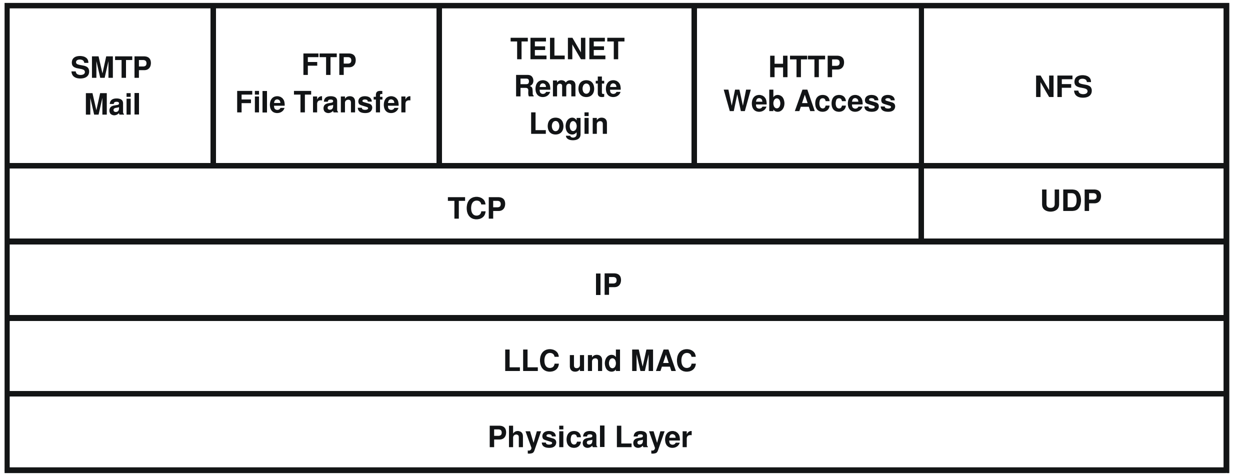
7. Application Layer

7.1 Internet Application Layer Architecture



SMTP	=	Simple Mail Transfer Protocol
FTP	=	File Transfer Protocol
TELNET	=	Remote Login Protocol
UDP	=	User Datagram Protocol
NFS	=	Network File System
TCP	=	Transmission Protocol
IP	=	Internet Protocol
LLC	=	Logical Link Control
MAC	=	Media Access Control

7.2 SMTP for Electronic Mail

SMTP: Simple Mail Transfer Protocol (RFC 822)

- Electronic mail in the Internet
- Uses a *direct* TCP connection to the destination host (no mail forwarding in layer 7!)
- TCP-Port: 25
- Examples for protocol data units:

HELO	Introduction
MAIL	Declaration of the sender
RCPT	Declaration of the receiver
DATA	Sending of the messages
QUIT	End
VRFY	Verifying of the user name
EXPN	Declaration of distributionlists

Functionality of SMTP

- The SMPT protocol handles the transmission of the message, but not the intermediate storage or presentation of the message to the user ("local matter").
- Email applications communicate by means of readable text (ASCII); SMTP PDUs do not contain binary data fields.
- The receiver confirms every message.
- Several messages (mails) can be sent over the same TCP connection if the receivers are on the same host.

Example Interaction with SMTP

```
R:     220 Beta.GOV Simple Mail Transfer Service
       Ready
S:     HELO Alpha.EDU
R:     250 Beta.GOV

S:     MAIL FROM:<Smith@Slphs.EDU>
R:     250 OK

S:     RCPT TO:<Green@Beta.GOV>
R:     550 No such user here

S:     RCPT TO:<Brown@Beta.GOV>
R:     250 OK

S:     DATA
R:     354 Start mail input; end with
       <CR>LF>.<CR><LF>
S:     ...sends body of mail message...
S:     ...continues for as many lines as message
       contains
S:     <CR><LF>.<CR><LF>
R:     250 OK

S:     QUIT
R:     221 Beta.GOV Service closing transmission
       channel
```

MIME

MIME (Multimedia Internet Mail Extension)

The early mail standards were designed for the transmission of text only, for human readers. In the early years, only 7-Bit US ASCII was allowed in the body. For the transmission of binary files (images, spreadsheet data, executable programs, etc.), ftp had to be used.

The MIME standard serves to convert arbitrary binary-coded data into an ASCII data stream that then pass all ASCII mail systems and mail gateways. It is now used very widely for all kinds of mail attachments. A MIME e-mail can have many "body parts“, each with a different *MIME type*.

7.3 FTP for File Transfer

ftp (file transfer protocol)

tftp (trivial file transfer protocol)

Functionality of FTP

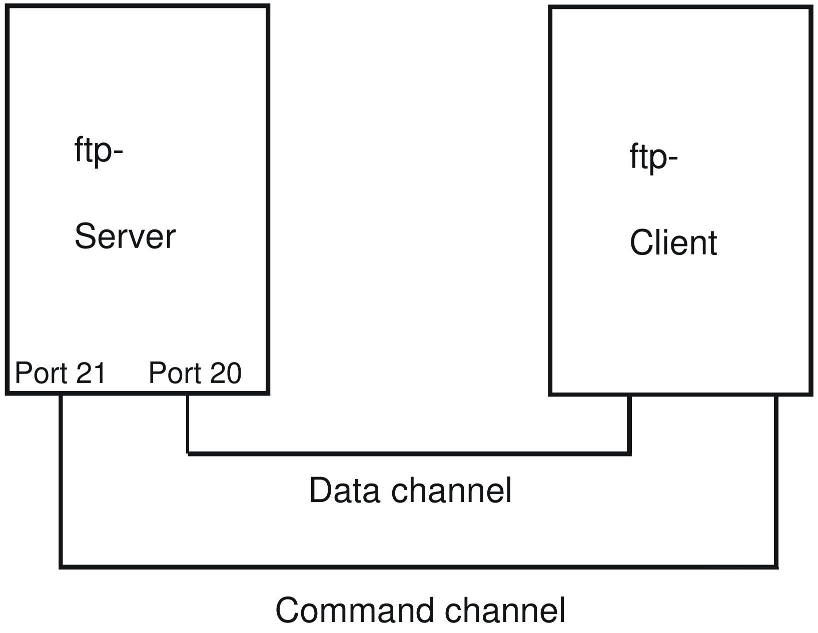
- Sending, receiving, deleting and renaming of files
- Creation and deletion of directories
- Change of the working directory
- etc.

Data transfer can take place in binary or ASCII mode.

In binary mode (also called "image file type“, type I) the bit stream is read from the sender’s memory and transferred unaltered.

In ASCII mode ftp assumes that only alphanumeric characters are to be transmitted. ASCII is chosen as a “transfer coding”: if the sender and the receiver have different local representations the text is transcoded. In transfer, “end of line” is codes as <CR><LF>; if the sender or the receiver has another local representation, this is also transcoded.

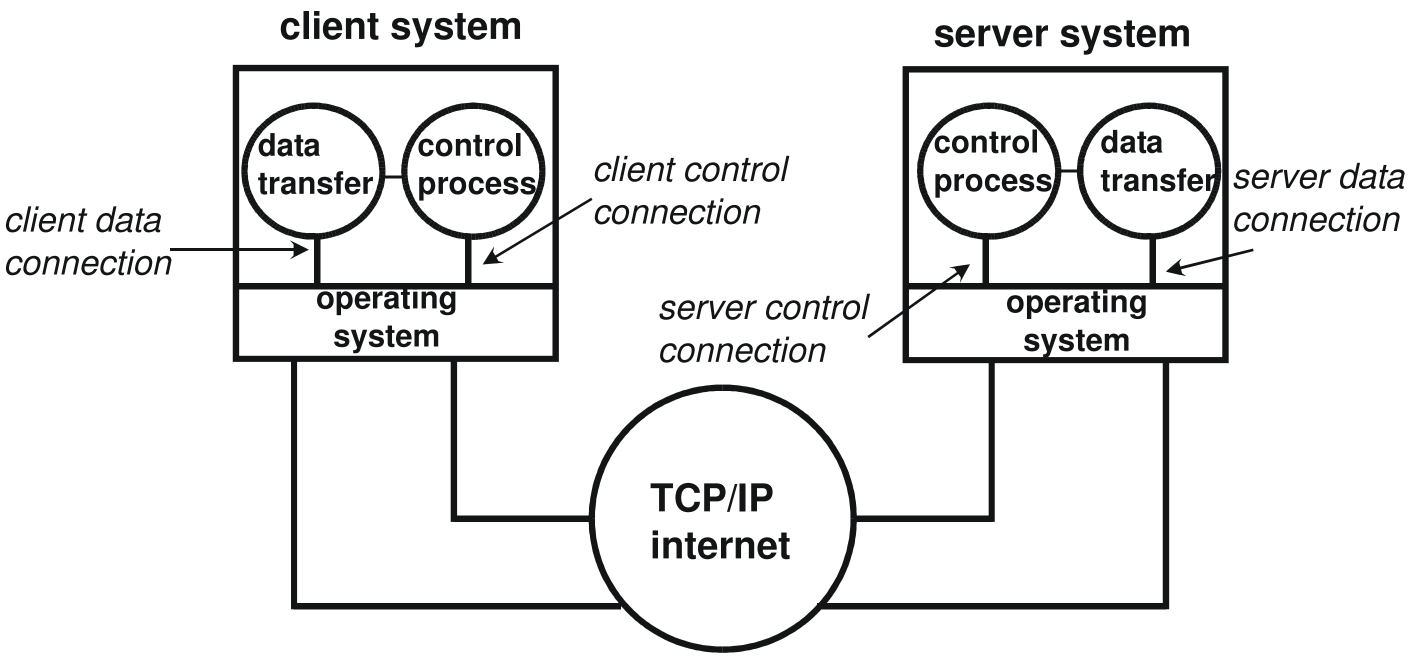
FTP Basics



- FTP commands are transferred as four-letter character sequences plus options (e.g., PASS xyz for the password).
- As an answer, a sequence of three numbers is delivered. The first number provides information on the type of the answer (1,2,3 => no error, 4,5 => error etc.).

FTP Architecture (1)

The ftp client runs as an application program in the address space of the user. There is no integration into the local file system.



FTP Architecture (2)

- Separate TCP connections for control protocol and data.
- Authentication (password exchange) when the control connection is established.
- Directory operations supported (ls, cd, rm, ...)
- “put” and “get” for data transfer
- help functions

7.4 NFS for Remote Access to Files

NFS = Network File System

History

1984	Announced by Sun Microsystems
1985	First product on a SUN available
1986	Porting for Unix System-V-release-2 completed
1986	NFS 3.0: (improved YP) and PC-NFS
1987	NFS 3.2: File Locking
1989	NFS 4.0: Encryption
1989	Licensed by 260 manufacturers

Characteristics of NFS

- Transparent access to files on remote file systems
- Integrated into the operating system/file system
- client/server model
- Developed since 1984 by SUN
- Standard on all UNIX computers
- Also available for MS Windows and mainframe operating systems
- Open system (public specification)
- Easy porting
- Public reference implementation
- Import/export of directories
- Communication over UDP (connectionless)
- No presentation layer services, only reading and writing of byte streams!

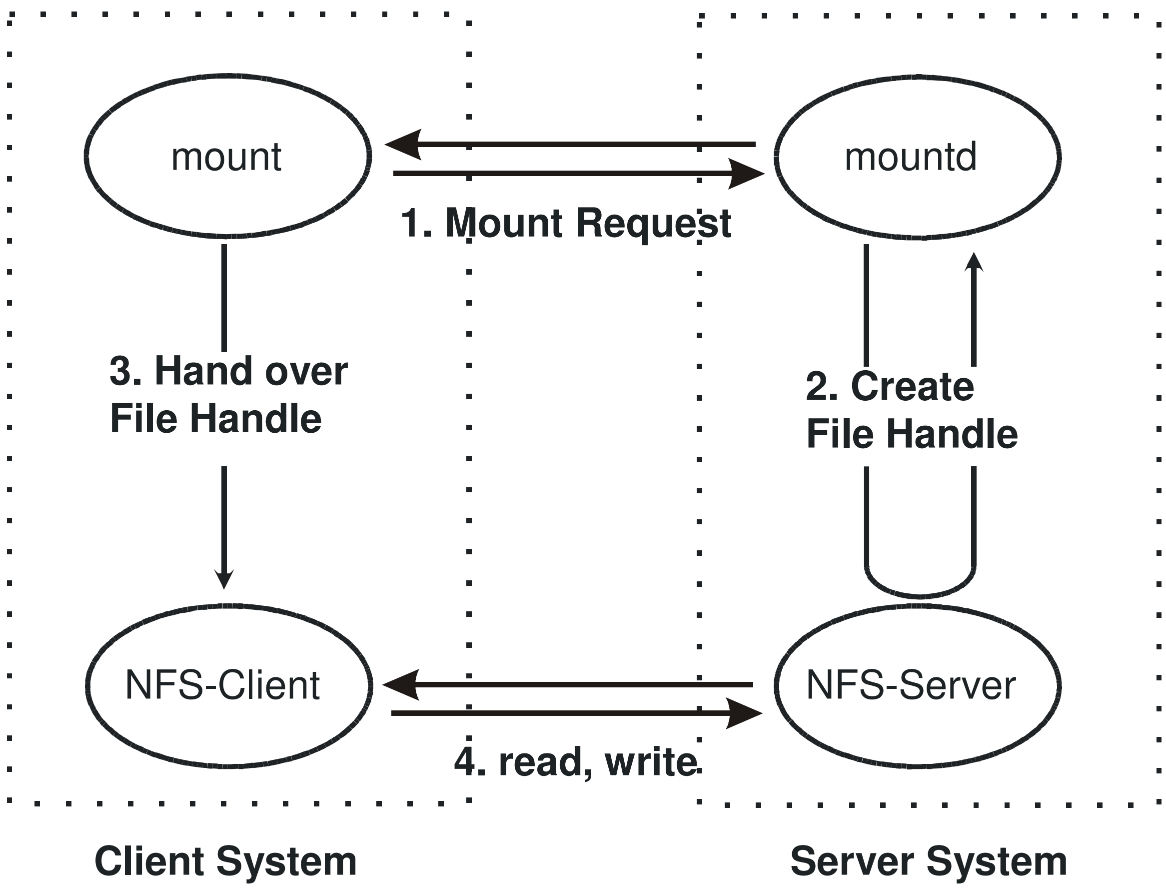
Mounting

The *NFS protocol* is only a file access protocol (reading and writing). The initialization of remote directories/files is performed by the *mount* protocol.

mount connects a remote file system with a local directory, i.e., the entire remote file tree is added into the local directory system. Afterwards, remote files can be accessed over NFS as if they were local files.

mount and *NFS* are separate protocols, mount and/or mountd only provide information (control blocks) for NFS and/or nfsd (e.g., computer names and directory paths).

NFS Protocol and MOUNT Protocol (1)



NFS Protocol and MOUNT Protocol (2)

Typically, *mountd* and *nfsd* ("demons" in the Unix sense) are started automatically during the boot up of the server. nfsd activates the NFS server code in the operating system.

During the MOUNT procedure a "file handle" (unique file system control block) is created on the server side and returned to the client. The NFS client uses this file handle in all later accesses to the remote (partial) directory tree.

Lock Manager

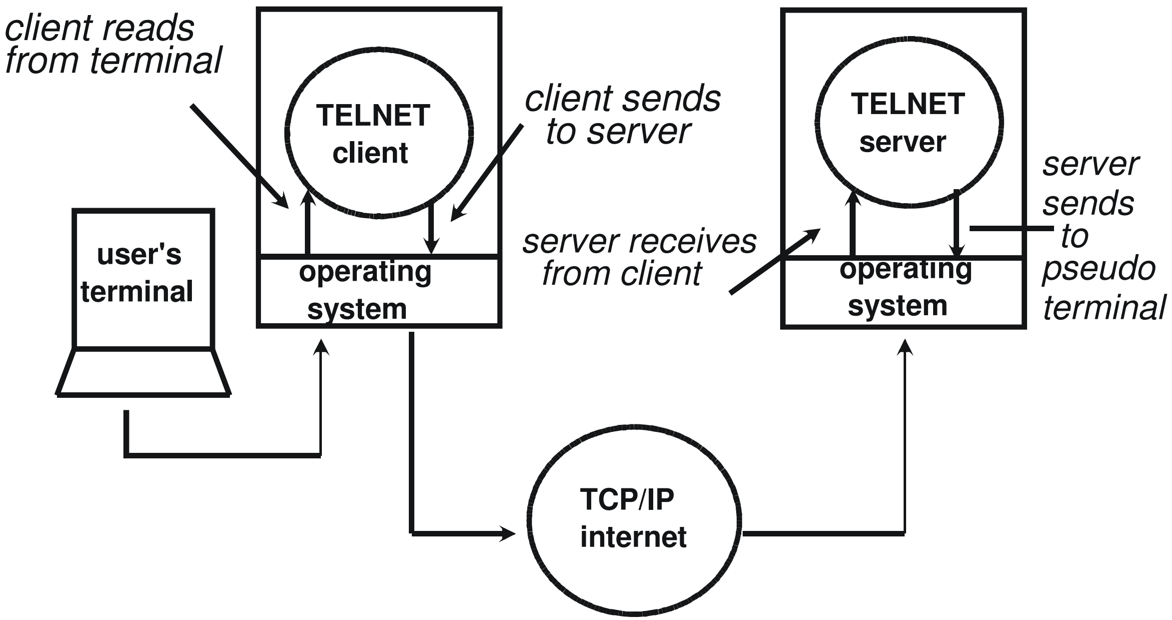
Problem

Simultaneous write access by several remote NFS clients can occur

- The lock manager provides locks for files.
- Service parallel to NFS (lockd)
- No deadlock detection!
- Not mandatory for all NFS clients!

7.5 TELNET for Virtual Terminal (Remote Login)

- Serves to provide a “virtual terminal” to a remote computer
- Alphanumerical, “full screen” with scrolling, flashing, etc. But does not support pixel graphics (like X.11 or other more modern protocols)
- Uses TCP



TCP connection between client and server

7.6 HTTP for the World Wide Web

HTTP - Hypertext Transfer Protocol

- Simple request/response protocol between a web client and a web server, all PDUs in ASCII format.
- Minimum transaction load of the server: **stateless protocol**
- In HTTP 1.0 a TCP connection is established and closed again for each individual document, the server process no longer occupies resources after the transmission of the document.
- Caching of DNS information (Client Caching).
- Transmission over a reliable transport protocol, typically TCP, other protocols possible.

HTTP - History

History

- 1989: HTTP 0.9 designed exclusively for Hyper**text**
- 1990: HTTP 1.0 Transmission of arbitrary data formats
- 7/1993: HTTP Internet Draft, first version
- 1996: HTTP 1.0 7th version
- 04/1999: HTTP 1.1 Internet Draft, stable

HTTP - Transaction

1. Connection establishment

- WWW client establishes a TCP/IP connection to the WWW server on
- TCP-Port: 80

2. Request

- Client sends requests over the established connection (for example GET, PUT, POST)

3. Response

- Reaction of the server to the request, for example dispatch of the requested document
- Code over status of the request

4. Connection takedown

- After conclusion of the transmission, termination of the connection by the server.

Example: ASCII Communication of HTTP

	bash$ telnet numalfix 80	
	Trying...	
	Connected to numalfix.wifo.uni-mannheim.de	*Blank line*
	Escape character is '^]'.	
Client:	**GET /index.html HTTP/1.0**	
Request	**Accept: image/gif**	*Method*
Server:	HTTP/1.0 200 Document follows	
Response	Date: Sun, 09 Jun 1996 13:13:09 GMT	*Document head*
	Server: NCSA/1.5	
	Content-type: text/html	
	Last-modified: Thu, 30 May 1996 10:42:31 GMT	
	Content-length: 1751	
	<html><head><title>BWL-Hauptseite</title><head>	
	<body>	
	...	
	<IMG SRC=/images/unilogo.gif...	
	<IMG SRC=/images/FakBWL2.gif...	
	<IMG SRC=/images/ball.red.gif...	
	...	

Extract from the Access-Log of the WWW Server

obelix.wifo.uni-mannheim.de - - [Date]	"GET/index.html HTTP/1.0" 200
obelix.wifo.uni-mannheim.de - - [Date]	"GET/images/neu2.gif HTTP/1.0" 200
obelix.wifo.uni-mannheim.de - - [Date]	"GET/images/FakBWL2.gif HTTP/1.0" 200
obelix.wifo.uni-mannheim.de - - [Date]	"GET/images/unilogo.gif HTTP/1.0" 200
obelix.wifo.uni-mannheim.de - - [Date]	"GET/images/ball.red.gif HTTP/1.0" 200
obelix.wifo.uni-mannheim.de - - [Date]	"GET/images/ball.green.gif HTTP/1.0" 200
obelix.wifo.uni-mannheim.de - - [Date]	"GET/images/minfo.gif HTTP/1.0" 200
obelix.wifo.uni-mannheim.de - - [Date]	"GET/images/minfo.gif HTTP/1.0" 200
obelix.wifo.uni-mannheim.de - - [Date]	"GET/images/mup.gif HTTP/1.0" 304

Improvements

- TCP connections persist, if several files are fetched from the same server (possible starting from HTTP 1,1).
- Establishment of several parallel TCP connections to the same server for acceleration of the data communication.

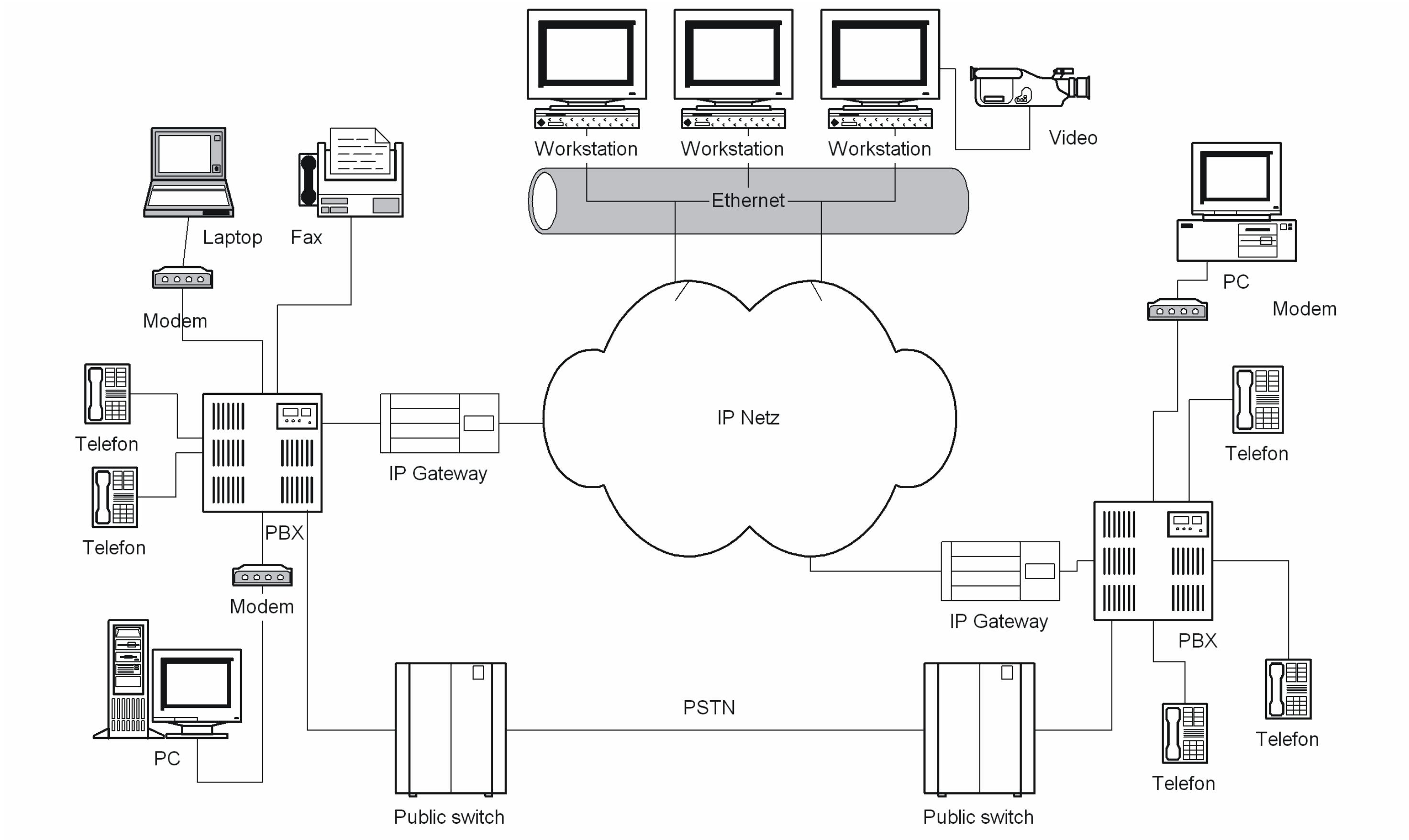
8.7 Telephone Services over IP

Basics and protocols

- Efficient realization, since packet switching over IP brings about fewer overhead and more flexibility.
- More functionality than the conventional telephony in the line-switched telephone network (PSTN)
- Goal: Multimedia communication and intelligent services in IP based networks.

*Ich danke Robert Denda und Dr. Andreas Grebe für die Überlassung von Folien für Kapitel 8.7

Architecture



Reasons for IP Telephony (1)

- Conventional telephone network: connection-oriented, line-switched, complex switching equipment (PBXs)
- IP-Telephony:
  - packet-switching, statistic multiplex effect, simple routers
  - low bandwidth due to audio compression (e.g. G.723.1 only 5,3-6,3 kbit/s compared to example 64 kbit/s PCM with ISDN)
- Flexibility with signaling
- Integration of multimedia data
- Upgradeable by intelligent network services (call forwarding, knocking, multipoint connections).
- Scalability of the communication service quality.
- Variety of terminal equipment: PC, IP telephone, Fax, etc.
- Use of existing data networks possible.

Reasons for IP Telephony (2)

Motivation for the user

- Cost advantages
- PC integration is often practical

Requirements for IP Telephony (1)

Main problem: Quality of Service

- **Delay**:

  Experimentally measured delays:

  - Coding/Decoding delay:
    approx. 30 ms (G.729A) - 82 ms (G.723.1)
  - Network delay (transmission, routing, etc.):
    Gateway-Gateway: 30 - 100 ms
    PC-PC: 50 - 140 ms
  - Access delay (operating system, sound and video cards, DSPs, ...):
    Gateway-Gateway: 40 - 80 ms
    PC-PC: 100 - 340 ms
    But: human ear very sensitive in this regard, desirable delay: < 100 ms!
- **Packet loss:** FEC necessarily, increases delay and data rate
- **Multicast**: Heterogeneity of the participants, dynamic Join and Leave require active, adaptive QoS mechanisms in the network

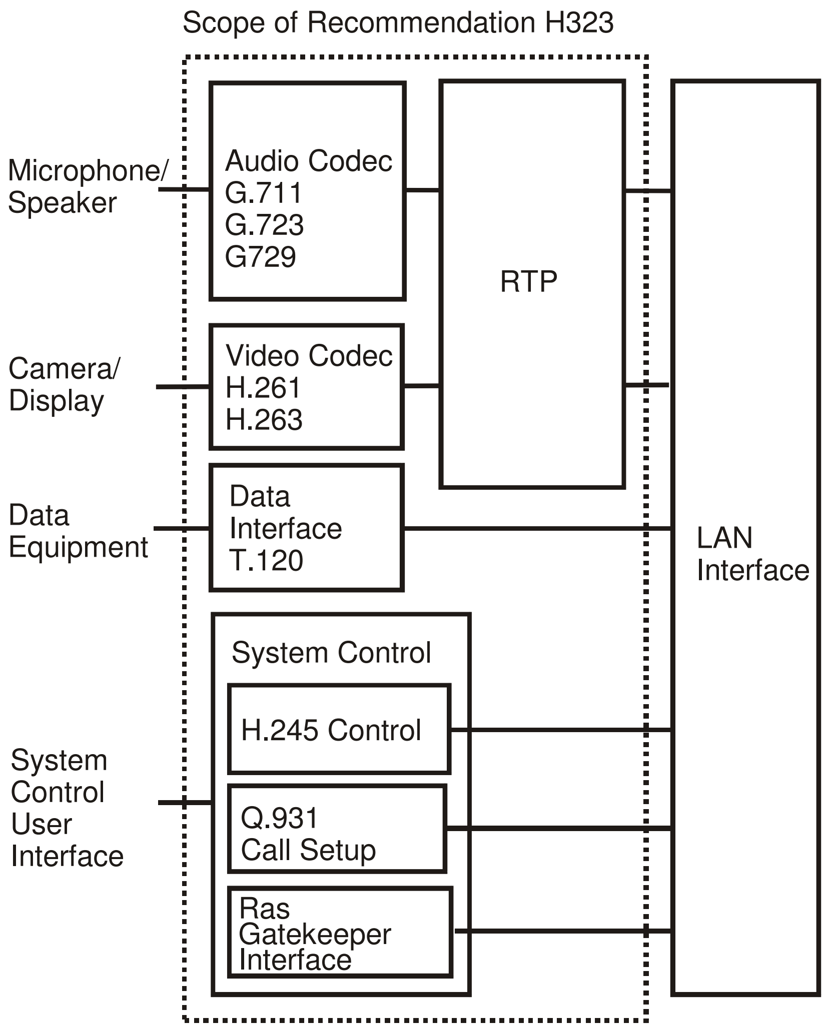
Requirements for IP Telephony (2)

- Intelligent services
  - "standard" of IN/AIN services: Knocking, answering machines in the network etc.
  - New intelligent services: Directory services, www interfaces etc.
- Signaling
  - lightweight signaling (internet vs. IN/AIN)
  - new media, extended services, charging requires new signaling mechanisms

Requirements for IP Telephony (3)

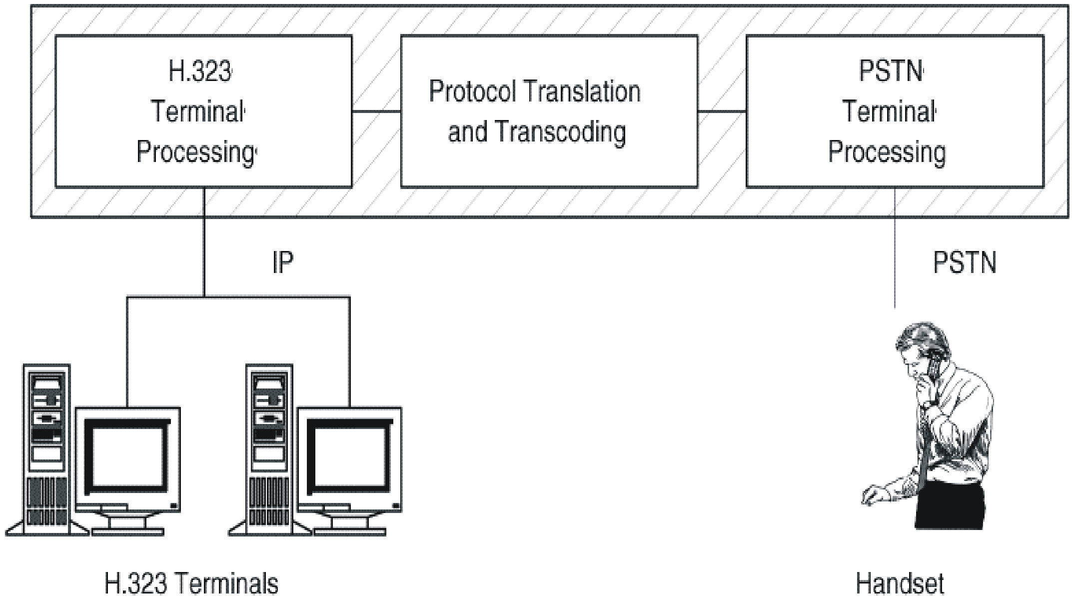
- Mobility, Interworking
  - Interworking of LAN, ATM, wide-banded entrance networks, Wireless LAN/WAN
- Security
- Accounting system (charging, accounting)

IP Telephony Standard: ITU Recommendation H.323



H.323 (1)

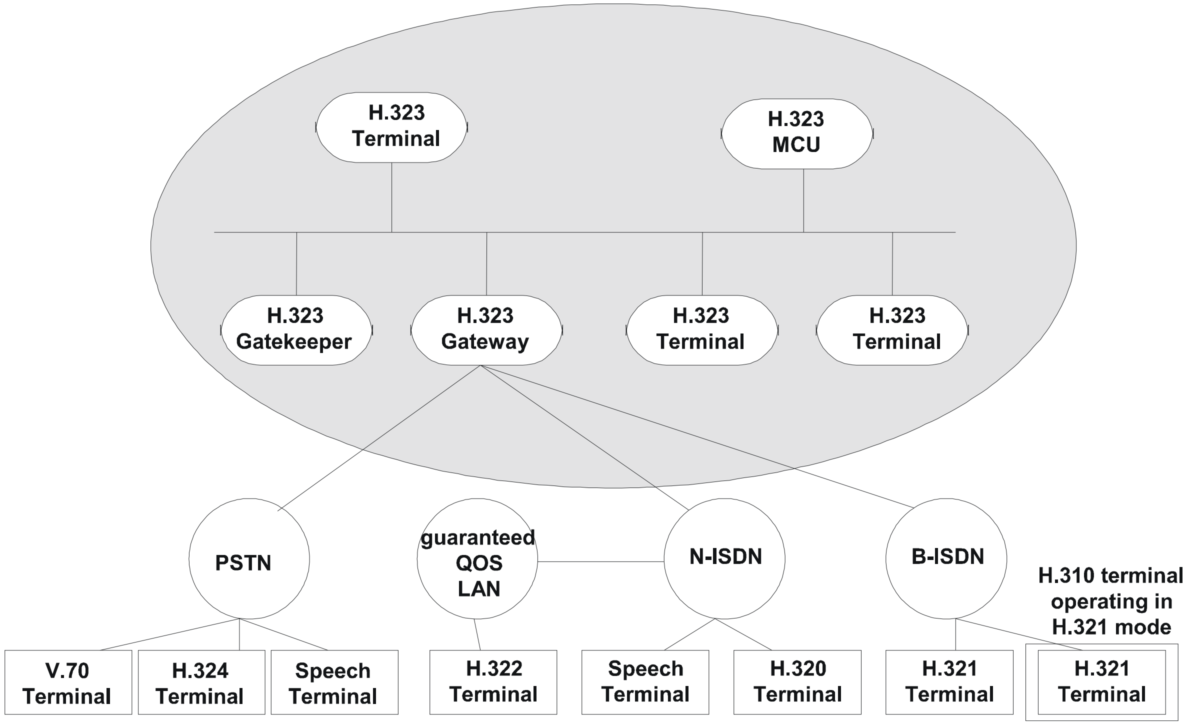
Version 1 of H.323 was adopted by the ITU-T in 1996, version 2 in January 1998. Version 3 is planned. H.323 is supported by most products for Voice over IP (VoIP) and includes both conventional telephone devices and PCs.



H.323 (2)

H.323 covers call control, multimedia and bandwidth management, and defines the interfaces between LANs and other networks.

Protocols for IP Telephony (1)



MCU= Multipoint Control Unit

N-ISDN= Narrowband ISDN

B-ISDN= Broadband ISDN

Protocols for IP Telephony (2)

Signaling Protocols

- Control Protocol for Multimedia Communication (H.245): End-to-End-Signaling, very complex
- Session Initiation Protocol (SIP): a simple text-based signaling protocol for internet conferences and telephony, supports among other things the transparent illustration of names in addresses and call diversion
- Digital Subscriber Signaling System No.1 (DSS1, Q.931), Q.93B, Q.932: ISDN signaling protocols
- RSVP – Resource ReSerVation Protocol: a receiver-oriented protocol for QoS and bandwidth reservation. However did not establish itself in the internet.

Protocols for IP Telephony (3)

Media stream control protocols

- Real Time Protocol (RTP) / Real Time Control Protocol (RTCP): internet protocol. Supports multimedia communication in real time. Profiles are defined for different payload types
- Real Time Streaming Protocol (RTSP): allows bi-directional transmission based on RTP, contain security mechanisms

"Intelligent Network" (IN, from telephony)

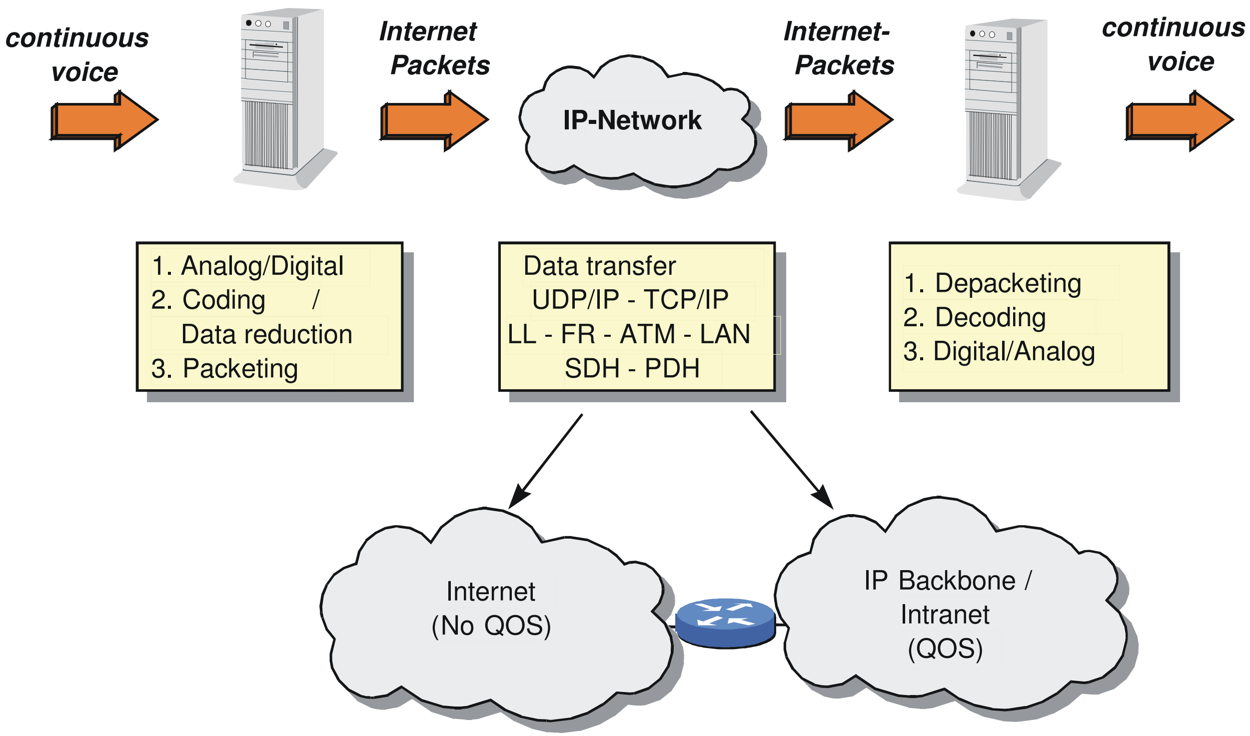
- PSTN: Intelligent Network (IN), Advanced Intelligent Network (AIN)
- IP-Telephony: Telecommunication Information Networking Architecture (TINA)
- Field of current research: Use of modern distributed systems (CORBA, Java RMI, DCOM), active networks and mobile agents for the implementation of more network intelligence

Voice over IP (VoIP)

Voice services	Fax	Business	value-added network services
PC-to-PC (fern/international)	Single fax	VPN-Service integration	Electronic Commerce (Hotline/Call center)
PC-to- Phone (long-distance/ international)	Fax-Broadcast	Teleworking / Video telephone	Entertainment (Chat/Games)
Phone-to- Phone (long-distance/ international)			

- Today, VoIP services are value-added application services for Internet service providers
- In the future: a standard service for all Internet users?

Principle of Voice over IP



- VoIP today: usable, some commercial services, based on standard IP ("best effort")
- VoIP tomorrow: good quality (QoS), international services/alliances?

Voice Coding (1)

There are voice coders for IP telephony with good quality with very low bit rate (e.g. GSM 06.10 with 13,2 kbit/s or G.723.1 with 5.3 kbit/s - 6.3 kbit/s).

Used procedures are usually based on **Linear Predictive Coding (LPC):** For each frame of voice samples s[i], p weights

lpc[0], .. , lpc[p-1]

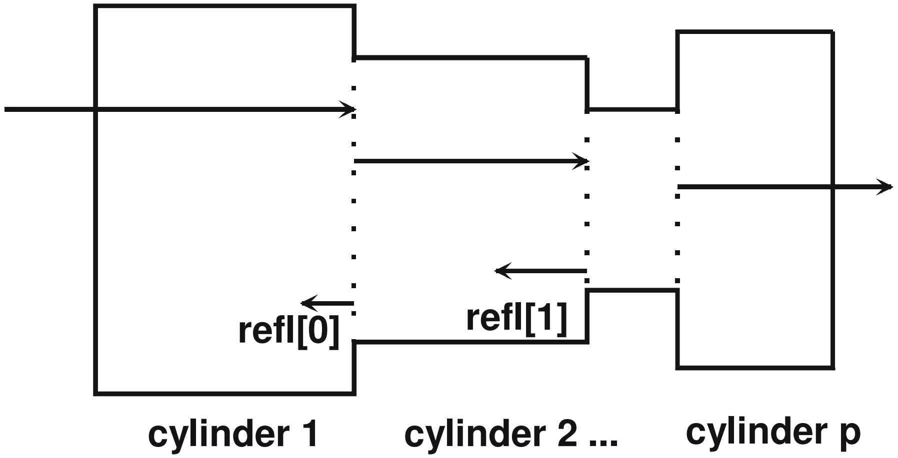
are computed, so that applies: s[i] is as well as possible approximated through

lpc[0] * s[i-1] + lpc[1] * s[i-2] + ... + lpc[p-1] * s[i-p]

Common values for p are 8 or 14.

Voice Coding (2)

With LPC the human voice organ is modeled as a system of connected, differently large cylinders:



Coding Model

Acoustic waves run through a system of cylinders. At transitions of cylinders with different diameters, they are partly reflected and thus interfere with the following waves. The reflection rate is represented by the reflection coefficients

refl[0], ..., refl[p-1]

These correspond almost to the lpc coefficients.

Coding:

For every frame:

- Computing of the lpc/refl coefficients.
- A synthetically generated signal serves as input of the model and results in synthetic voice.
- Differences ε_i between synthetic voice and samples are coded (ε_i small with voicing phonemes); the lpc/refl coefficients are coded and transferred.

Example: G.723

G.723.1

Adaptive CELP coder (CELP = Code Excited Linear Predictor)

CELP: The ε_i are coded as indices in a codebook.

ACELP: like CELP, but the codebook is adaptive

GSM 06.10: Regular Pulse Excitation – Long Term Prediction (RPE-LTP)

Comparison of Different VoIP Codecs

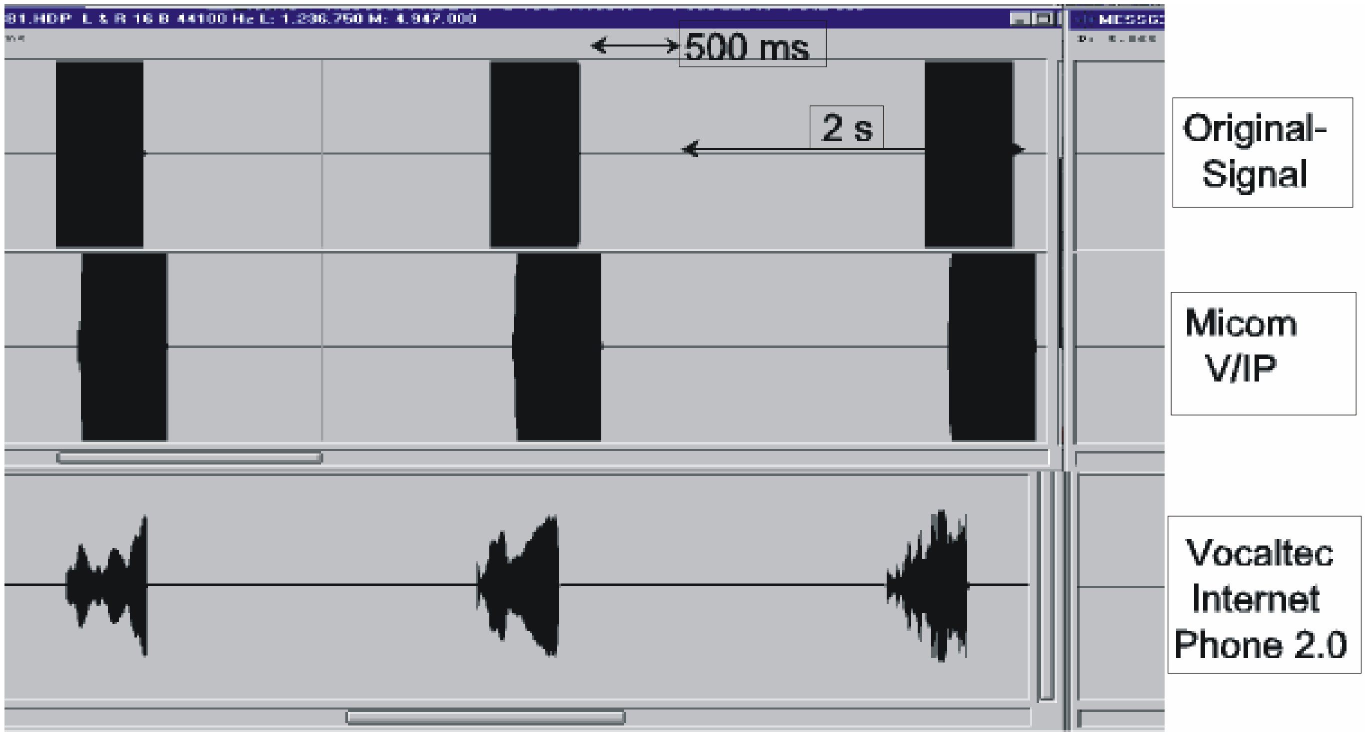
Codec	Method	Bit rate	Quality (MOS)	Standard	Use	Coding Delay	Power con-sumption (100 MHz Pentium)
G.711	PCM	64 kbit/s	4,0	H.323	ISDN	< 1 ms	< 1%
G.723.1	ACELP MP-MLQ	5,3 kbit/s 6,3 kbit/s	3,88 3,88	H.324/H.323	PSTN Video telephone Voice over IP	97,5 ms 97,5 ms	35-49% 35-49%
G.728	LD-CELP	16 kbit/s	3,93	H.323	Voice over IP	3 ms	approx. 65%
G.729	CS-ACEL	8 kbit/s	3,90	H.323	Voice over IP Frame Relay ATM	35 ms	approx. 50%
GSM 6.10	RELP	13 kbit/s	3,80 at 0% errors	not included in H.323	Mobil	ca. 40 ms	real-time coding at 486PC 66MHz
Lucent SX7300P	CS-ACEL	7,3 kbit/s	3,88	not included in H.323	Voice over IP	35 ms	approx. 13,5%

Mean Opinion Score (MOS): Questioning of test persons

>3.8 acceptable

>4.0 very good quality

Distortion of VoIP Codecs



- Codecs show strongly varying distortion characteristics
- Psycho acoustic codecs show clearly stronger distortion
- Mean Opinion Score (MOS) in both cases acceptable