Part 1: Basics and Principles

- Telecommunications
  - Networks, Applications, Examples
- Multimedia and Communication Architectures
  - Systems, Protocols, Characteristics
- High Speed Networking Principles
  - Characteristics, Services, Architectures, Switching

Telecommunications

Networks, Applications, Examples

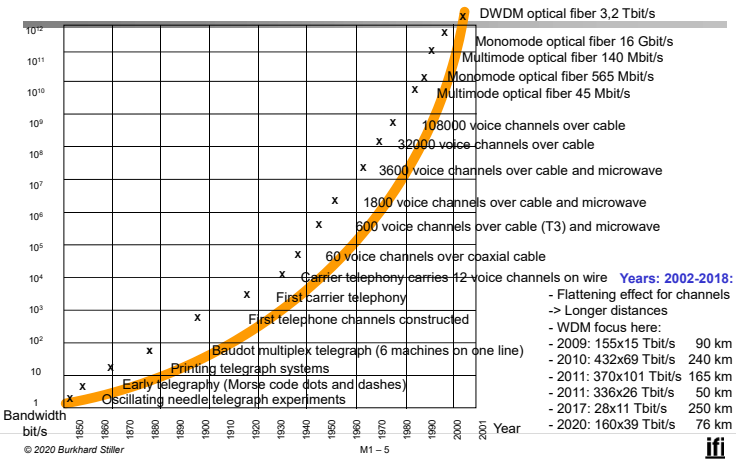
Telecommunication

- Communication: The method to call, contact, shout, yell, or alarm somebody.
  - Traditionally, using voice, letters, or drawings expressing abstract ideas in a negotiated language (natural language).
  - Face-to-face communication or some postal provider offer a communication service (still human-to-human).
- Telecommunication: The method to call, contact, phone, or ring somebody or some machine, which is supported technologically.
  - Tele: (Greek) distant or remote.
  - Combination of telecommunication and computers is powerful to combine local processing power with distributed usage at remote places (already human-to-machine).

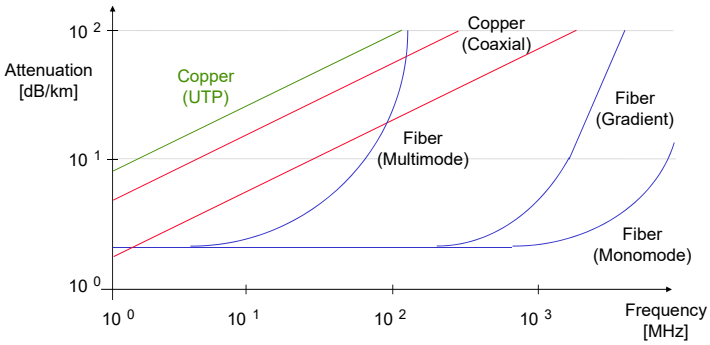
Information, Data, and Message

- Information is defined as the knowledge of subjects, topics, activities, opinions, issues, devices
  - Informal definitions, a collection of contents.
- Data are characters or continuous functions which represent information to be processed in a well-defined manner (DIN 44300) – Verarbeitung.
  - Digital data are coded in discrete characters.
  - Analog data are represented by continuous functions.
- Messages are information in digital representation (data) to be used for transmission – Übermittlung.
  - The structure of messages and its exchange rules are defined in communication protocols.

Inventions in Telecommunications



Transmission Media



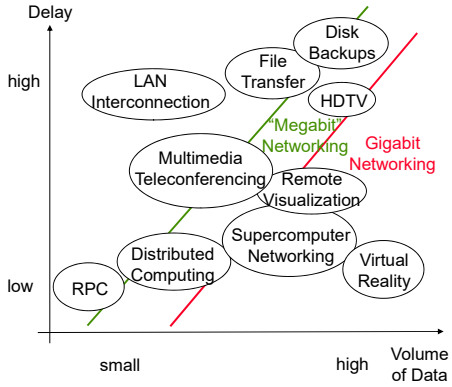
- Fiber optical media:
  - Low error rates, long distances, low attenuation.

Applications – Traditional and Advanced

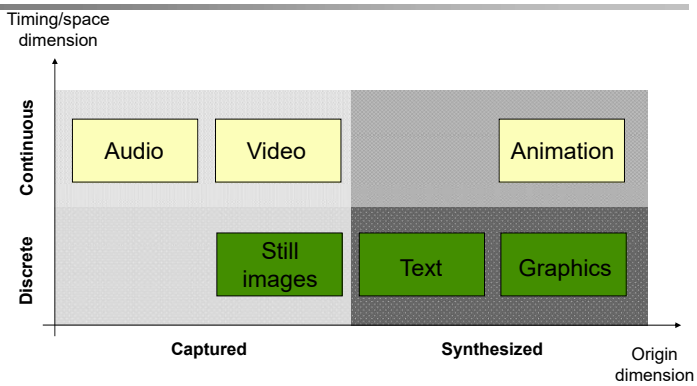
- Well-known and regularly used applications:
  - Analog data communicating applications:
    - Telegraphy since 1850s
    - Phone (invention 1879, establishment in Germany 1882)
    - Television since 1950s
  - Digital data communicating applications (since early 1960s):
    - File Transfer Protocol (FTP), Terminal emulation (telnet), …
    - FAX (early 1980s)
- "Demanding" applications:
  - Digitized analog applications: video/audio broadcasting, streaming, picture phone, HDTV, conferencing
  - Digital applications per se: network management, secure messaging, virtual reality, and local gaming
- Distributed applications:
  - Collaborative desktop (CSCW)
  - CAx (x = Design, Manufacturing, Training, or Education)

High Performance Applications

- "Standard":
  - Small volumes of data and
  - High delays.
- "Emerging":
  - Medium volumes of data and
  - Medium delays.
- "Gigabit":
  - High volumes of data and
  - Low delays.



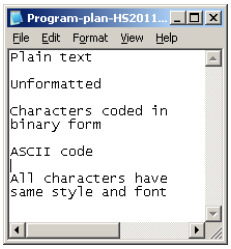
Conventional Classification of Data

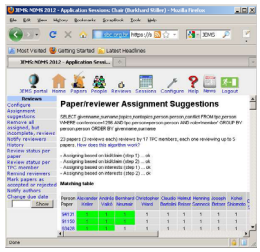


Temporal vs. spatial dimensions: 2D images and 3D holography.

Text Variants

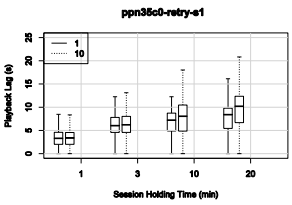
- Plain text
  - Unformatted
  - Characters coded in binary form
  - ASCII code
  - All characters have same style and font
- Rich text
  - Formatted
  - Contains various format information besides pure codes of characters
  - No predominant standards
  - Characters of various size, shape and style, *e.g.*, **bold**, *colorful*

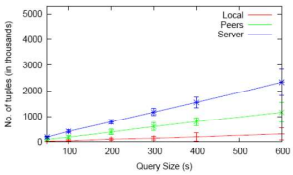




Graphics

- Revisable document that retains structural information
- Consists of objects such as lines, curves, and circles
- Usually generated by graphical editor of computer programs





Still Images

- Non-revisable document
- Document format is unaware of any structural information
- Semantic content is not preserved
- Described as bitmaps formed of individual pixels
  - Monochrome image – pixel depth 1 bit
  - Gray-scale – pixel depth 8 bit
  - Color image
    - High color – pixel depth 15/16 bit
    - True color – pixel depth 24 bit
    - Deep color – pixel depth 30/36/48 bit

- 1 bit = 2 colors, 8 bit = 256 colors, 24 bit = 16,277,216 colors

Video and Animation

- Both type of images and graphics can be displayed as a successive view, which creates an impression of a movement

- Video: Moving images or moving pictures
  - Captured or synthesized
  - Consists of a series of bitmap images
  - Each image is called a frame
  - Frame rate: the speed to playback the video (frame per second)

- Animation: Moving graphics
  - Typically generated by computer program (animation authoring tools)
  - Consists of a set of objects
  - Movements of objects are calculated and the view is updated at playback time
  - No "frame rate" defined

TV Format Examples

- Throughput:
  - Video in digital formats:
    - NTSC (National Television Systems Committee)
    - PAL (Phase Alternating Line)
    - HDTV (High Definition Television)
    - UHDTV (Ultra High Definition Television) or "4k"/"8k"

TV: Television
SD: Standard Definition

https://en.wikipedia.org/wiki/Ultra-high-definition_television

  - Scan/row x visible rows x bit/scan x frames/sec = data rate:

Name	Bandwidth Requirements	Frame Rate	Resolution
NTSC	≅ 90 Mbit/s	(30 frames/s)	720 pixels
NTSC with compression	≅ 3 … 6 Mbit/s		
PAL	≅ 95 Mbit/s	(25 frames/s)	720 pixels
HDTV	≅ 1,152 Gbit/s	(60 frames/s)	1080 pixels
HDTV with compression	≅ 25 … 34 Mbit/s		
UHDTV with compression	≅ 20-40 Mbit/s	(50...120 frames/se)	3840 pixels
UHDTV-2 uncompressed	≅ 144 Gbit/s	(120 frames/s == Hz)	7680 pixels

Video File Formats

- AVI (Audio Video Interleave) and WMV (Windows Media Video)
  - Long running digital video workhorse (Microsoft), smaller compression, limited
- MOV or QT (QuickTime)
  - Supporting a wide range of codecs (Apple), cross-platform (PC, Apple)
- AVCHD (Advanced Video Coding, High Definition)
  - Data compressed with H.264 codec (Panasonic and Sony)
- MPEG (Motion Pictures Expert Group)
  - Integrates MPEG-1 and MPEG-2 video and audio compression
- MP4 (MPEG-4 Video File)
  - Video and audio tracks are compressed separately (Web)
- H.264 (ITU-T Recommendation – Standard) 16:9 resolutions in comparison [Wikipedia]
  - Standard for high definition digital video (MOV, AVC, AVCHD, MPEG, Divx)
- H.265 or High Efficiency Video Coding (HEVC)
  - Stream full 1080p HD video (vertical resolution)
  - Streaming 4k (horizontal resolution in pixels) ≈ 20 Mbit/s

Audio

- One-dimensional time-based signal

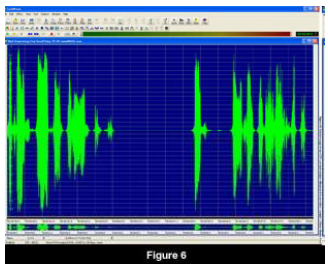


Figure 6

- Speech vs. non-speech sound
  - Speech supports spoken language and has a semantic content
  - Non-speech does not convey semantics in general
- Natural vs. structured sound
  - Natural sound is recorded
  - Generated sound is artificial
  - All represented as wave in digital signal
    - Example: Audio in CD, WAV files
  - Structured sound conforms to a synthesized sound in a symbolic way
    - Example: MIDI file

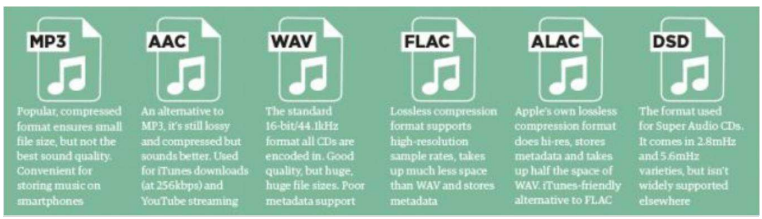
Audio Examples

- Throughput:
  - Audio in digital formats:
    - Telephone quality (Pulse Code Modulation, PCM):
      - 8,000 sample/s x 8 bit/sample = 64 kbit/s.
    - Compact Disk (CD) stereo quality:
      - 44,100 sample/s x 16 bit/sample x 2 = 1.4112 Mbit/s.
  - Compression techniques:
    - Telephone standard G.711 with 32 kbit/s:
      - More advanced techniques compress up to 4 kbit/s.
    - CD MUSICAM standard (MPEG) with 192 kbit/s:
      - Studio quality needs 384 kbit/s.
- Delay:
  - One-way (unidirectional) versus interactive (bidirectional).

Audio File Formats

- High-resolution audio
  - No single standard
  - Formal definition as "Lossless audio that is capable of reproducing the full range of sound from recordings that have been mastered from better than CD quality music sources"
  - Typical sampling frequencies of 96 kHz or 192 kHz at 24 bit
    - 88.2 kHz or 176.4 kHz

https://www.whathifi.com/advice/high-resolution-audio-everything-you-need-to-know



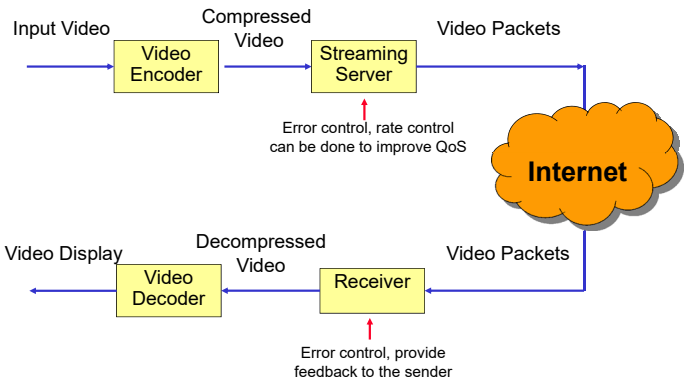
Overview of Sample Application Data

	Throughput	Bit Error Rate	Delay	Jitter
Text (A4 page)	50 kbit/s	$< 10^{-5}$	1 - 10 s	-
Data	some 10 Mbit/s	0	< 1 s	-
Real-time data	some 1 Mbit/s	0	0.001 - 1 s	-
Voice	64 kbit/s	10^{-1}	< 0.25 s	10 ms
CD	1.4 Mbit/s	10^{-8}	< 0.25 s	10 ms
Graphics	some 100 kbit/s	$< 10^{-4}$	< 1 s	-
Pictures	some 1 Mbit/s	$< 10^{-4}$	< 1 s	-
Video (uncomp.)	150 Mbit/s	$< 10^{-2}$	< 0.25 s	25 ms
Video (comp.)	some 10 Mbit/s	$< 10^{-8}$	< 0.25 s	25 ms
"8k" TV (uncomp.)	144 Gbit/s	$< 10^{-8}$	< 0.25 s	25 ms

Networked Multimedia Systems – Examples

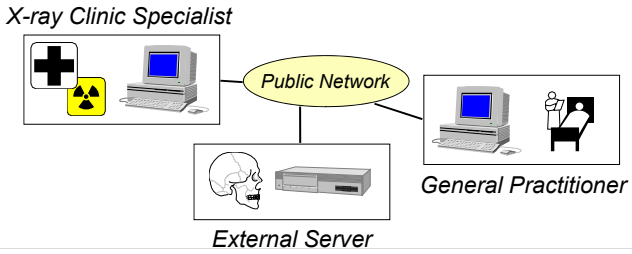
- System 1: Live media transmission
  - Capture, compress, and transmit the media on the fly
  - IPTV, streaming systems
- System 2: Sending stored media across the network
  - Media is pre-compressed and stored at the server
  - System delivers stored media to one or multiple receivers
  - Video-on-Demand (VoD)
- Differences between the two:
  - For live media deliveries:
    - Real-time media capture needs hardware support
    - Real-time compression with correct speed
    - Compression procedure can be adjusted based on network conditions
  - For stored media delivery
    - Offline compression with better compression results
    - Compression can not be adjusted during transmission

Example 1: Media Streaming System

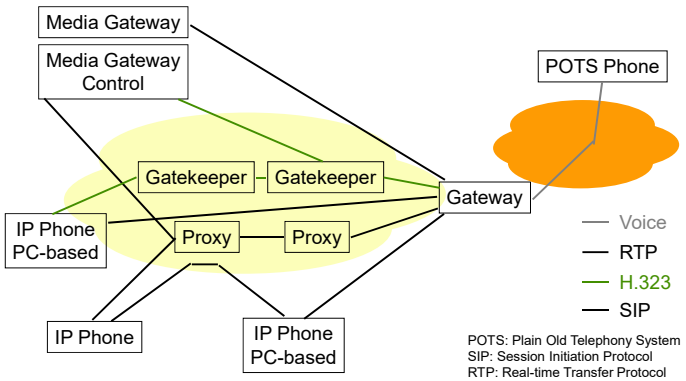


Example 2: Medical Imaging

- X-ray images require high data volumes for storage and transmission (up to 600-1,000 Mbit per image)
- Real-time often necessary (3-D graphics)
- Compression techniques not acceptable



Example 3: IP Telephony Protocol Architecture



POTS: Plain Old Telephony System
SIP: Session Initiation Protocol
RTP: Real-time Transfer Protocol

Signaling Protocols

- Roles of signaling protocols in IP telephony:
  - User location: Users can be at different places at different times reachable by different means.
    E.g., mobile laptops, PC vs. phone, non-permanent IP address.
  - Session establishment: Accept, reject, or redirect a call.
  - Session negotiation: Settle a number of audio, video, shared application streams (addresses, ports, capabilities).
  - Call participation management: Adding, leaving.
  - Feature invocation: *E.g.*, hold, transfer, mute require communications between participants.
- Example protocols: ITU-T H.323 and IETF SIP, SDP.

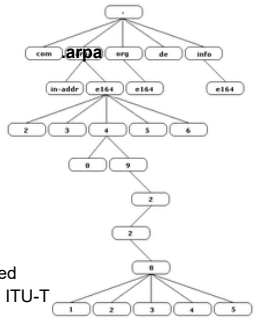
SDP: Session Description Protocol

Session Initiation Protocol (SIP)

- Standardized by the IETF (RFC 3261)
- Signaling protocol to manage sessions in IP networks:
  - Establishment, modification, and termination of sessions
  - Session details negotiation (using SDP, cf. later)
  - User location tracking (mobility support)
  - User availability
- SIP is independent of the type of session being managed.
- SIP addressing:
  - Based on SMTP (Simple Mail Transfer Protocol) addresses.
  - SIP addresses are known as SIP URI (Uniform Resource Identifier)
  - A user is uniquely identified by its AoR (Address of Record)
  - SIP URI examples:
    - stiller@ifi.uzh.ch (AoR)
    - morariu@138.4.7.148
    - +123456789@iptel.org

ENUM (1)

- ENUM: E.164 NUmber Mapping
  - Application of Domain Name System (DNS) to map phone numbers into Internet Protocol addresses
  - RFC 6116 (March 2011)
  - Enables a network-independent reachability
- Example:
  - Phone number 123456789 results in an ENUM look-up and the destination DNS name determination of subscriber123@iptel.org
- ENUM and world-wide phone numbers
  - Phone numbers follow international format E.164
  - ENUM zone represented as DNS hierarchy
    - ENUM mappings are simple DNS entries
  - ENUM Top Level Domain ".e164.arpa" administered by Réseaux IP Européens (RIPE), delegated from ITU-T

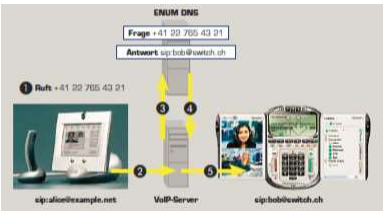


ENUM (2)

- Mapping algorithm
  - +41 44 2345 6789
    Full E.164 number
  - 414423456789
    Deletion of all characters except for numeric characters
  - 987654324414
    Change of order of numeric characters
  - 9.8.7.6.5.4.3.2.4.4.1.4
    Addition of dots between numeric characters
  - 9.8.7.6.5.4.3.2.4.4.1.4.e164.arpa
    Addition of ENUM domain suffix ".e164.arpa"
- Switzerland:
  - SWITCH runs an ENUM DNS since 2007 (formal operations)

Local Example:
UZH moved to VoIP and respective hardware (phones) in July/August 2014

Connection set-up



Example SDP description

- SDP describes multimedia sessions:
  - Streams,
  - Payload types,
  - Addresses (destination),
  - Ports, and
  - Start and stop times.
- Textual format, making SDP a description format.
- SIP contains SDP bodies.

Example SDP Description:

v = 0
o = g.bell 87728 8772 IN IP4 132.151.1.19
s = Come here, Watson!
u = http://www.ietf.org
e = g.bell@bell-telephone.com
c = IN IP4 132.151.1.19
b = CT:64
t = 3086272736 0
k = clear:manhole cover
m = audio 3456 RTP/AVP 96
a = rtpmap:96 VDVI/8000/1
m = video 3458 RTP/AVP 31
m = application 32416 udp wb
a = orient:portrait

Key Question of the Lecture

- How are multimedia applications and scenarios (all separately and combined) supported sufficiently by
  (1) a corresponding communication architecture,
  (2) existing network/multiple networks, and
  (3) "suitable" communication protocols?
  - The answer will be found while taking a look into:
    - Requirements of applications (Part 1a – just done in an overview)
    - Multimedia and communication architectures (Part 1b),
    - Network technologies and prerequisites (Part 1c and 2)
    - Quality-of-Service/Quality-of-Experience models and mechanisms (Part 3)
    - Protocols for multimedia data (Part 4)
    - Selected economic facets (Part 5)

Multimedia and Communication Architectures

Systems, Protocols, Characteristics

Basic Multimedia Assumptions and Etymology

- Multimedia widely used buzzword, but what is "real" content?
  - Many different definitions exist, even "no" definition
  - **Main characteristics** comprise
    - Digital information (uniform representation of data)
    - Produced, stored, handled, transported, and presented (A/D conversions)
    - Computer controlled
      - Standard devices (screen, storage) and modern devices (camera, microphone)
    - Integrated and interactive (not necessary, but most multimedia applications)
- Etymology
  - **Multi** (Latin): numerous and **Media** (Latin), plural of "medium": middle or center
- Some contexts of "media"
  - Press: intermediaries between producers and consumers: Newspaper, radio, TV
  - Telecommunications: physical means of transmission: Cable types, satellite, radio
  - Computer science: storage for digitized data.
    - Magnetic tape, magnetic disk (floppy), optical disk
  - Types of data: nature of expected presentation: Separate audio, video data streams

Requirements of Networked Multimedia

- Requirements of multimedia applications on the network
  - Delay requirements
  - Quality requirements
    - Satisfactory quality of media presentation
    - Synchronization requirements
    - Continuous requirements (no jerky video/audio)
    - Can tolerate some degree of information loss
- Challenges of multimedia networking
  1. Conflict between media size and bandwidth limitations of the network
  2. Conflict between user requirements of multimedia applications and a best-effort network
  3. How to meet different requirements of different users within the same network?

Mechanisms for Networked Multimedia

- Media compression – reducing the volume of data
  Addressing challenge 1:
  - Image compression
  - Video compression
  - Audio compression
- Multimedia protocols (and transmission technology)
  Addressing challenges 2 and 3:
  - Protocols for real-time transmission
  - Quality-of-Service (QoS) support mechanisms
  - Resource reservation
  - Rate/congestion control, error control
  - Network management and economic incentives

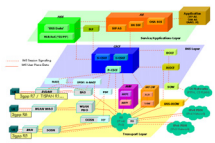
Multimedia System

A multimedia system is **defined** as a system that processes in a computer-controlled fashion more than one media in terms of integrated and interactive, digital information, which include at least one continuous and one discrete medium.

This digital information is represented in terms of data streams forming a sequence of packets, messages, datagrams, cells, or bytes.

MM Conferencing Systems

MM System



IP Multimedia Subsystem

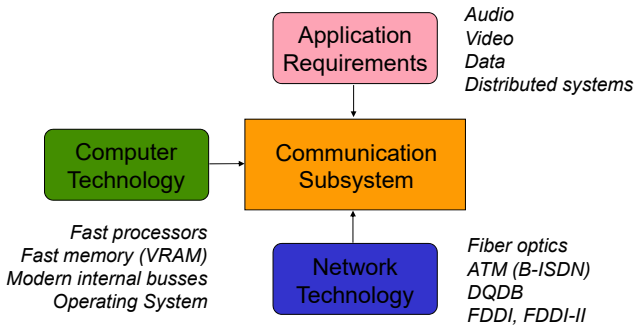
Key Multimedia Requirements

- Integration:
  - Video, audio, and data transfer → Multimedia
  - Multiple services per network → Multi-service
- Performance:
  - High data rates (throughput) and small delays
  - Minimal jitter and low error rates (bit or frame error rate)
- Availability
- Security
- Diversity

- For "critical" applications: safety

General Influences

- Service integration for applications and networks

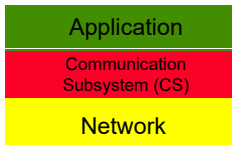


Important Influences of End-Systems

- Computer technology:
  - Type and number of processors
  - Type of memory (RAM, VRAM, …)
  - Internal communication topology (bus, star, …)
  - Network adapter
- Operating System (OS):
  - Structure of the system (I/O, interrupts, memory access)
  - Process and thread management
  - Memory management
- Performance:
  - 20-30% protocol processing; 70-80% OS, I/O overhead
  - Service guarantees require scheduling methods

Communication Architectures

- Applications require services
- Networks provide service
- Question:
  What does interconnect sufficiently services required and provided?
  - A communication subsystem
- Problems:
  - Are network services useful for applications?
  - Is the communication subsystem efficient?
  - What are the influences?

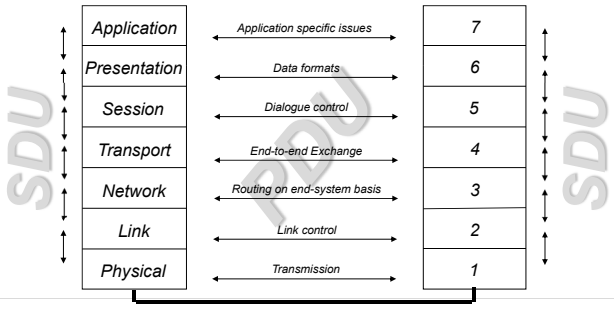


(Ideal situation)

ISO/OSI Basic Reference Model

Reminder

- Concept of hierarchical layering
- Protocol Data Units (PDU) exchanged between peers
- Service Data Units (SDU) exchanged between layers
- Interface Data Units (IDU) exchanged between interfaces



Tasks of BRM Layers (1)

Reminder

- Application Layer:
  - Handle the interface between the communication (sub-)system and applications and
  - Functional standards, *e.g.*, transaction processing (TP), file access and management (FTAM), or virtual terminal (VT).
- Presentation Layer:
  - Establish a common syntax for commands,
  - Provide an agreed upon structure for data exchange, and
  - Encryption and decryption.
- Session Layer:
  - Establish a common context for dialog control and
  - Identify the present message synchronization.

Tasks of BRM Layers (2)

Reminder

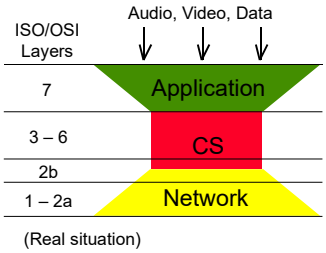
- Transport Layer:
  - Establish and terminate end-to-end connections,
  - Flow control and error control, and
  - Multiplexing.
- Network Layer:
  - Interconnect different networks (routing and addressing) and
  - Congestion control.
- Data Link Layer:
  - Establish and terminate hop-by-hop connections and
  - Flow control and bit error control (checksum).
- Physical Layer:
  - Define physical bit transmission media and devices.

Deficiencies of the BRM

- Communication between layers (inter-lay-com):
  - SDUs, layer-to-layer flow-control, and multiplexing.
- Communication between instances (intra-lay-com):
  - Set-up of connections, 2 · N control packets.
- Transparency:
  - Redundant functionality, *e.g.*, checksum or flow-control.
- Complex sub-layering:
  - Additional layers, such as MAC/LLC on layer 2, and
  - Separation of user data transfer, signaling, management.
- Limited selection of Quality-of-Service.
- Missing service integration:
  - Connectionless or connection-oriented interfaces.

The Service Gap

- The "narrow" Communication Subsystem is due to two reasons:
  - Performance
  - Functionality
- High-speed applications require high-speed nets
- Low delay applications require low delay services
- Isochronous applications require appropriate jitter
- Loss-intolerant applications require well suited error-control



CS: Communication Subsystem

3-Component Model

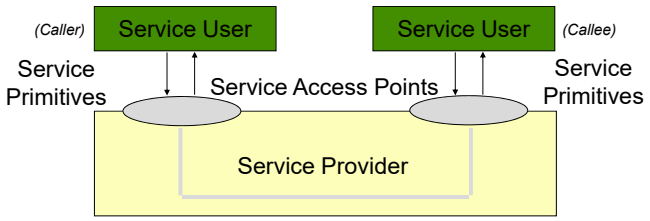
- Topic-dependent structure and coarse-grained layering:
  - ⇨ High flexibility in terms of additional services, less interfaces, and finer functional granularity within components
- Separated protocol functions per each of the 3 components:
  - Applications:
    - Entities within an end-system allow for handling a certain task that delivers specified services to users
  - Communication subsystem:
    - Part of an end-system that processes communication-relevant tasks, *e.g.*, communication protocol processing
    - Based on existing end-systems (hardware: computer) including operating systems (software)
  - Networks:
    - The infrastructure to transport information in form of packets, cells, or byte or bit streams
    - Examples as mentioned above

Application-oriented
Communication-relevant
Network-dependent

Service and Protocol

Reminder

- A service is defined at a Service Access Point via Service Primitives to be accessed by a Service User
- A protocol is specified for a Service Provider using Service Primitives to stimulate control mechanisms



Communication Services

Reminder

- General definition:
  - A service is defined by features and capabilities between two general objects that are related to each other
- ISO definition (ISO/OSI Basic Reference Model):
  - An (N)-service is the capability of the (N+1)-layer and the layer beneath it, which is provided to (N+1)-entities at the boundary between the (N)-layer and the (N+1)-layer
- Service types:
  - Connectionless or connection-oriented
  - Acknowledged or non-acknowledged
- Service primitives:
  - Request, indication, response, and confirmation

Service Phases

Reminder

- Connectionless service:
  - Data transfer phase.
- Connection-oriented service:
  - Connection set-up phase:
    - Synonyms: establish, create, initiate, connect.
  - Data transfer phase.
  - Connection tear-down phase:
    - Synonyms: terminate, close, release, disconnect.
- In case of an "abstract" connection – not being related to a specific layer – the wording of an "association" (in contrast to the ISO/OSI BRM terminology) is used.

Communication Protocols

Reminder

- General definition:
  - A protocol is a set of rules to exchange information between entities in order to provide a service.
- ISO definition (ISO/OSI Basic Reference Model):
  - A set of rules and formats (semantic and syntactic) which determine the communication behavior of entities in the performance of functions.
- Multimedia and high-performance definition:
  - A protocol is a set of rules to exchange information with high-performance between distributed entities across high-performance communication subsystems and networks in order to provide an efficient and specified multimedia service.

High Speed Networking Principles

Characteristics, Services, Architectures, Switching

Characteristics of High Speed Networks

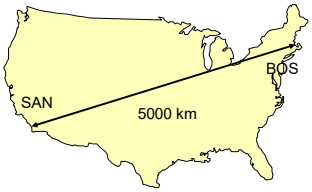
- Characteristics of high speed networks:
  - Low bit error rate (fiber optical media),
  - Higher packet error/loss rate (buffer overflow),
  - Existing Jitter (different buffer lengths),
  - Small transmission units (cells),
  - Many connections (context data),
  - High bandwidth (fiber optical media), and
  - Extreme bandwidth-delay product.
- Protocols have to deal with these issues to alleviate influences of delayed, corrupted, or lost data
- Protocols have to be aware of significant network improvements to reward multimedia applications

Effects on Data in High Speed Transit

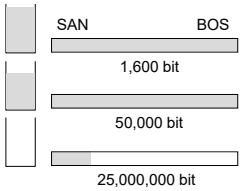
- Buffer overflows dominate occurring errors
  - The effect grows worse for fast and real-time traffic
- Error recovery has to be a trade-off between a waste of bandwidth or extending delays
  - Bandwidth is much cheaper than tolerating high delays
  - Multimedia applications normally do not like delays
- Signal delay is dominating the transmission delay
  - This grows worse for high transmission rates
- A huge amount of data is in transit within high speed and long distance networks
  - ⇨ Path Capacity P_C: $P_C = B * D_{signal}$ B: Bandwidth, D_{signal}: Signal delay.

File Transfer Example

File size: 1 Mbyte
Link: San Diego – Boston
Signal delay: 25 ms



- 64 kbit/s channel:
  - 64 kbit/s * 25 ms = 1,600 bit
  - 0.02 % of file on link
- 2 Mbit/s channel:
  - 2 Mbit/s * 25 ms = 50,000 bit
  - 0.6 % of file on link
- 1 Gbit/s channel:
  - 1 Gbit/s * 25 ms = 25,000,000 bit
  - 8 ms for transmission, 17 ms idle



Bits are smaller – not faster !

Path Capacity Calculations

- Link length: 1000 km
- Optimal physical delay per 1 km: 5 µs
- Transmission delay due to signal delay: 10 ms for the link
- $P_C = B * D_{signal}$: P_C = 10 kbit/s * 1000 km * 5 µs/km = 50 bit

Type of Network	Low Capacity	High Capacity
Transmission Rate	10 kbit/s	1 Gbit/s
Path Capacity: Bits on Link	50 bit	5 Mbit
Delay per Bit	100 µs	1 ns
Delay per 1000 bit	100 ms	1 µs

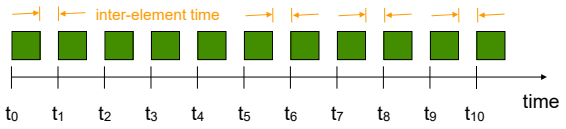
- Round-trip time (RTT): > 20 ms
- Time for 3-way hand-shake: > 30 ms
- 10 Mbit ≈ 157 packets á 8 kByte ≈ 26.000 cells á 53 Byte

Service Categories

- Four main categories (transmission link):
  - Synchronous
  - Asynchronous
  - Isochronous
  - Plesiochronous
- Two main categories (application view):
  - Best effort service
  - Guaranteed service
    - Statistic
    - Deterministic
- Independent structuring of these categories.

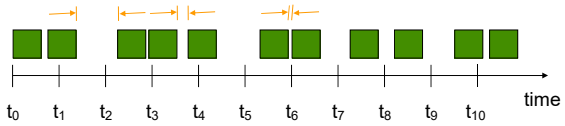
Synchronous Service

- User data are transmitted in data elements, such as
  - Cells, Packets
  - Messages
  - Protocol data units
- Consecutive data units follow a strict timing.
  - Exact inter-element times
  - No variance of this time at all
- A maximum end-to-end delay is defined



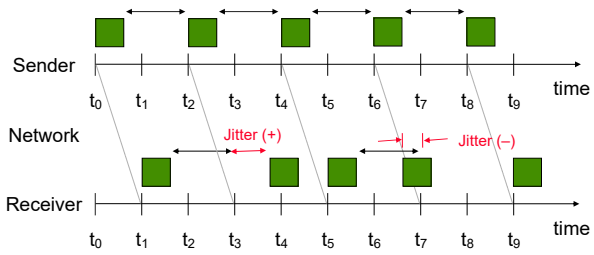
Asynchronous Service

- Consecutive data units do not follow a strict timing
  - Inter-element times may vary
  - Variance of this time does not follow any rules



Jitter (1)

- Jitter or delay jitter is defined as the variance of the inter-element time between consecutive elements
- Jitter is introduced by networks in data flows due to variable transmission delays

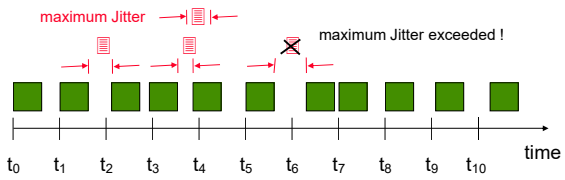


Jitter (2)

- Reasons for jitter (variable network delays):
  - Variable service times of intermediate nodes, such as routers or switches
  - Use of different data paths for data elements of a single flow
- Even sending data continuously does not avoid jitter
- Solution:
  - If a maximum end-to-end delay is guaranteed and a maximum buffer length at the receiver is available, the jitter can be compensated.

Isochronous Service (1)

- Consecutive data units do not follow a strict, but a bounded timing
  - Inter-element time may vary according to maximum/minimum (absolute) values
  - Variance of this time does follow defined rules



Isochronous Service (2)

- Definition of a maximum jitter derives a maximum and minimum delay for all elements, if the sender sends continuously at a fixed rate
- Isochronous services may be offered at
  - The service interface of the network-dependent component
  - The service interface of the communication-relevant component, *e.g.*, a transport service interface
- Isochronous services at the transport interface may be provided even without isochronous transmissions

Architectural View

- Lower layers of an architecture comprise the physical layer and the transmission media in addition to the access to a network
- Local Area Networks (LAN) subdivide the data link layer due to a variety of different access methods, such as token ring, carrier sense, or FDDI
- Modern Wide Area Networks (WAN) take a different approach

ISO/OSI BRM	LAN	WAN
Data Link Layer	Logical Link Control Medium Access	Transfer Modi
Physical Layer	Physical	Physical

Transfer Modi

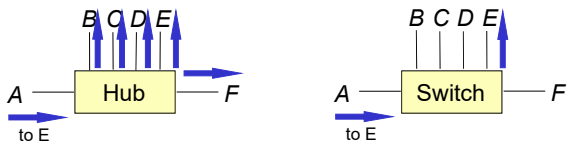
- A transfer modus describes a network technique to handle transmission, multiplexing, and switching of data
- Transmission (Übertragung):
  - Physical methods to inject data – represented as digital bits and transformed into signals – into a network
- Multiplexing (Überlagerung):
  - Methods to cumulate or crack data onto/from channels
  - Time, space, or frequency division multiplexing
- Switching (Vermittlung):
  - Methods to deal with formatted data, *e.g.*, packets or cells
  - Methods to transfer data between entities

Multiplexing

- Transmitting more than one signal over one path
  - Space-division multiplexing:
    - More than one physical path is grouped together
- Splitting a physical path and occupying the two-dimensional continuum of frequency and time
  - Frequency-division multiplexing:
    - Quantity of information transmitted using a certain range of frequency
    - Example, FM radio
  - Time-division multiplexing:
    - Quantity of information transmitted using a certain range of time
- Applying coding techniques onto multiple signals
  - Code-division multiplexing
  - Orthogonal codes, mainly applied in wireless communications

Switching (1)

- Hubs vs. Switches:
  - Similar locations in networks
  - Hubs repeat all packets while switches examine all of them
  - Switches require address examination and forwarding
    - Store-and-forward: Analyze the entire packet
    - Cut-through: Only examine destination and forward
    - Blocking vs. non-blocking architectures
    - Buffering: backpressure or large buffers

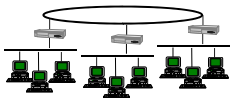


Switching (2)

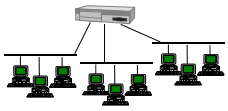
- Handle packets at wire-speed
- Layer-2-Switching:
  - cf. before
- Layer-3-Switching:
  - Combination of switching speed and router functionality
  - Similar terminology: Routing switches or IP switches
  - Identification for common traffic flows on layer 3 and switch these flows on the hardware level for speed. Other traffic will be routed as usual
- Layer-4-Switching:
  - Includes application-level control by applying filters, *e.g.*, security, and QoS-control on specific application flows

Topologies and Switching (1)

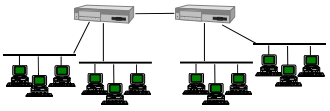
- Backbone



- Backbone Switching (collapsed backbone)

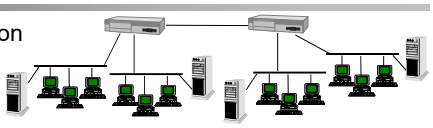


- Multi-switch Backbone

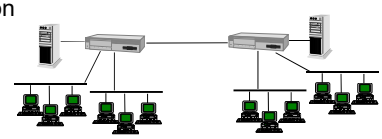


Topologies and Switching (2)

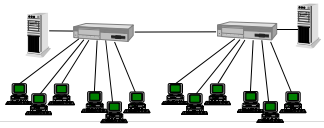
- Workgroup Segmentation (decentralized)



- Workgroup Segmentation (centralized)

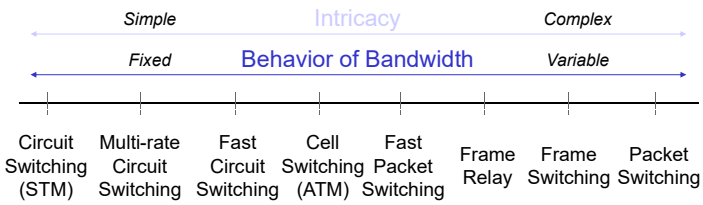


- Micro Segmentation



Switching Techniques – Overview

- Switching Techniques:
  - Based on circuits, cells, frames, or packets.



- Circuit switching suffers from fixed bandwidth constraints for bursty traffic
- Packet switching allows for variable bandwidth

Circuit Switching (1)

- Establishment of a connection (circuit) required
- Example comprises the telephone networks or the Narrow-band Integrated Services Digital Network (N-ISDN)
- Synchronous Transfer Mode (STM):
  - Time-division multiplexing-based
  - Basic unit of transmission is a time slot
  - Periodic frames contain a fixed number of time slots which are dedicated to specific circuits due to the establishment phase
    - *E.g.*, periodicity of 8 bit every 125 µs for 64 kbit/s
    - Static mapping of incoming to outgoing links

Circuit Switching (2)

- Circuit information are stored during establishment times in a translation table (switching table)
- Bit error rates are caused by single bit errors (switch-ing malfunction) or bursts (loss of synchronization)
- Delay is determined by the propagation delay and the processing in switches
  - Bounded to 450µs by ITU-T
- Inflexible, *e.g.*, G.703 PCM
  - Fixed bandwidth

Incoming Link	Time Slot	Outgoing Link	Time Slot
I_1	1 2 … m	O_1 O_2 O_3 …	3 2 … 1
I_2	1 2 … m	O_1 O_2 O_3 …	4 2 … m
I_m	1 2 … m	O_1 O_2 O_3 …	1 m … 1

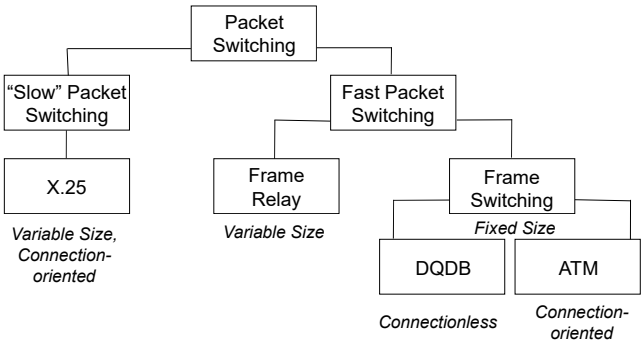
Multi-rate Circuit Switching

- Overcoming the inflexibility of fixed bandwidth
- Multiple basic fixed rate bandwidth circuits are used:
  - *E.g.*, n times m kbit/s
- Limits the static bandwidth behavior
- Increases the complexity of switching systems
- Synchronization between circuits has to be maintained explicitly
- Selection of basic bandwidth m is still a problem:
  - Solution with multiple (i) basic rates m_i feasible

Fast Circuit Switching

- Circuit establishment only, if data are to be sent
- Allocation of resources on a burst basis in terms of:
  - Bandwidth is defined as k times of a basic circuit rate
  - Destination is stored within the switching system
  - Circuit is identified by a header in the signaling circuit
  - On sending data, resources are allocated immediately
- Complexity of the switching system still remains.

Packet Switching Techniques



Packet Switching

- Based on user data that are encapsulated in packets
- Link-based error control in complex protocols
- Variable sized packets require a complex buffering

- Concept is based on technology available in the 60s:
  - Erroneous links
  - Low bandwidth links
  - Software-based protocol implementations
- High delays (due to retransmissions) and low speed (due to protocol processing) avoid the support of real-time and multimedia traffic
- X.25 is the oldest example of packet switching nets

Fast Packet Switching

- Applies to systems operating at high speed switching

- Synonyms of this historic name are:
  - Asynchronous Time Division (ATD) or the well-known
  - Asynchronous Transfer Mode (ATM)
- Asynchronous operation between sender and receiver is possible (no clock dependency)
  - Applying fixed-size packets: cells
  - Reached via inserting/removing "idle packets" (content-less information) in a flow of data
- Main advantage:
  - Transport of data feasible, irrespective of their characteristics, such as bit rate, bursty nature, or quality

Frame Switching and Relaying

- Frame switching became feasible due to higher bandwidths available per link
  - No multiplexing of logical channels required
  - Traditional functionality is performed on the link level

- Frame relaying became feasible due to lower error rates available per link
  - Less protocol functionality on the link level (no retransmissions, flow control, and multiplexing) required
  - An error detection function is required only to discard erroneous frames (CRC checksum)

Frame Relay (1)

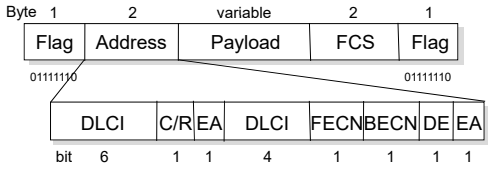
- Frame Relay is a standardized WAN technology specifying the physical and data link layers of digital telecommunications channels using a frame switching methodology.

- Frame Relay supports connection-oriented services
- Subscriber Network Interface (SNI) defined between customer (router) and PTO equipment
  - Support of pure data, not particularly voice
  - Multiplexes flows of data being divided in data blocks
  - Flows are carried in virtual channels which may exceed their bandwidth as other channels are idle
  - Frame Relay may carry X.25 packets/frames
- Performance:
  - Different implementation approaches exist, 2 Mbit/s access speeds are common
  - Insufficient guarantees on bandwidth and delay variation

PTO: Public Telecommunication Operator

Frame Relay (2)

- Physical and data link layer specifications available
- The data link layer is based on LAPD (ISDN):
  - Data link services: addressing (DLCI, local significance)
  - Error control is left out as an end-to-end function
  - Core LAPD frame:



DLCI: Data Link Connection Identifier
C/R: not used
EA: Extended Address

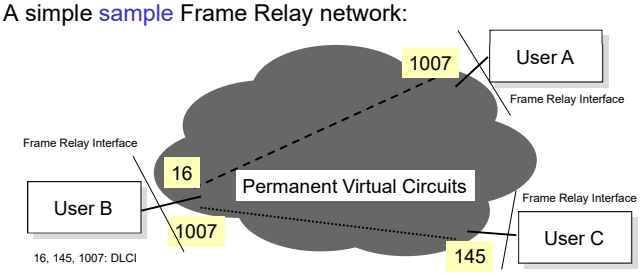
FECN: Forward Error Congestion Notification
BECN: Backward Error Congestion Notification
DE: Discard Eligibility

Frame Relay (3)

- Frame Relay DLCI assignments:

0	Reserved for Call Control Signaling
1-15	Reserved
16-1007	Assigned to Permanent Virtual Circuits (PVC)
1008-1022	Reserved
1023	Local Management Interface

- A simple sample Frame Relay network:



Comparison

- Summary of important functional differences:

	X.2	Frame Switching	Frame Relay
Connection-oriented	5	X	X
Connectionless	X	or X	–
Frame Boundaries	–	X	X
Bit Stuffing	X	X	X
CRC	X	X	X
Error-Control ARQ	X	X	–
Flow-Control	X	X	–
Multiplexing of log. Channels	X	–	–
	X		

Heavy weight ← → Light weight

Technology

CRC: Cyclic Redundancy Check

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