Voice over IP

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FTW-A0 / IKN

Overview

- Introduction
- State of the Art
  - – Speech Coding and Quality
  - – Signaling
  - – Media Transport
  - – Security
- Research Directions
  - – Example ABE
- Conclusions

Introduction

- “It has not yet been decided whether it is possible to reliably send phone calls over the open Internet or whether it is better to create private IP networks to send them.” (pulver.com)

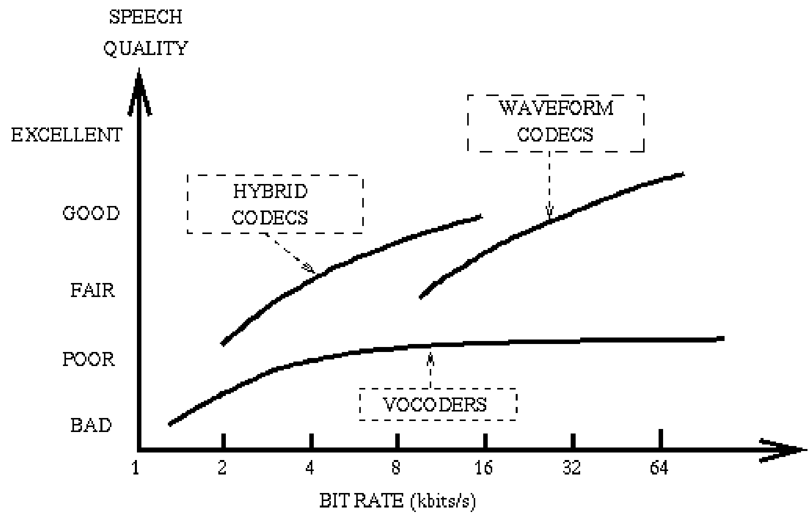
State of the Art

- Speech Coding
  - – Waveform Codecs, Source Codecs, Hybrid Codecs
- Signaling
  - – H.323, SIP
- Media Transport
  - – RTP and RTCP
  - – ABE, SCTP

Speech Coding

- From 64 kbit/s down to 2.4 kbit/s and less
- From full duplex CBR to Spurt/Gap Bursts
  - about 40 to 50 % bandwidth reduction

Common Classes of Codecs



Waveform Codecs - Time Domain

- Pulse Code Modulation (**PCM**)
  - simple sampling and quantizing
  - approx. to logarithmic quantizer: A-law or µ-law
- Differential PCM (**DPCM**)
  - prediction; code only differences
- Adaptive Differential PCM (**ADPCM**)
  - adaptive; change to match speech characteristics

Waveform Codecs - Frequency Domain

- Sub-Band Coding (**SBC**)
  - split frequency bands („sub-bands“)
  - code according to perceptual importance
- Adaptive Transform Coding (**ATC**)
  - fast transformation, like discrete cosine transformation (**DCT**)
  - large number of frequency bands

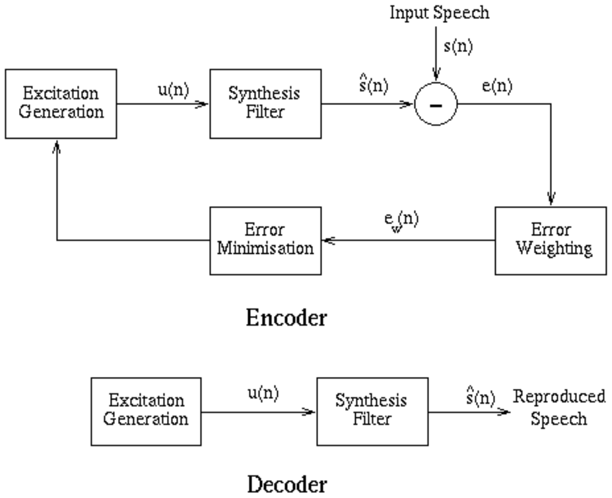
Source Codecs

- also called vocoders
- use a model of how the source was generated
  - simple vocal tract representation
- extract and code parameters of this model
- very low speed (around 2.4 kbit/s)

Hybrid Codecs

- time domain Analysis-by-Synthesis (**AbS**) Codecs
  - most successful, commonly used
- Multi-Pulse Excited (**MPE**), later Regular-Pulse Excited (**RPE**)
  - GSM: simplified RPE Codec at 13 kbit/s
- Code-Excited Linear Predictive (**CELP**) Codec

Analysis-by-Synthesis (AbS) codecs



Low Bitrate Codecs

- Multi-Band Excitation (**MBE**) Codecs
- declaring some regions in the frequency domain as voiced, others as unvoiced
- 3 to 4 kbit/s for „good quality“ speech

Some Standard Codecs

- **G.711** PCM A-law, µ-law
  - 64 kbit/s, “standard reference”
- **G.721, G.726, G.727** ADPCM Codecs
  - 16 to 48 kbit/s
- **G.728** Low Delay CELP Codec
  - 16 kbit/s
- **G.729** CS-CELP (Conjugate Structure)
  - 8 kbit/s
  - Annex B: Voice Activity Detector

Speech Quality Factors

- Delay
  - below 100 to 300 ms
- Echo
- Clarity
  - Intelligibility, Noise, Fading, Cross talk
  - Packet Loss, Bandwidth, Compression
- Delay-Jitter

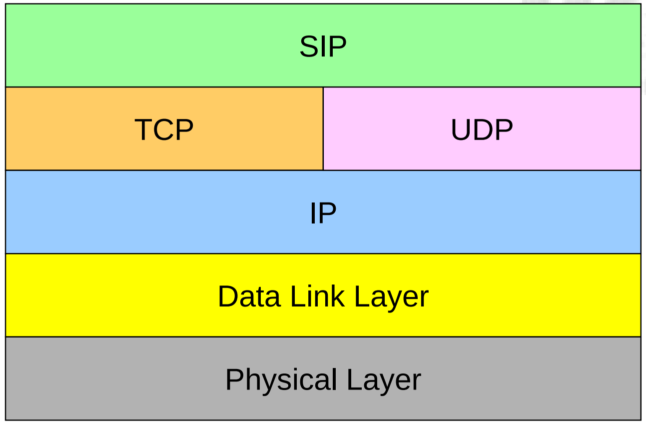
Overview

- Introduction and Motivation
- State of the Art
  - Speech Coding and Quality
  - **Signaling**
  - Media Transport
  - Security
- Research Directions
- Conclusions

Signaling

- IETF Session Initiation Protocol **SIP**
  - Session Description Protocol **SDP**
- ITU-T **H.323**

SIP Overview



Session Initiation Protocol

- IETF RFC 2543 of **MMUSIC** (Multiparty Multimedia Session Control) Working Group
- **Simple text-based** Protocol similar to HTTP and SMTP
- **Independent** from underlying Protocol (UDP, TCP …)
- Uses **URLs for Addressing** e.g.: user@domain, user@ip or phonnumber@gateway

SIP Scope

- **Setup, tear down** and **parameter change**
- of Calls between two or more Endpoints in an IP-based Network
- **Call**: Consists of all Participants of a Session which have been invited from the same Source

IETF - Conference Control Architecture

- SIP is part of IETF – Conference Control Architecture

Also there:

- **RTSP**: Realtime Streaming Protocol
- **RTP**: Realtime Transmission Protocol
- **RTCP**: Realtime Transmission Control Protocol
- **SDP**: Session Description Protocol

SIP Components

- User Agents (UA) ➔ Initiate or receive Calls
  - UA Clients (UAC): Initiates Call
  - UA Server (UAS): Receives Call
- Network Servers (NS) ➔ name resolution and user location
  - Proxy Server and Redirect Server

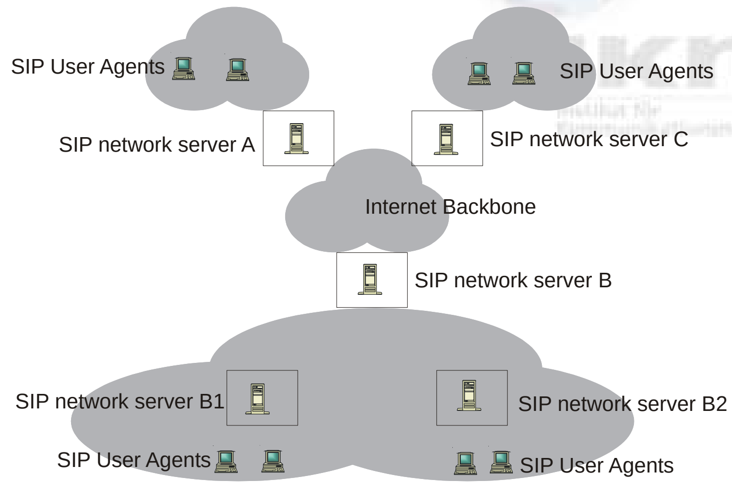
Handshake

- All Transactions consist of 3 way handshake
- Request ➔ Responses ➔ Acknowledge

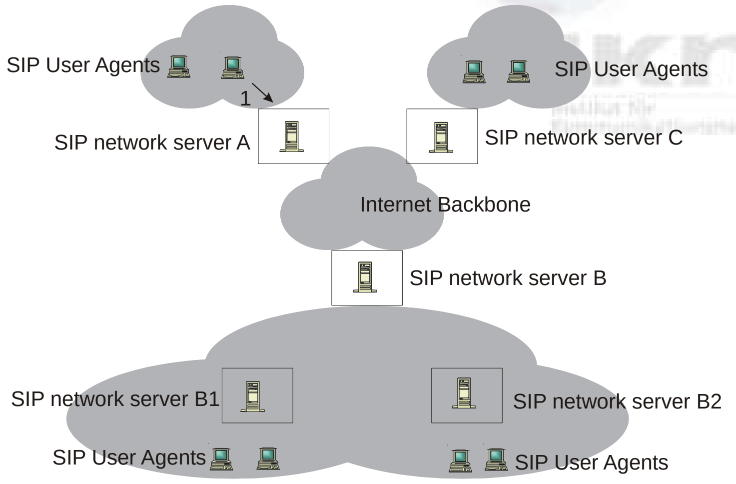
SIP Requests

- Seven Requests defined in SIP:
  - **Invite**: Invites User to Calls or changes Parameters
  - **Register**: UAS informs NS about Existence and Location
  - **Bye**: Terminate a Call
  - **Options**: Transports Information about Possibilities of Caller (no Call Setup)
  - **Ack**: Acknowledges Reception of Final Responses
  - **Cancel**: Ends a pending Call Setup
  - **Info**: for Mid Session Signaling

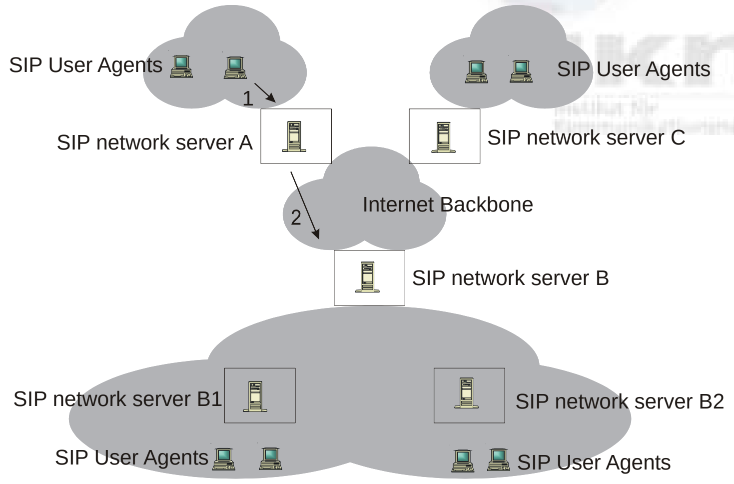
Sample: SIP Concept (0)



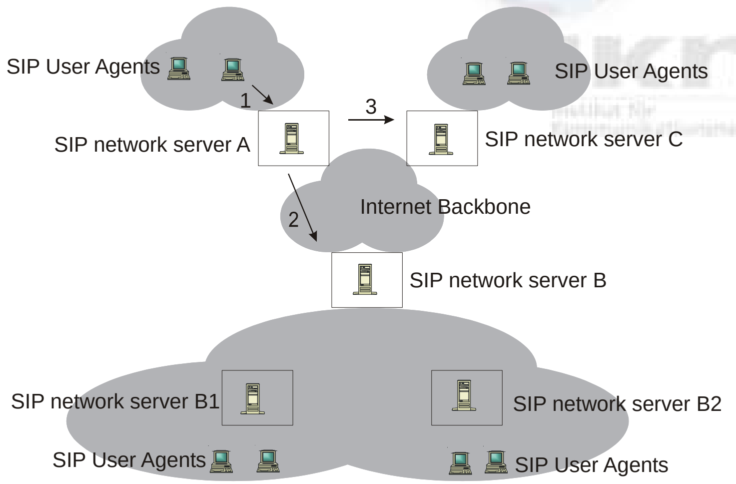
Sample: SIP Concept (1)



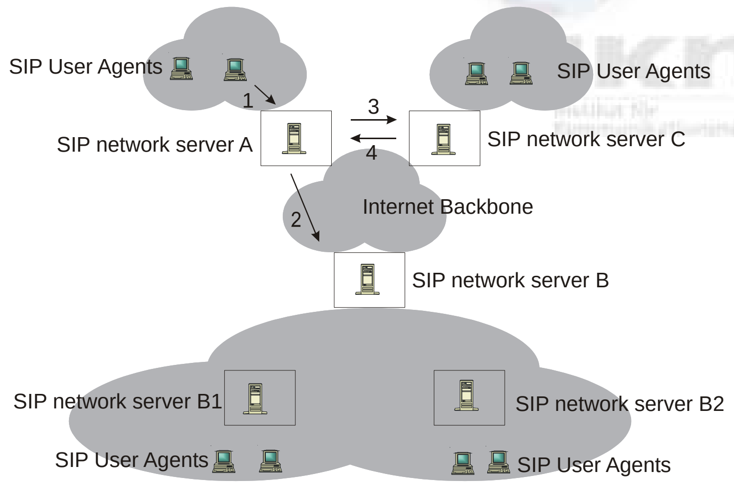
Sample: SIP Concept (2)



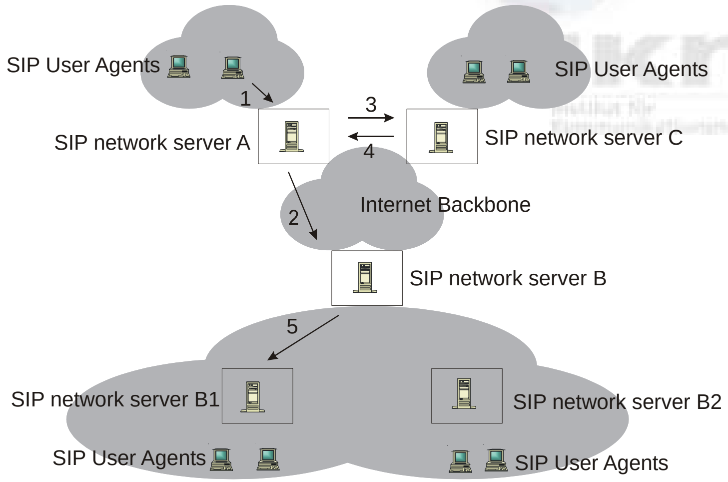
Sample: SIP Concept (3)



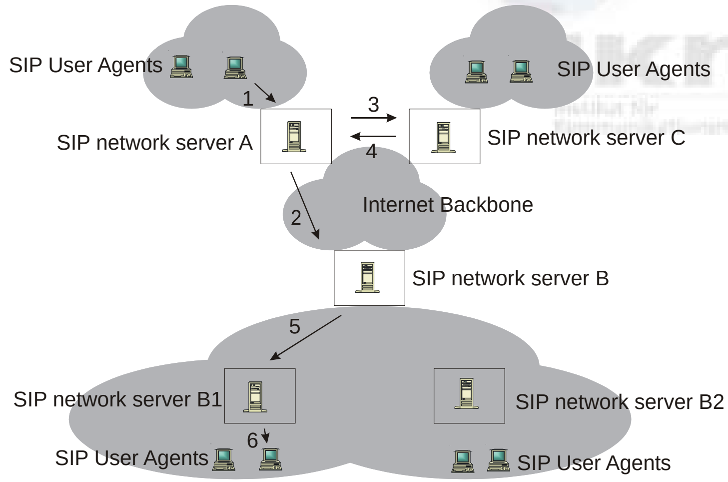
Sample: SIP Concept (4)



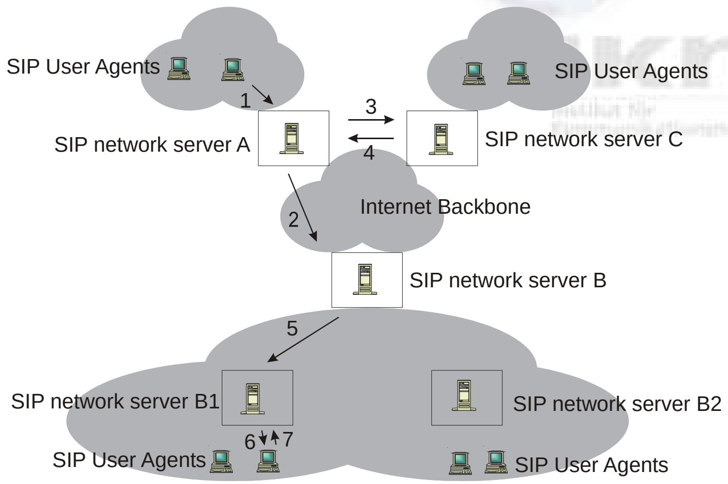
Sample: SIP Concept (5)



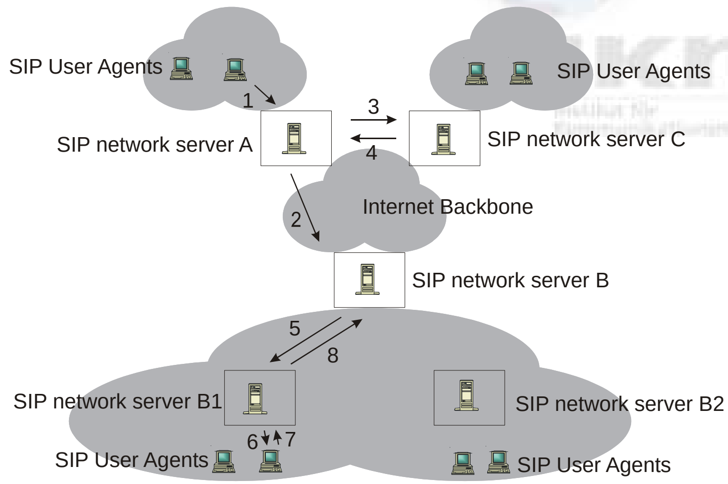
Sample: SIP Concept (6)



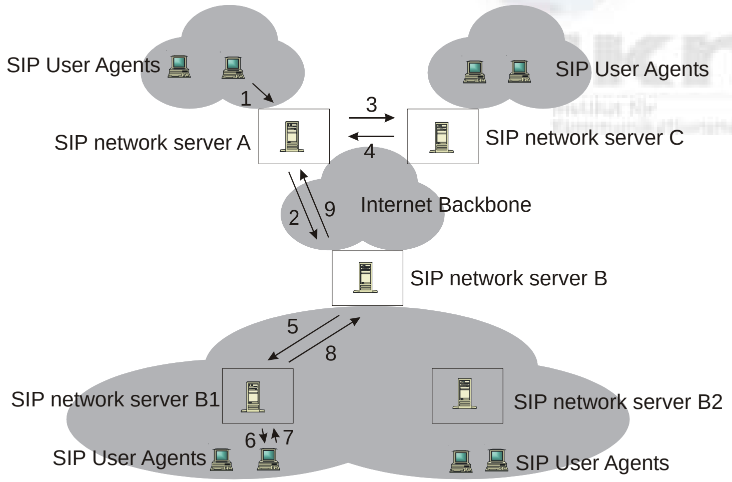
Sample: SIP Concept (7)



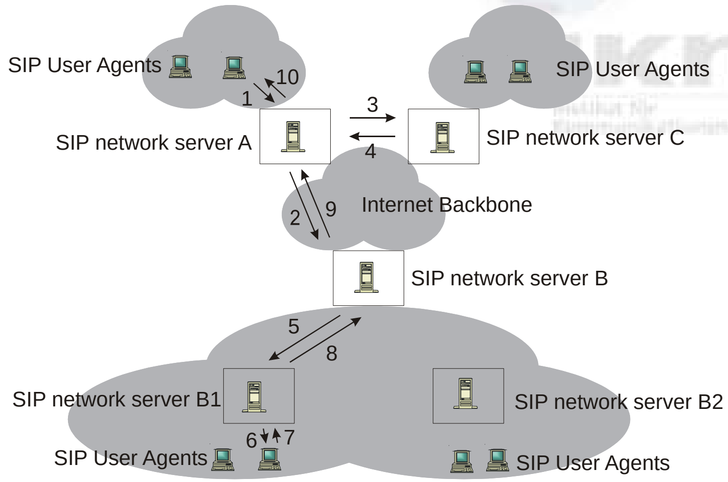
Sample: SIP Concept (8)



Sample: SIP Concept (9)



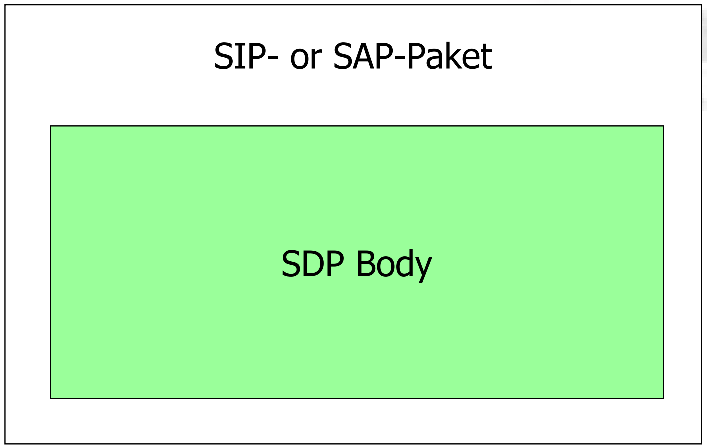
Sample: SIP Concept (10)



SDP – Session Description Protocol

- not a Signaling Protocol
- Description of Multimedia Sessions
- Excellent for Use with SIP
- SAP – Session Advertising Protocol for MBONE developed -> Need for SDP recognized
- Description of:
  - available MM Connections
  - Transport Protocol
  - IP Addresses and Port Numbers

Usage of SDP



Example for SDP Message

INVITE sip:Mittenecker@IKN.tuwien.ac.at SIP/2.0
..
Subject: New error codes

SIP Header

v=0
o=Mustermax 51633745 1348648134
IN IP4 128.3.4.5
s= New error codes
c=IN student.tuwien.ac.at
t=0 0
m=audio 3456 RTP/PCM 0 22
a=rtpmap:22 PCMU/8000

SDP Body

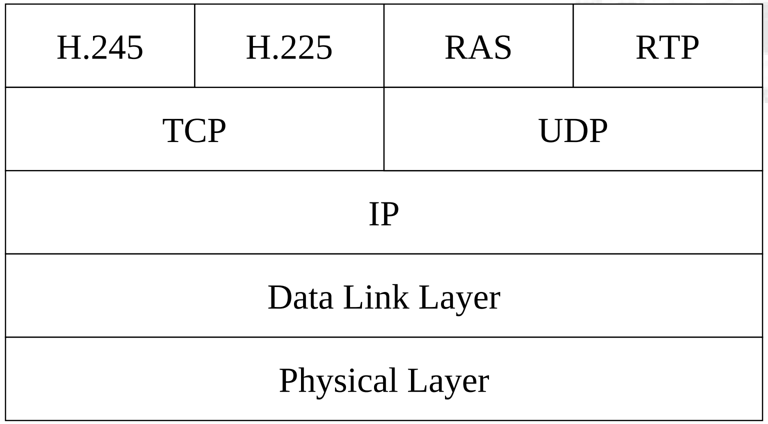
H.323

- ITU Protocol
- „Competitor“ to SIP
- 1996 first Version (3 years before SIP)
- Investments in H.323 initially slowed down SIP spreading
- Also Uni- and Multicast Connections supported
- Also different other Media Types supported
  - while developed for VoIP

Comparison H.323 - SIP

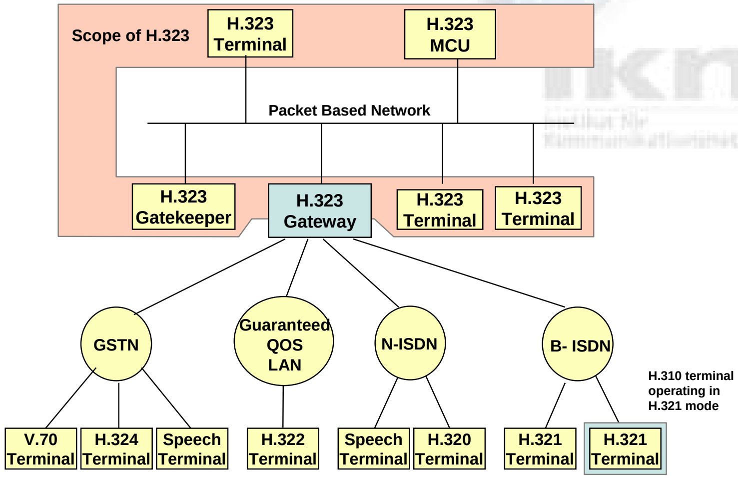
	H.323	SIP
Registration	RAS	SIP
Call Control	H.225	SIP
Negotiation	H.245	SDP

H.323 Protocol Stack

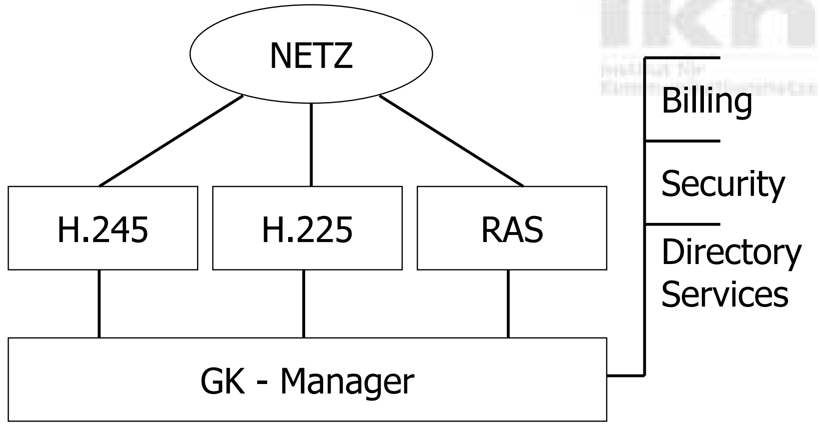


RAS .. Registration, Admission and Status

Scope of H.323



Structure of a Gatekeeper



Verbindungsaufbau

- SIP
  - Registration (SIP)
  - Call Control (SIP and SDP)
- H.323
  - Gatekeeper-Search (RAS)
  - Gatekeeper-Registration (RAS)
  - Admission Control
  - Call Signaling (H.225)
  - Negotiation (H.245)
- Direct Routed Call : Gatekeeper Routed Call

Gatekeeper Routed Call

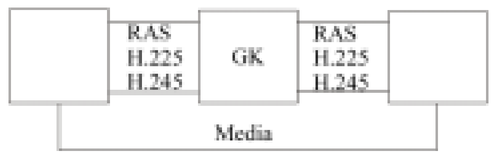
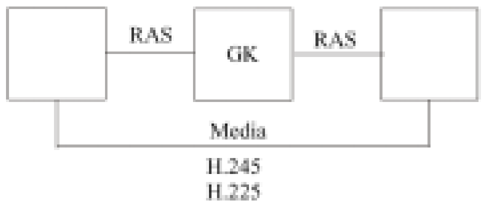


Bild 2.11: Gatekeeper Routed Call

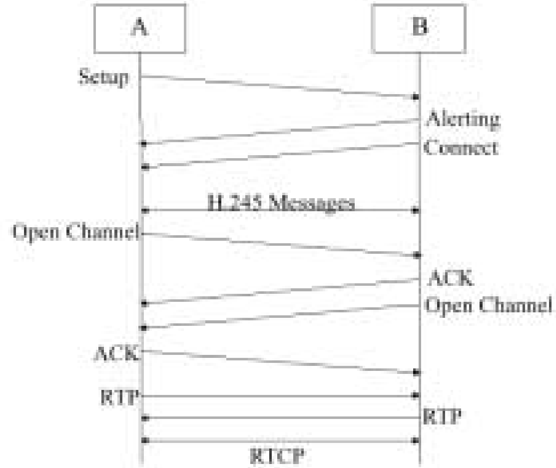
- Gatekeeper does complete Signaling
- Only Data channels are established by Terminals

Direct Routed Call



- Gatekeeper only for RAS (Registration, Admission and Status) responsible

Sequence Diagram for H.323 Call Setup



SIP – H.323

- SIP Advantages
  - Text based, few request types, simple protocol
  - Short documentation (~150 pages), easy to implement
  - Simple integration into IP networks
  - Services are controlled by the clients
  - 3GPP (3^{rd} generation partnership project, UMTS) voted for SIP
- H.323 Advantages
  - Longer available on the market
  - Integration into existing telephone system

H.323 vs. SIP: Origins

H.323	SIP-telephony
Origins in (video) telephony applications with compression capabilities. Conferencing is done with Multipoint Control Units (MCU's).	**Origins in the "Multi-party Multimedia Session Control" (MMUSIC) & SIP. This loosly coupled Conference concept works for 2-party conferences aka telephony, too.**
Designed for LAN's and currently extended for inter administrative domain telephony	**Designed for the flat Internet and assumed to provide open access to all IP users**
Defines codecs, terminals, Gatekeepers and gateways	**Defines SIP-servers (Gatekeeper like), SIP-clients (Terminals) no Gateway**
Call control in the gatekeeper	**Call control in the terminal**

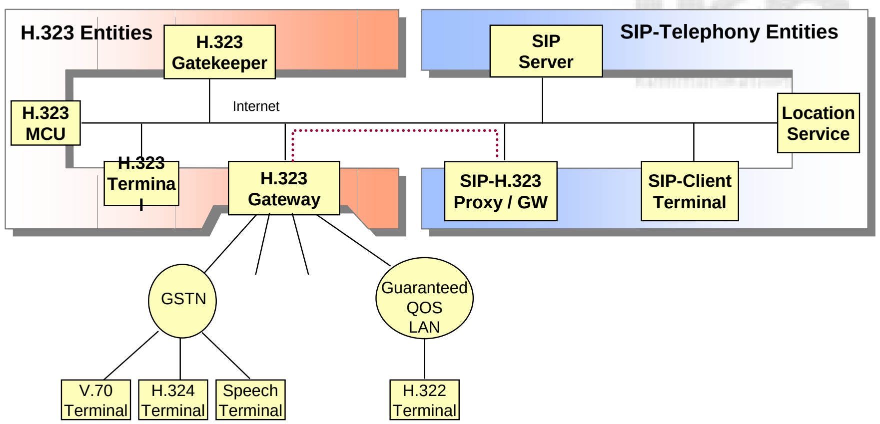
H.323 vs. SIP: Function Split

H.323	SIP-telephony
• Centralized call control ISDN (Q.931) style in the gatekeeper - PSTN interworking through gwys.	• Decentralized call control IP style in the user / terminal domain - No PSTN interworking
• User location via gatekeeper by use of IP (SIP) or PSTN / IN services	• User location via SIP services by use of directory services LDAP / DNS
• Inter domain user location is weak. - Gateway, user discovery (RAS)	• Inter domain user location is possible by existing services as above.
• Central control in Gatekeeper - No user call control possible	• Call Control Syntax - End teminal control script

H.323 vs. SIP: Protocols

H.323	SIP-telephony
• H.245: Capability exchange, None H.225 signals (DTMF...), Round trip delay,	• SIP: Session Initiation Protocol
• H.225: Call control (Q.931 Subset), RAS (Registration Admission and Status) , Media stream encoding	• SDP: Session Description Protocol (RFC 2327)
	• SAP: Session Announcement Protocol
	• RTSP: Real Time Streaming Protocol (RFC 2326)
• RTP Realtime Transport Protocol (RFC 1889)	• RTP Realtime Transport Protocol (RFC 1889)

H.323 vs. SIP: Joint environment



Media Transport

- Transport Protocols
  - Realtime Transport Protocol RTP
  - Realtime Transport Control Protocol RTCP
- other Transport Technologies
  - see QoS and MPLS Tutorials!

Media Transport Protocol Stack

RTP / RTCP
UDP
IP
Data Link Layer
Physical Layer

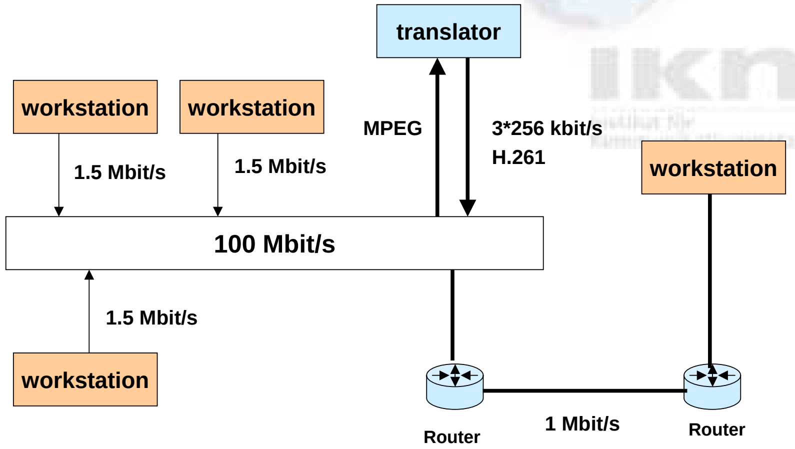
Realtime Transport Protocol RTP

- UDP does not support Synchronization
- RTP adds
  - Sequence Checking
  - Loss Detection / Retransmission
  - Intra- and Inter-Media Synchronization
  - Sender Identification
  - Multiplexing (Aggregation)

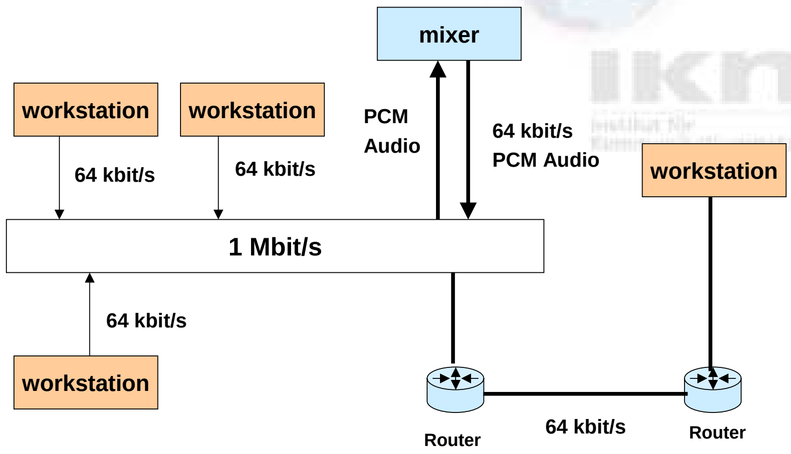
RTP Components

- RTP Mixer
  - Combine multiple Streams, Mixer acts as Sender
  - Decode Content and Codes in other Format
- RTP Translator
  - Translate between different Protocols
  - Solves the Firewall Problem

RTP Translator



RTP Mixer



Real Time Control Protocol RTCP

- Developed for cooperation with RTP
- Endpoints send periodic RTCP Packets during Sessions for Transmission Quality Feedback
- Five message types
  - Sender Report, Reception Report, Source Description, Explicit Leave, Extensions

Security

- „Same functionality as today's circuit-switched voice"
- Circuit-switched not encrypted, but wire-tapping for access necessary
- Public Internet or managed IP backbone
- Tunneling (e.g. L2TP- RFC 2661)
- Encryption: only for sensitive data

Research Directions

- H.323 – SIP Gateway
  - Testbed Implementation
- QoS Classification Model
  - Application Requirements → Flow Types → NW Classes
- Transport Mechanisms for Telephone Applications
- CODEC Performance
  - Simulation and Testbed

Alternative Best Effort ABE

- Best Effort in TCP/IP
- „Better QoS“ Strategies
  - IntServ –scalability issues
  - DiffServ – MPLS
  - ABE – Evolution
  - these are different technologies – not necessarily competing!
- TCP-Friendliness

ABE Ideas

- „Providing a Low-Delay Service within Best Effort“
- Simplicity of original Internet single class best effort service
- Additional Low-Delay Service for interactive, adaptive applications
- applications choose
  - lower end-to-end delay
  - more overall throughput

Application Marks Packets

- **green packet** = low bounded delay in every router, more likely to be dropped (or marked using congestion notification)
  - Interactive Audio, Video, .. (real time deadlines)
- **blue packet** = minimize overall transfer time
  - HTTP, binary data, „normal“ file transfers
- **flat pricing** may be maintained
- **no** need for **reservations or profiles**
- **new dimension** to best-effort services

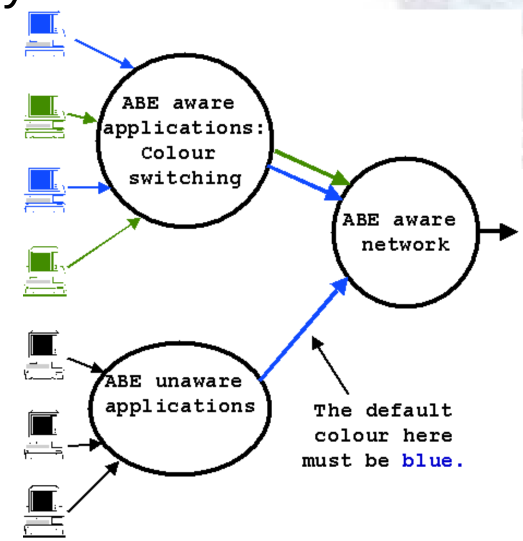
ABE Design Principles

- Support **rate-adaptive** multimedia applications in a best effort environment
- Explicit Congestion Notification (**ECN**) considered
- Rate adaption performed “**TCP-friendly**”
  - not more throughput than TCP flows
  - not so with e.g. UDP
- **Green** does not hurt **Blue**

ABE Router Requirements

- Give low bounded delay to **green**.
  - e.g. 5 – 20 ms (per Router)
- Conform to Local Transparency to **blue**.
  - not more delay and not more dropping for **blue** compared to flat best effort
- Conform to Throughput Transparency to **blue**.
  - **green** flows get lesser or equal throughput
- Minimize **green** losses as much as possible subject to the above requirements.

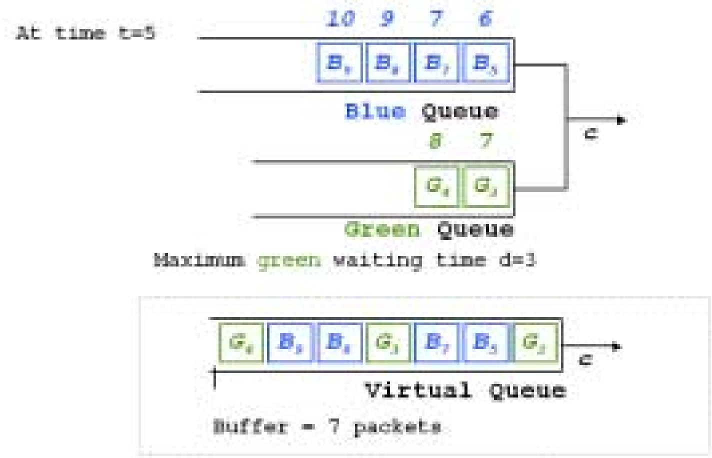
Deployment in Parts of NW



DSD Router Implementation

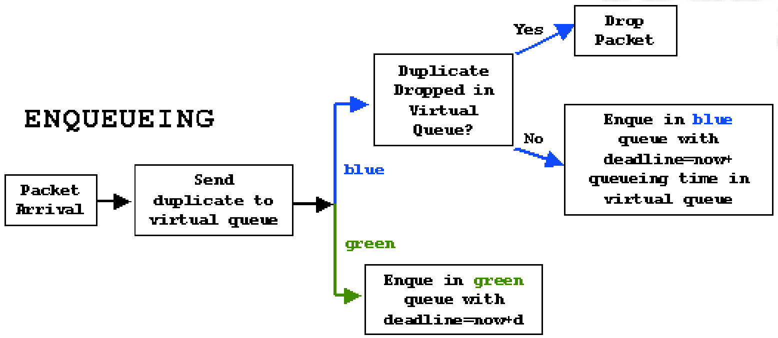
- Duplicate Scheduling with Deadlines DSD
- only Output Port Queuing used
- other implementations possible and do exist
- not only drop tail queues but also e.g. RED scheme would be possible

DSD Example

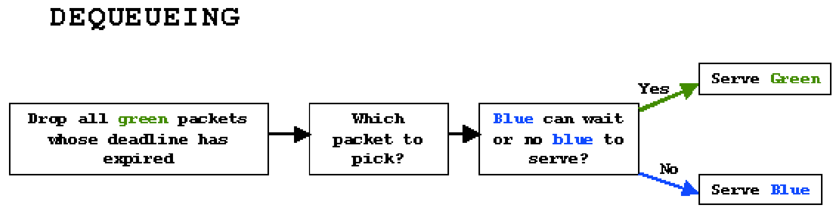


Fixed packet sizes, using packet time

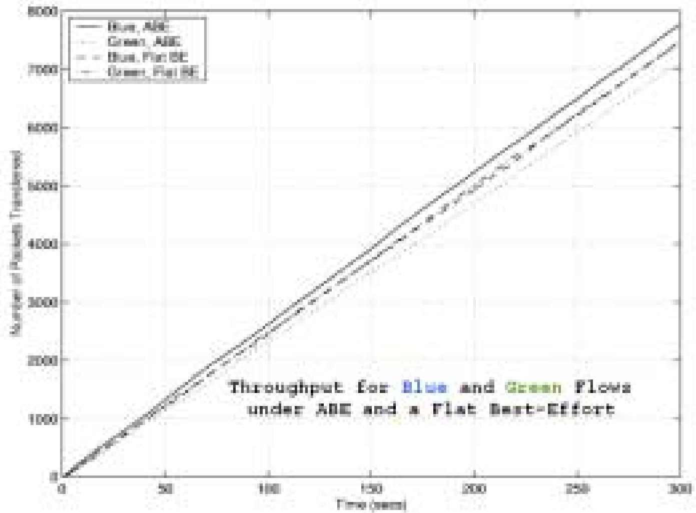
Enqueueing



Dequeueing

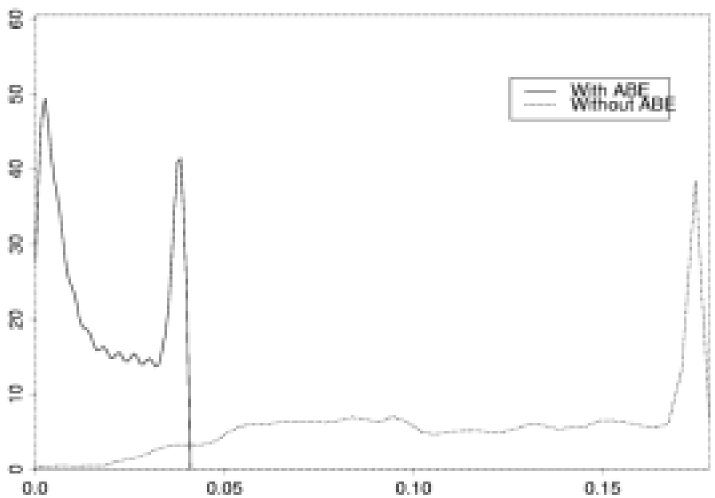


Simulation Throughput



Simulation Delays

Distribution of Queueing Delays of Green Flows



Who works on it?

- ICA, Swiss Federal Institute of Technology, Lausanne, Switzerland.
- Sprint ATL, California, USA
- Department of Computer Science, University of Leeds, UK.

Future Work

- Many open topics
  – e.g. Appropriate Application Marking
- Do own simulations in A0 with respect to VoIP and Multimedia Apps
- Compare with and simulate DiffServ Scenarios

Conclusion

- “Voice Telephone Service will be for free very soon.” (Prof. van As, 1996)
- Ad-sponsored “free” phone services available
  – e.g. www.ritstele.com
- Large User Bases for cheap calls
  – www.dialpad.com
  – www.phonefree.com
- Community Telephone Service
  – www.pulver.com/fwd/

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