Services

Basic Connectivity –> ..... –> Value added services



- interoperating terminals
- IP Telephony in and between **INTRANETS**
- Interoperable and interacting gateways
- IP Telephony enhanced / based PBXes
- IP Telephony as common service in the **INTERNET**
- **NEW SERVICES** and related applications

Major difference to POTS

- **telephony service e.g.,**
  - H.450 like features
  - click to dial & Web callback
- **new services, e.g.**
  - unified messaging
  - Web assistance
  - voice access to content
  - application sharing
  - electronic whiteboard

Value Added Services (2)

Mobility

- wireless (moving while communicating)
- nomadic (go off-line and move, then go on-line anywhere)
- be mobile

Personalization

- user profiles
- context awareness (i.e. situation defines/provides for facilities)
- group communication & multiple media functions
- support for various stati
  - reachable
  - temporary busy
  - permanent absent

Protection of investment

- independence of
  - endsystems, terminal equipment
  - destination medium (fax, sms, eMail)
  - location
  - physical number

Value Added Services (3)

Workflow integration

- **well-known in working processes**
- **integration of communication methods in workflows**
- **gateways to third party products (SAP R/3)**

Service-Gateways

- **coexistence**
  - transition to convergence
  - bridge between PSTN and IP-networks
  - integration into workflow

$\Rightarrow$ **It is more important to reach to the partner than to know where he/she is!**

8.1 Interfaces

TAPI (Telephony Application Programming Interface)

- TAPI standard was created jointly by Microsoft and Intel.
- in a TAPI environment, the physical connection is made at the desktop level
- only for WinX platforms
- packaged in a dynamic link library (DLL)
- mainly for CTI
- hardware independence

Telephony Service Provider Interface

- interface for hardware drivers (= service provider)
- access for TAPI applications to the hardware

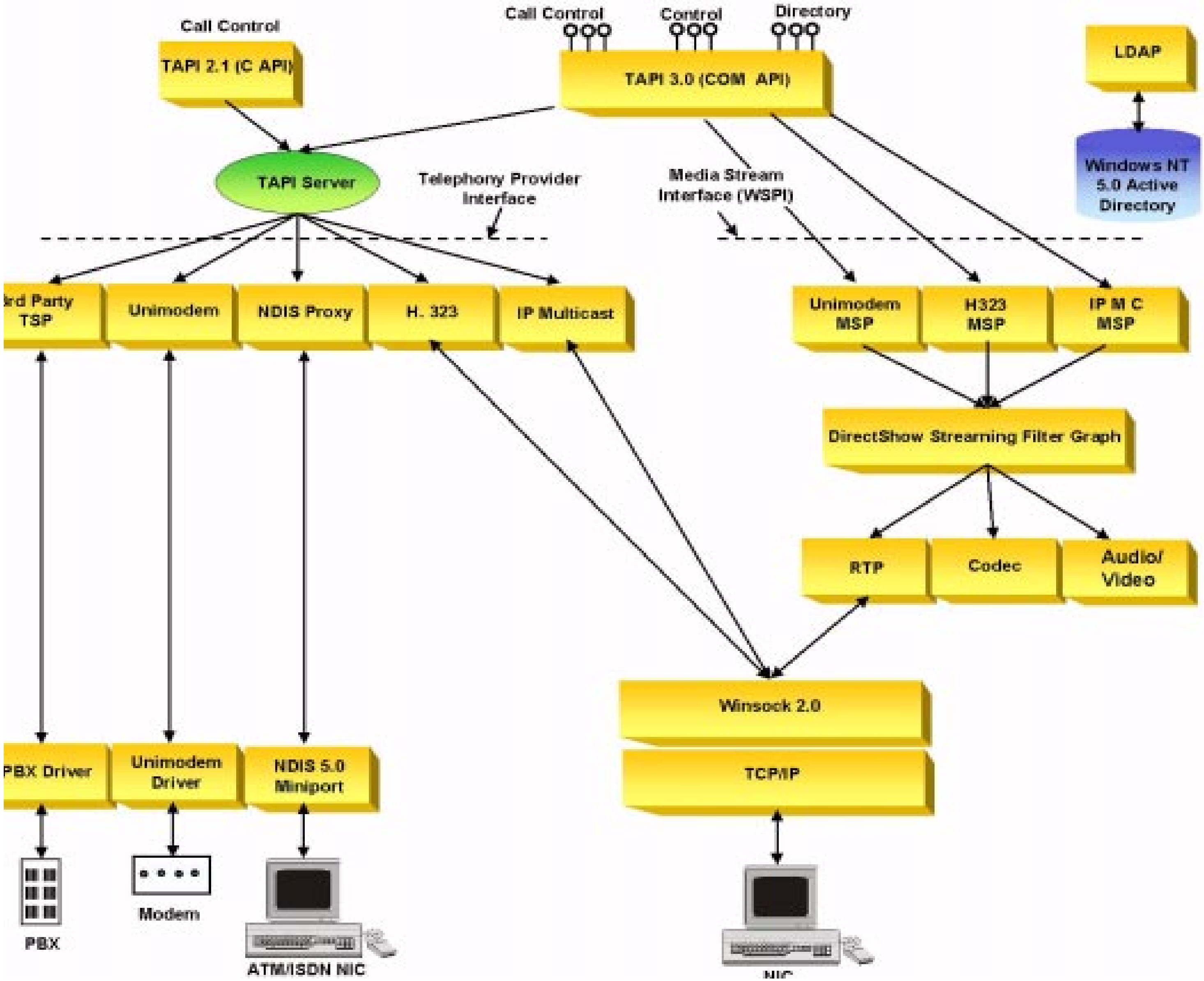
TAPI (Telephony Application Programming Interface) (2)

Version 3.0

- user-level API
- enhanced CTI methods
- H.323 based
- QoS support
- COM model
- built-in Internet Protocol Telephony (IPT) components for creating new-world call center applications

TAPI (Telephony Application Programming Interface) (3)

Architecture



JTAPI (Java Telephony API)

- **object orientated interface**
- **only the description not an implementation**
- **platform independent**
- **located between application and proprietary API**

Java Applet		Java Program	
Java Implementation			
Java Runtime			
MS TAPI	TSAPI	Dialogic	other
Telephone hardware			

- **applets**
  - Web callback
  - predictive calling

JTAPI (Java Telephony API) (2)

Architecture

- **modular**
- **extendable**

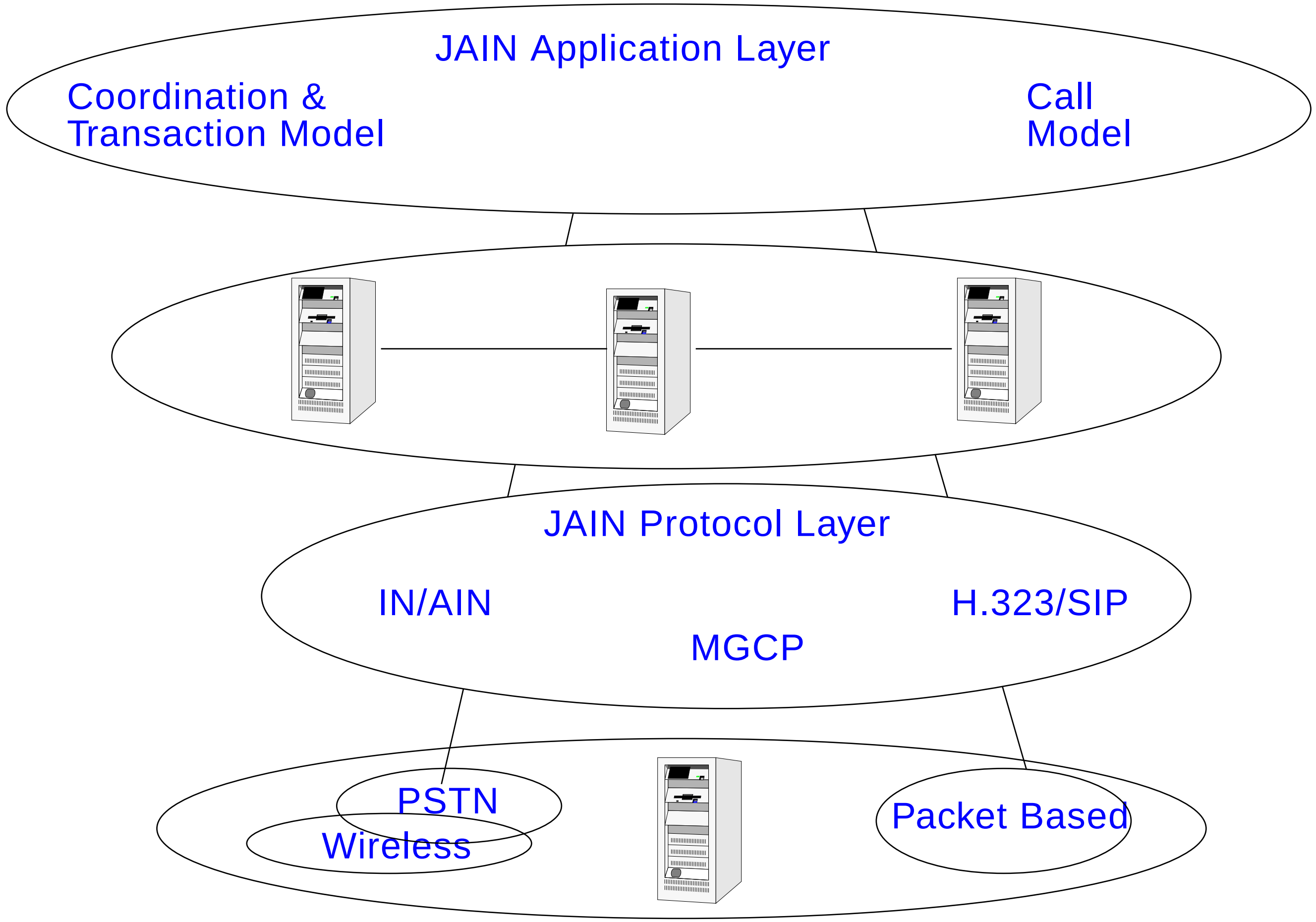
JTAPI 1.3

- **class packages**

Modul	Packet
Core	javax.telephony
Call Control	javax.telephony.callcontrol
Call Center	javax.telephony.callcenter
Media	javax.telephony.media
Mobile	javax.telephony.mobile
Phone	javax.telephony.phone
Private Data	javax.telephony.privatedata

JAIN (Java API for Integrated Networks)

- integration of Internet and intelligent networks



- new abstraction layer
- removal of proprietary roadblocks
- set of software component library

CSTA Computer Supported Telecommunication Applications

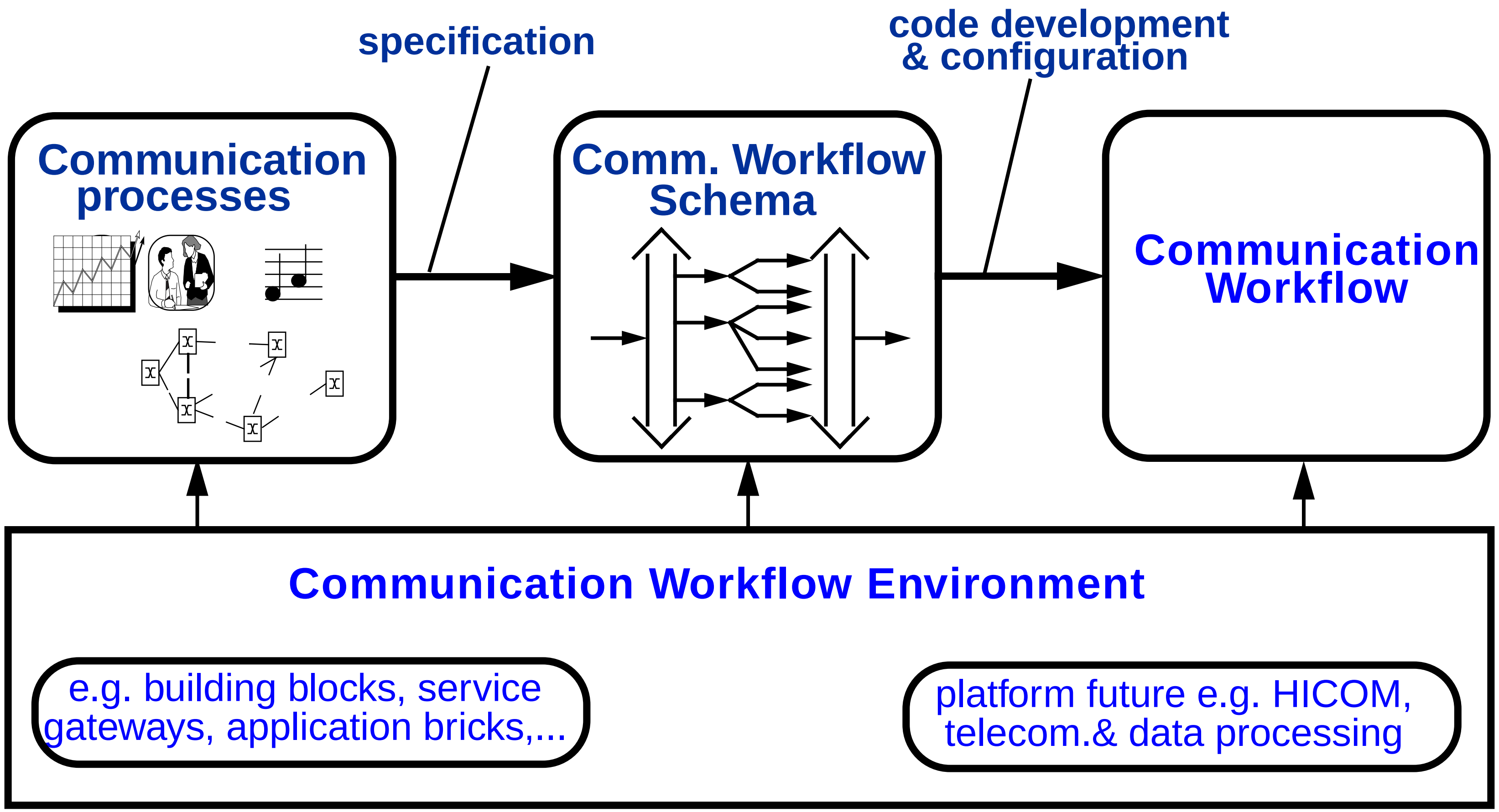
- **CSTA is a CTI interface**
  - provides access to telecommunication functions
  - can be used with APIs, such as TAPI, JTAPI, JavaPhone, CallPath, TSAPI
  - may be used by 3rd party applications

Task Group, ECMA TC32-TG11

- **development and refinement**
  - a standardized Computer-Telephony Integration (CTI) interface
  - interactions between computer applications and the telecommunications network

8.2 Communication Workflow

Target scenario

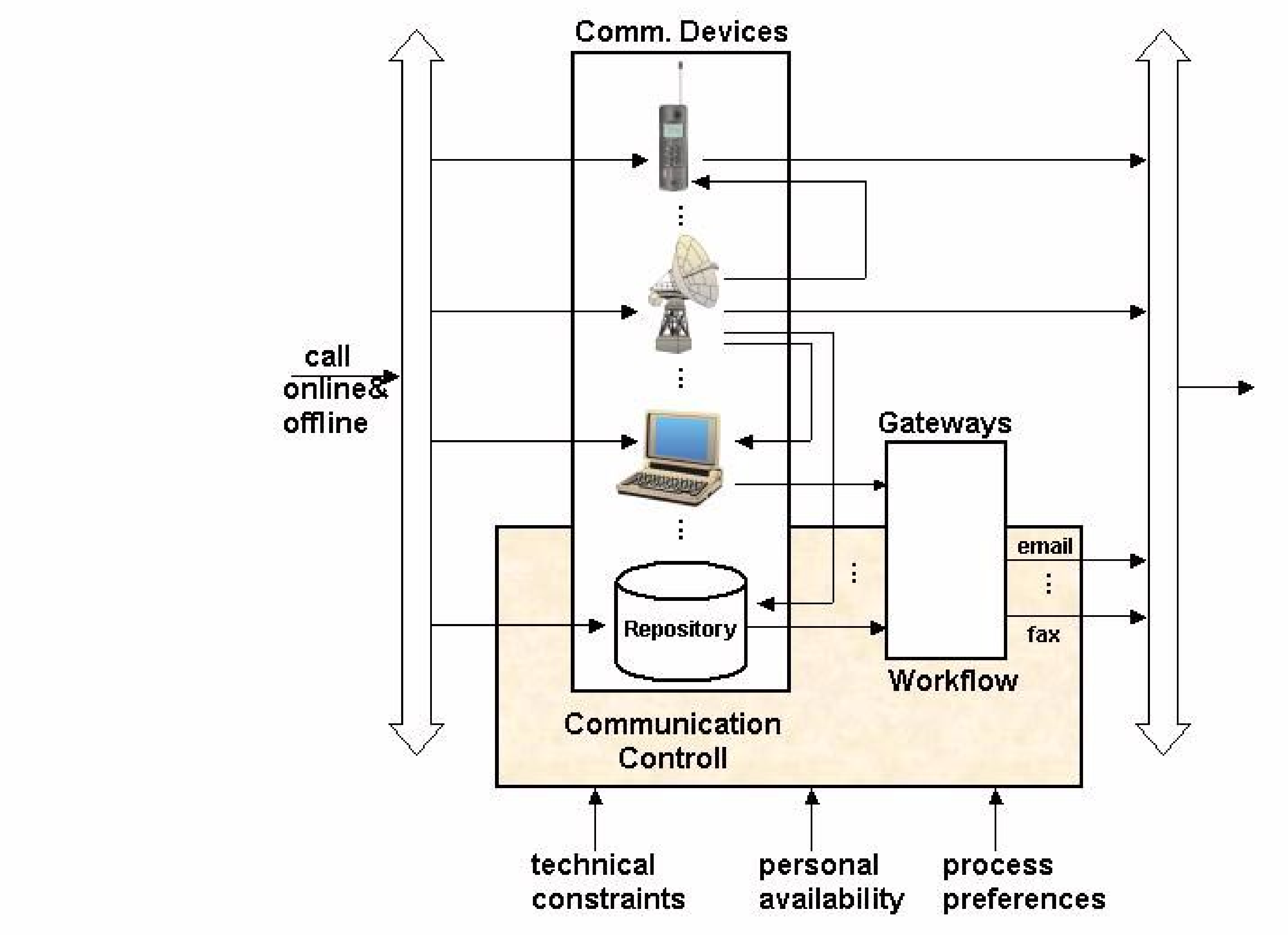


A communication workflow represents the seamless execution of a communication flow.

- **it is created by cross media integration using well defined & well known communication paradigms & devices support by the PBX**

Communication Workflow: Example of a Communication Process

Communication Workflow:



9. Products and Prototypes

Broad range

- **diverse functionality**

Communication Workflow: Experiences

IP-Tel means to work in

- **research**
- **development**
- **deployment**

And available / research

- **Software Only Systems**

And e.g., in Darmstadt

- **actual**
  - Major joint IP-Tel-PABX Infrastructure Project at Darmstadt
  - Open Source Developments
- **completed projects**
  - Darmstadt Example: Interaction with Firewalls
  - Darmstadt Example: MBone2Tel-Gateway - Concept
  - Darmstadt Example: A “Virtual PBX”
  - Darmstadt Example: QoS & Charging via RSVP

9.1 Software Only Systems

E.g. Skype

`http://www.skype.com`

- **like**
  - Instant Messaging
  - KaZaA, P2P (peer-to-peer) technology
    - to connect you to other users to talk and chat
- **very good audio quality**

9.2 Major joint IP-Tel-PABX Infrastructure Project at Darmstadt

Goals

- to make extensive use of IP Telephony (field test on a broader basis)
- to operate & to research on PABX(es) and attached IP Telephony
- to install IP-phones with the following expected amount
  - starting with few islands of 30 to 100 phones
  - via 10% IP-phones (600)
  - towards 33% (2000)

Customers (about 6000 phones)

- University TUD and attached institutions (FHD, ...)
- research institutes (Fraunhofer-IPSI , Fraunhofer-SIT)

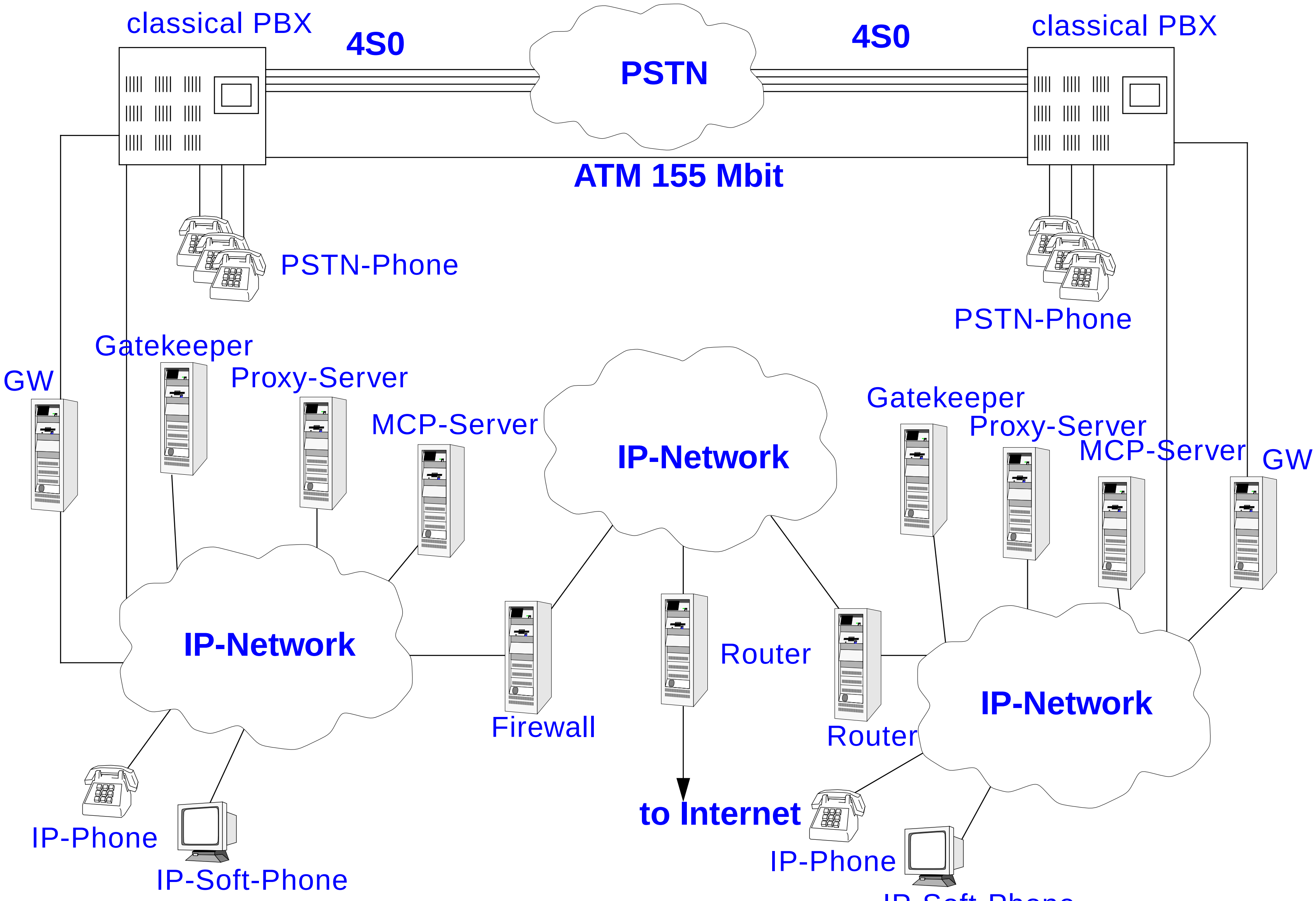
Constraints (some)

- 1-2 years timeframe of replacement
- operational <u>and</u> R&D (continuous enhancements)
- larger scale field trial

Status

- replacement of the PBX for the scientific institutions on the way
- 6000 telephones to be replaced and a planned

Test-Scenario



9.3 Open Source Developments

As always - there's (expensive) commercial equipment

- **Dialogic, RADVision, ...**

But - there's Open Source software as well - and it's growing rapidly

- **dedicated Hardware**
  - Quicknet PhoneJack/LineJack
    powerful codecs, "plain old phone" attachment and line interface

  `http://www.quicknet.net`

Protocol Stacks

- **just the major**
  - OpenH323 - having Voxilla as Netmeeting counterpart

  `http://www.openh323.org`
  - Vovida (SIP, H.323, RTP, MGCP)

  `http://www.vovida.com`

User Agents / Terminals / IP PBXs

- **Asteriks, 8x8, ...**

APIs

- **MS-TAPI (having powerful enhancements in its 3.0 version)**
- **JTAPI, ...**

9.4 Darmstadt Example: Interaction with Firewalls

Open Source - KOM

- **Contact: Utz Roedig (utz.roedig@KOM.tu-darmstadt.de)**
  `www.kom.e-technik.tu-darmstadt.de/KOMproxyd/`

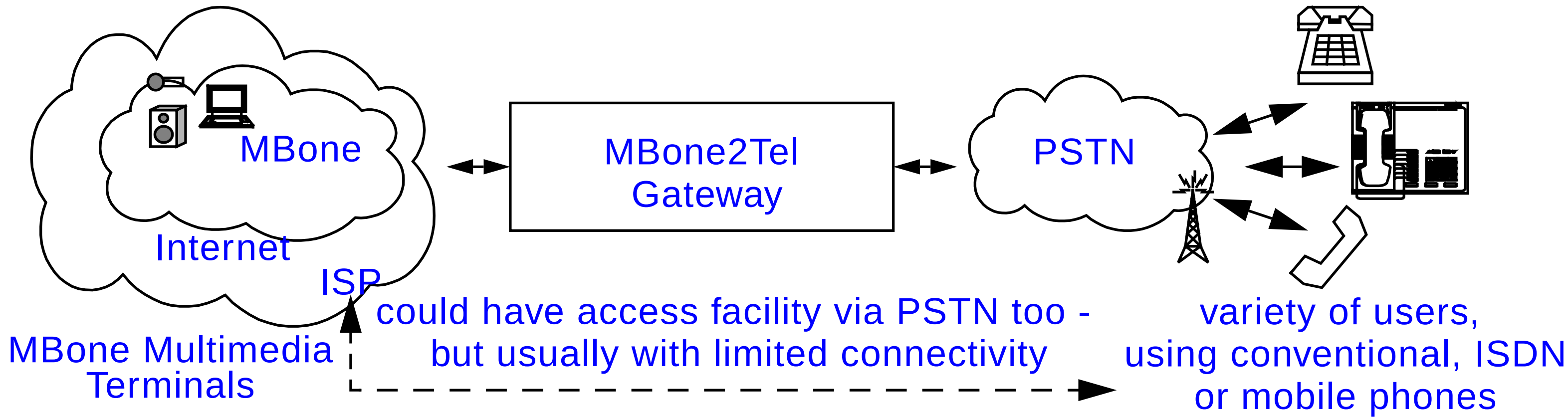
TECHNISCHE UNIVERSITÄT DARMSTADT

KOM Multimedia Communications
Downloads

KOMproxyd

- **set of modules**
  - for the integration of external proxies in a distributed firewall system
    - based most often on standard components
- **for H.323**
  - tested and in operation

9.5 Darmstadt Example: MBone2Tel-Gateway - Concept



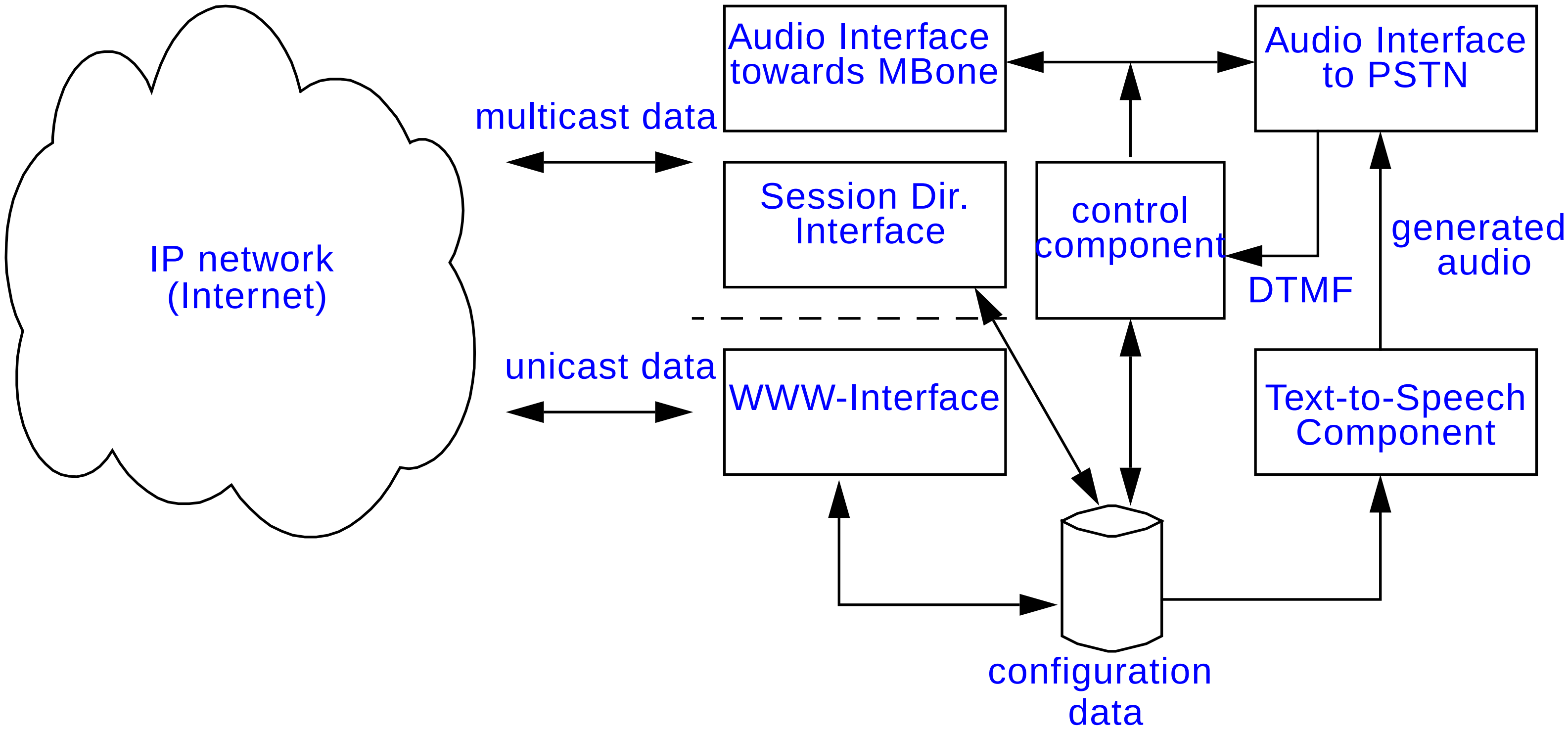
Basic Idea:

- deploy gateways as point of interaction
- intentionally use specifics / advantages of the connected areas
- deploy Value-Added Services

Basic Functionality:

- users may passively or actively take part in MBone audio conferences using their conventional telephone

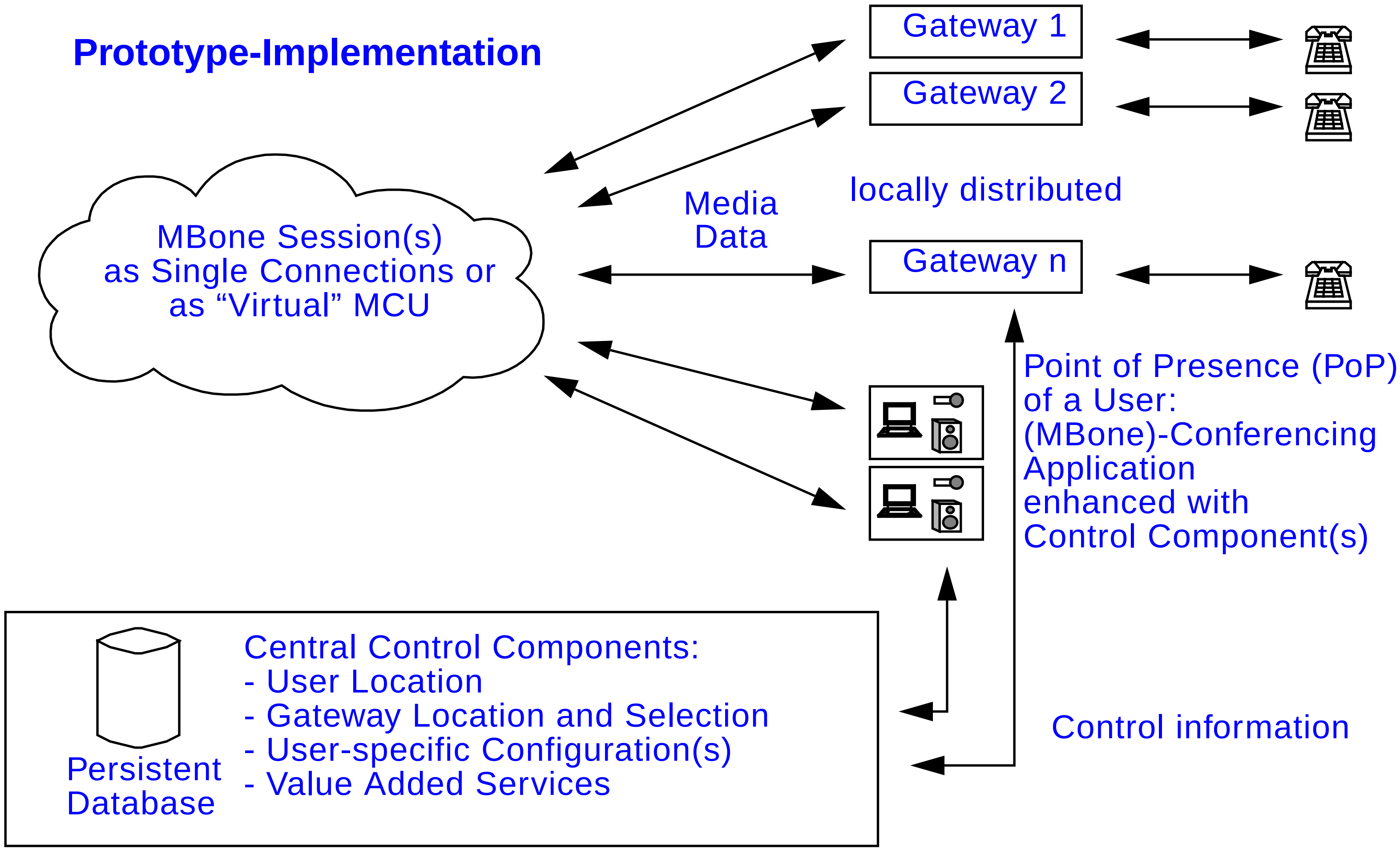
Example: MBone2Tel-Gateway - Components



Architecture

- modular
- based on well-evaluated / evolving components
- using a simple graphical user-interface
- building blocks for “Value Added Services” and “Gateway Farm”

9.6 Darmstadt Example: A “Virtual PBX”



Basic idea:

- **have working scenario and testbed / incorporate standard protocols !**
- **enhance existing PBX**

9.7 Darmstadt Example: QoS & Charging via RSVP

FACTS ARE E.G.

- **RSVP can be augmented by flexible charging mechanisms.**
- **an RSVP implementation can handle a sufficient number of sessions**
  - (i.e. over 50000 based on off-the-shelf hardware).

Charging Mechanisms and Price Calculation

- **design of charging mechanisms based on formal model**
- **development of calculation models**
- **application of charging mechanisms**

Internal Design of Protocol Engine

- **innovative design & implementation**
- **potential for improvements shown (and implemented)**
- **performance gains experimentally verified**

Performance for a Large Number of Unicast Sessions

- **experimentally demonstrated**
- **additional result: little impact of flow lifetime**

10. Future Trends: Today & Near future

Approaches
- device and
- technology driven

Network engineering
- reservation and (not vs.) overprovisioning

Comprehensive “umbrella” developments (and standards)
- more than “simple” IP-Telephony
- more than “simple” basic call processing

Voice quality is not perfect

Delays are still too long
- signaling
- audio transfer

Security
- new security issues arise with IP Telephony
- firewalls should communicate with other IP-Tel components

11. Annex

The annex shall help to

- **explore in more detail the explained topics**
- **find out about related topics**

Hence

- **only references to relevant information is provided**

11.1 References

URLs, please look at

- **General**

  `http://www.itu.ch`
  `http://www.ietf.org`
  `http://www.computertelephony.org`
- **H.323**

  `http://www.h323.org`
- **SIP**

  `http://www.sipforum.org/`

Books, please look at

- **IP Telephony - The Integration of Robust VoIP Services, Bill Douskalis,**
- **IP Telephony: Packet-Based Multimedia Communications Systems, Olivier Hersent, et al.**
- **Converged Networks and Services: Internetworking IP and the PSTN, Igor Faynberg, et al.**
- **IP Telephony with H.323, M. Korpi**

References (2)

Papers, please look at

http://www.kom.e-technik.tu-darmstadt.de/Research/Publications/publications.html

http://www.cs.columbia.edu/~hgs/sip/papers.html

Conferences, please look at

http://www.pulver.com

http://www.upperside.fr/basip.htm

11.2 Glossary

please have a look at

http://www.kom.e-technik.tu-darmstadt.de/Teaching/de/literatur/glossare.html

http://www.ucc.ie/info/net/acronyms/index.html

http://www.gateway.de/knowledge/lexikon/

RFC 1983: Internet Users' Glossary (IETF)

http://www.uni-koeln.de/allgemeines/glossar/

http://www.springer.de/iuk/glossar.htm

http://whatis.com

http://www.oreilly.com/reference/dictionary

11.3 Standardization: Some relevant groups

ETF Working Groups and Mailing Lists

- **IP Telephony (iptel) - Call Processing, Service Model, Gateway Attribute Distribution Protocol**
- **PSTN and Internet Internetworking (pint) - Architecture, Services and Management (CTI vs. IN, Click-to-Dial, Click-to-Fax, Access to voice content, ...)**
- **MMusic - Companion Protocols - SAP, SDP, SIP, RTSP, SCCP, (RTP/RTCP)**
- **Mailing List SGCP (Simple Gateway Control Protocol)**
  but, Transition to WG MEdia GAteway COntrol (megaco)

ITU-T SG 16

- **standardization of**
  - multimedia communications between terminals, network equipment and services in LAN and WAN environments
  - without explicit QoS support -
  - H.323 series of standards

www.kom.tu-darmstadt.de
www.httc.de

Standardization: Some relevant groups (2)

ETSI TIPHON - Telecommunications and Internet Protocol Harmonization over Networks

- **3 phases: IP->GSTN, GSTN ->IP, GSTN->GSTN over IP**
- **working groups on:**
  - requirements, security, charging
  - architecture and Reference Configuration
  - call control procedures, information flow, protocols
  - numbering, addressing and naming
  - QoS
  - interoperability testing

International Multimedia Conferencing Consortium (IMTC)

- **-Voice over IP Forum (VoIP)**

Internet Telephony Interoperability Consortium (ITC)

- **- architecture, resource allocation, business strategies**