Communication Networks

VITMAB06

IPTV

Gusztáv Adamis
BME TMIT
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Digital TV/radio broadcasting

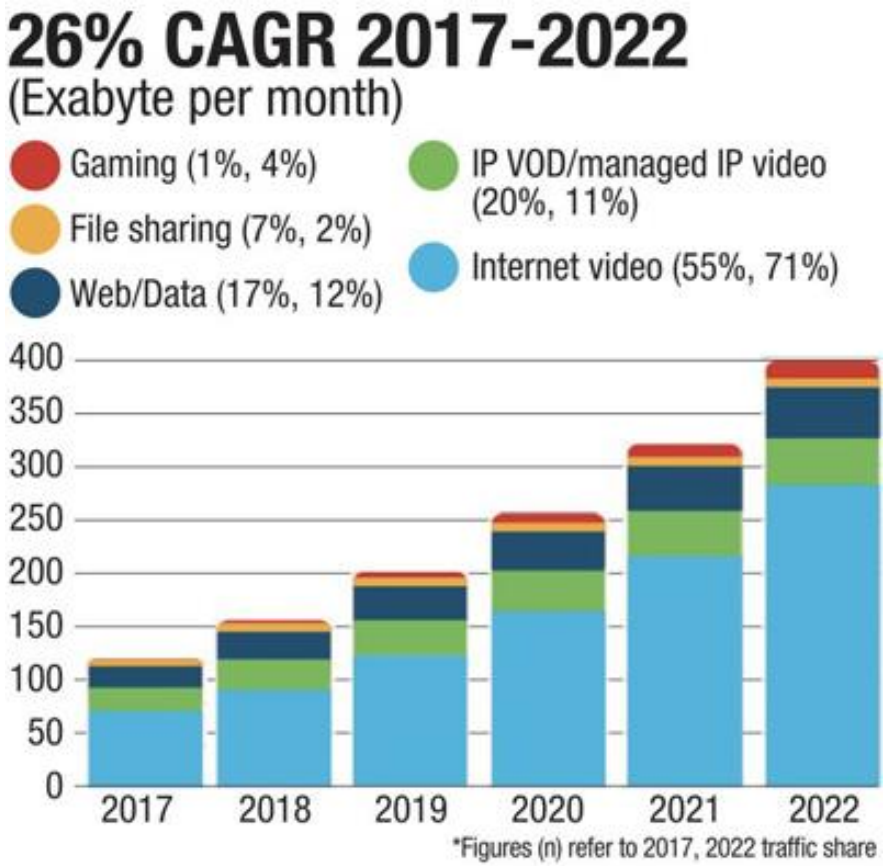
- DVB (Digital Video Broadcasting) systems
  - DVB-T – Terrestrial, transmitters and receiver antennas in houses
  - DVB-C – Cable, digital cable TV
  - DVB-S – Satellite

Other digital TV standards: ATSC (North America), ISDB (South America), DTMB (Asia)

- IP-based broadcasting
  - Internet Protocol Television (IPTV)
  - Internet TV
  - Are they the same?

Evolution of video traffic

You don't need to learn concrete traffic volumes.



CAGR: compound annual growth rate
exa: 10^18= Million*Million*Million

Forrás: https://broadbandlibrary.com/fdx-more-options-for-operators/

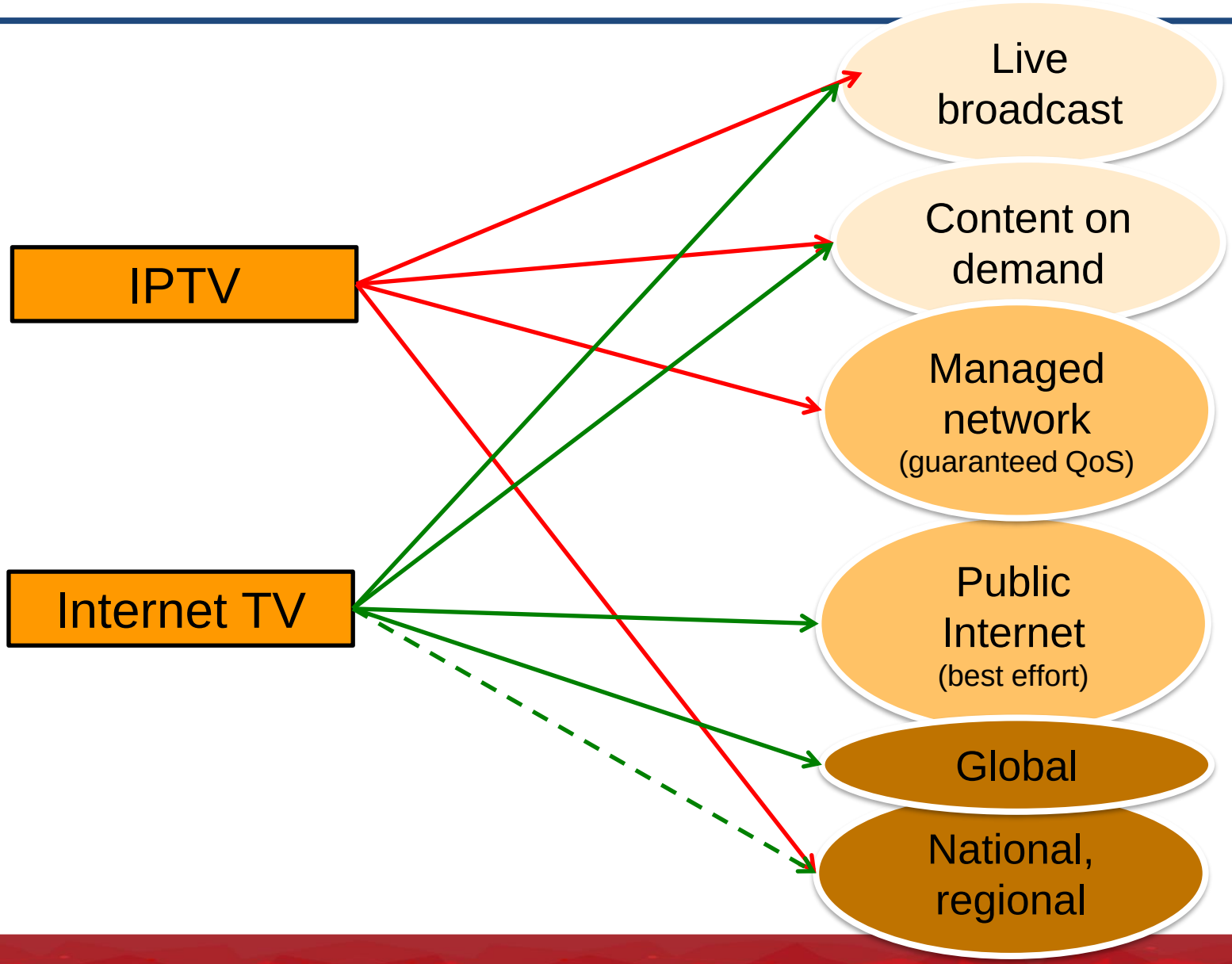
IPTV = Internet TV?

- Accessing the content
  - Live (linear) broadcast – specific broadcast time
  - Content on demand (Video on Demand - VOD) – at any time

- Distribution model
  - On a managed network, with guaranteed QoS, multicast communication
  - On a public network, without guaranteed QoS, unicast communication

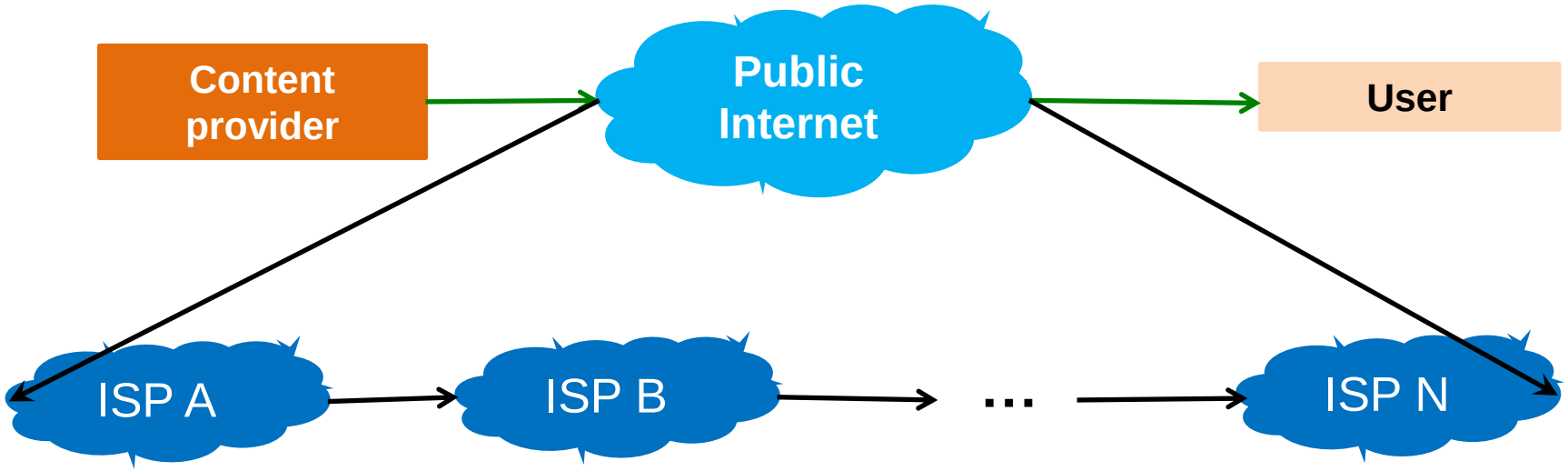
- Area of service
  - National, regional, city-wide
  - Global

IPTV ≠ Internet TV



Internet TV

Over-the-top (OTT) distribution model



- VoD/live video providers: Youtube, Netflix, Hulu, TV.Go, stb.
- Real-time traffic must be transmitted on a best effort (BE) network (there is no guarantee of transmission quality - QoS)
- Unicast client-server model: **high resource demand** (poor scalability) - solution: Content Delivery Network (CDN)

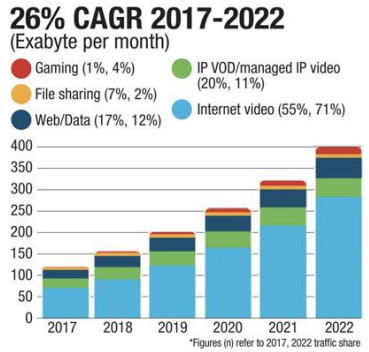
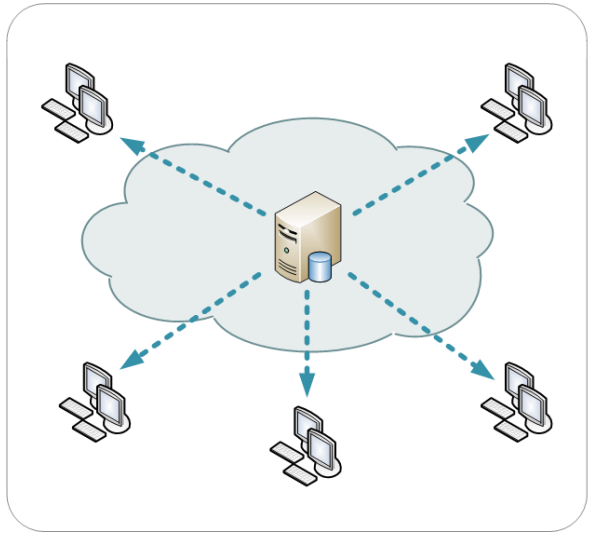
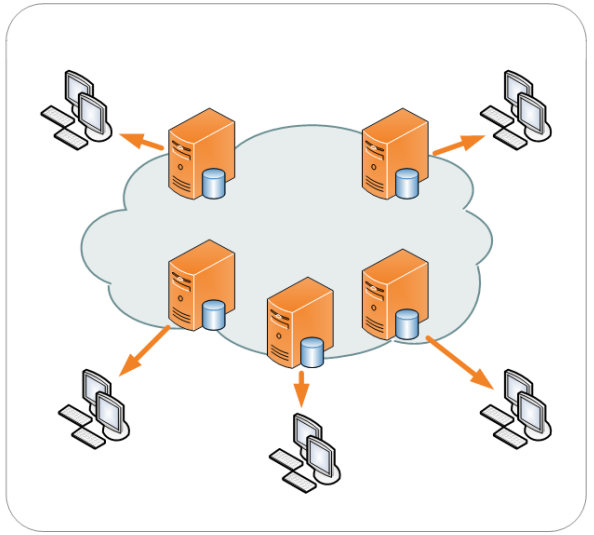


Illustration: Content Delivery Network (CDN)





Real-time on a best effort network

The IP protocol implements best-effort transmission, during which it is not guaranteed that …

- all packets of the message arrive to the recipient,
- the packets arrive in the correct order,
- small and constant transmission delay (no – or at least small – jitter) along the entire path (end-to-end).

Real time transmission – challenges

- A significant part of the real-time services that appeared in IP networks previously operated on dedicated physical infrastructure (e.g. telephone network, cable TV network, video conference system, etc.).
- The technical parameters of the dedicated networks were adapted to the nature of the service and ensured high availability.
- As a result of low construction and operating costs and flexible configurability, most interactive services are now typically available as services over IP

Real time transmission – solution

A technically sustainable solution to the transmission requirements of new applications must be provided

- The inherently democratic (FIFO-type) packet forwarding mechanism of IP routers has been replaced by new mechanisms: dynamic satisfaction of the needs of different types of traffic
- In order to maintain the quality of service at the appropriate level, the traffic of the network must be classified, and the resources must be allocated between the classes: **Quality of Service (QoS)**
- Purpose: end-to-end transmission guarantees

Real time transmission – solution

- Recommended QoS methods (about 20 years ago)
  - Integrated Services (IntServ): RSVP protocol
  - DiffServ
- Which actually works:
  - Partially DiffServ
  - Nowadays rather overprovisioning

Traffic pattern

On-demand content

- Typically, prefetching of longer, but not too long time segments (5-15s)

- The media content arrives at the endpoint in larger pieces (chunks).
- Typically, TCP is the transport protocol

Real-time (live) content

- Shorter prefetching (100-500 ms)
  - It would look silly if a goal was 5-15 seconds late in the match...
- Continuous transmission (non-chunk, almost fixed package rate)
- Typically, UDP/RTP are the transport protocols

Transport protocol (RTP/UDP vs TCP)

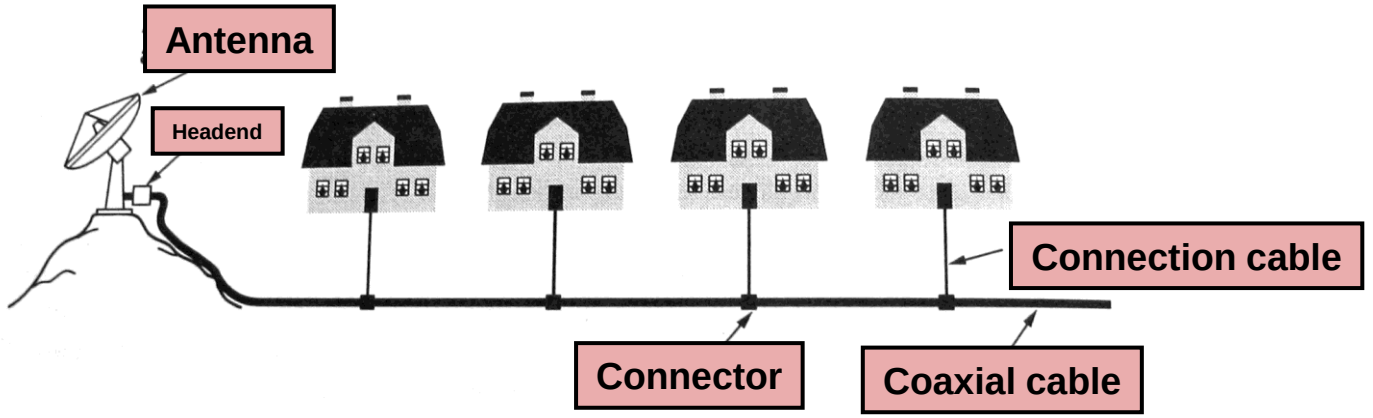
For streaming live media content

- Need:
  - To keep the sequence order (sequence number)
  - Timing – playing with constant speed (time stamp)
  - Low latency
- No need (rather no possibility):
  - To resend – too long
  - Congestion control
    - Lower bitrate is not sufficient – poor quality
    - Higher not needed

IPTV

Early cable TV systems

- Idea: at the end of 1940s (USA)
  - Better receiving in suburbs, in mountains
- Community Antenna Television – CATV
  - A big antenna on a hill
  - One amplifier (headend)
  - Coaxial cable
- Unidirectional transmission: from headend to users

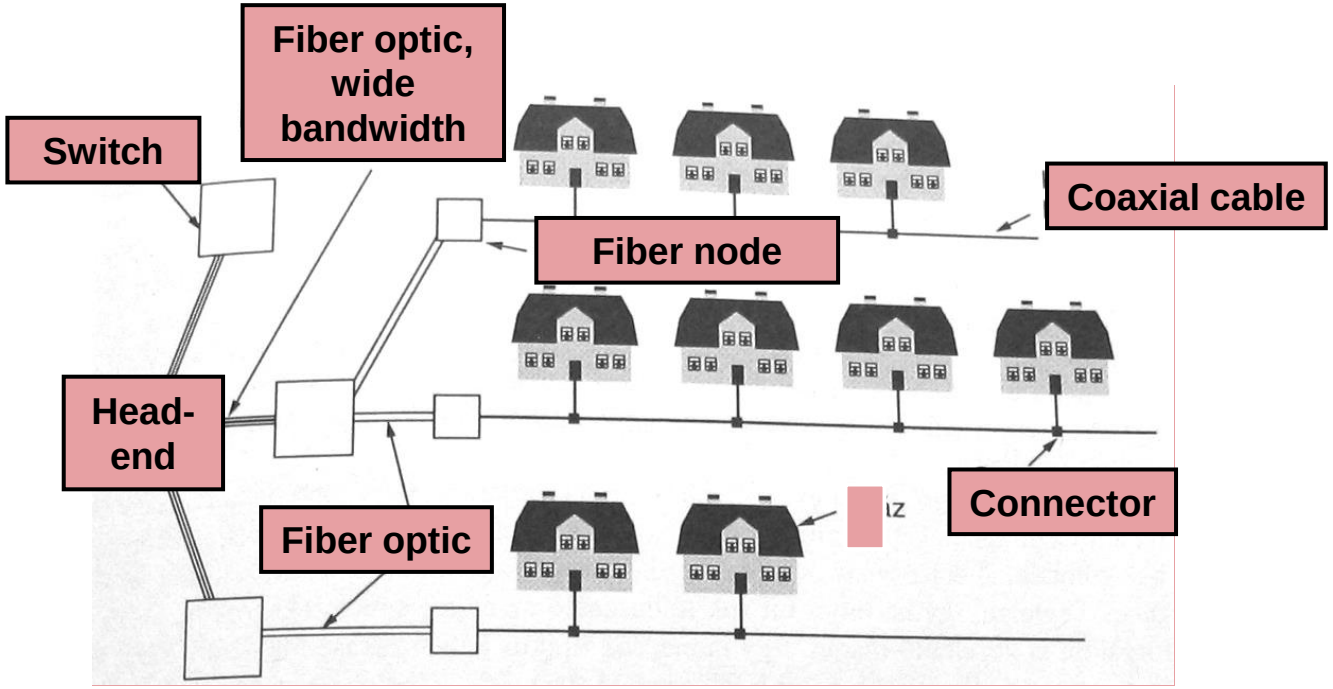


Development of cable TV

- 1970: several thousands independent systems (USA)
- 1974: HBO starts, only on cable
  - Several new cable channels – news, sport, cooking, etc.
- Large companies start to buy the existing cable networks up, new cables are laid down
  - Cables among towns to unite sytems
  - Later: inter-town cables were replaced by fiber optic

HFC system

- HFC - Hybrid Fiber Coax
  - Fiber-Coaxial hybrid system
    - Fiber: for long distance, coaxial cable: to subscribers
  - Fiber Node: FN
    - Electro-optical transformer
      - between the optical and electrical parts



Internet on cable TV

- Cable service providers started to introduce new services:
  - Internet access
  - Telephony (VoIP)
- Network must have been changed
  - Unidirectional amplifiers -> bidirectional
  - Improve the headend
    - „Stupid" amplifier -> intelligent computer system
    - Cable-Modem Termination System (CMTS)

Internet on cable TV

- Coaxial cable: shared resource: several users use in parallel
  - In telephone networks: every user has a dedicated wire pair (subscriber loop)
  - Not necessary for BROADCASTing TV programs
  - Internet: competition among users for the medium
  - But: coaxial cable has a (much) wider bandwidth than a copper wire pair
- Solution: divide a long cable into several shorter sections
  - All of them are directly connected to a fiber node
    - If there are not too many users on a section the traffic may be managed
    - Today: typically 500-2000 subscribers on a section
    - More and smaller sections as the number of subscribers and traffic grow
  - The bandwidth between the headend and fiber node is very high

Secure communication

- Cable is a shared medium
  - Anybody can look in the traffic goes by ☹
- Traffic must be encrypted in both directions

Spreading of IPTV service

- In the beginning, telecommunications providers provided it in their own networks, for their own subscribers
  - There was less and less revenue coming from traditional telecommunications services, so it was necessary to encourage customers to use new services
  - For this, the service must also work on the existing infrastructure
- Traditional cable TV providers have also switched to this
- It is also offered by traditional internet service providers
- They have now come together: triple play

Triple play

- Marketing name for an IP service that includes the following three services:
  - Internet
    - 5 Mb/s at least (old value, now ~100Mb/s)
  - Television
    - Simultaneous reception of at least 3 TV channels per household
  - Telephone
    - Voice over IP (VoIP, voice transmission over IP)
- It is a business model rather than a technology standard
- The carrier medium can be e.g.
  - twisted pair/ADSL (telephone companies)
  - coaxial cable (cable TV companies)
  - UTP (Unshielded Twisted Pair) cable / Ethernet (Internet Service Providers)
  - Fiber optic (all of the above...)
  - Increasingly: also via wireless access

IPTV services

- Transmission of live TV and radio broadcasts on an IP network
- Digital Rights Management (DRM)
- Electronic Program Guide (EPG) – program magazine
- Teletext
- Live broadcast recording: on the client side (set-top box) or on the server side (Catch up TV)
- Picture in Picture (PiP, Picture in Picture)
- Time shifting
- Multiple recordings + live broadcast at the same time (internet bandwidth may constraint)
- Programmed recording based on EPG
- Video on Demand – video library (TV shows, movies, series, etc.)
- Running applications (news, weather, exchange rates, messaging, etc.)

Teletext

- It is now an outdated text delivery system
  - Works also in analog broadcast (!!!)
  - Associated to a TV channel
  - Still exists (though in a lot of countries/channels ceased)
  - See https://teletext.hu/
  - https://teletext.orf.at/channel/orf1/

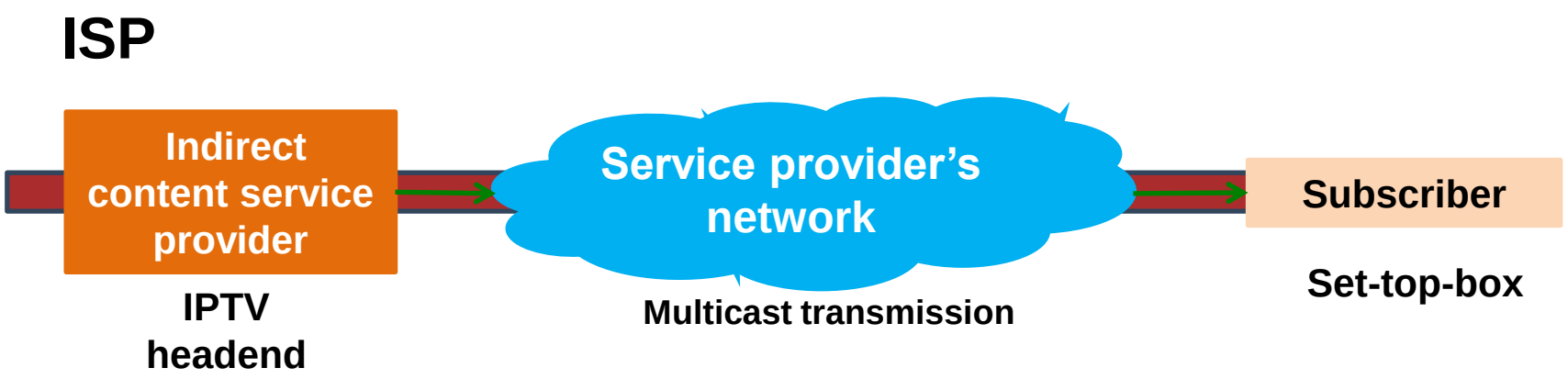
CeeFax (See the Facts) the 1st teletext of the world, 1974 BBC

EPG

- Electronic Program Guide

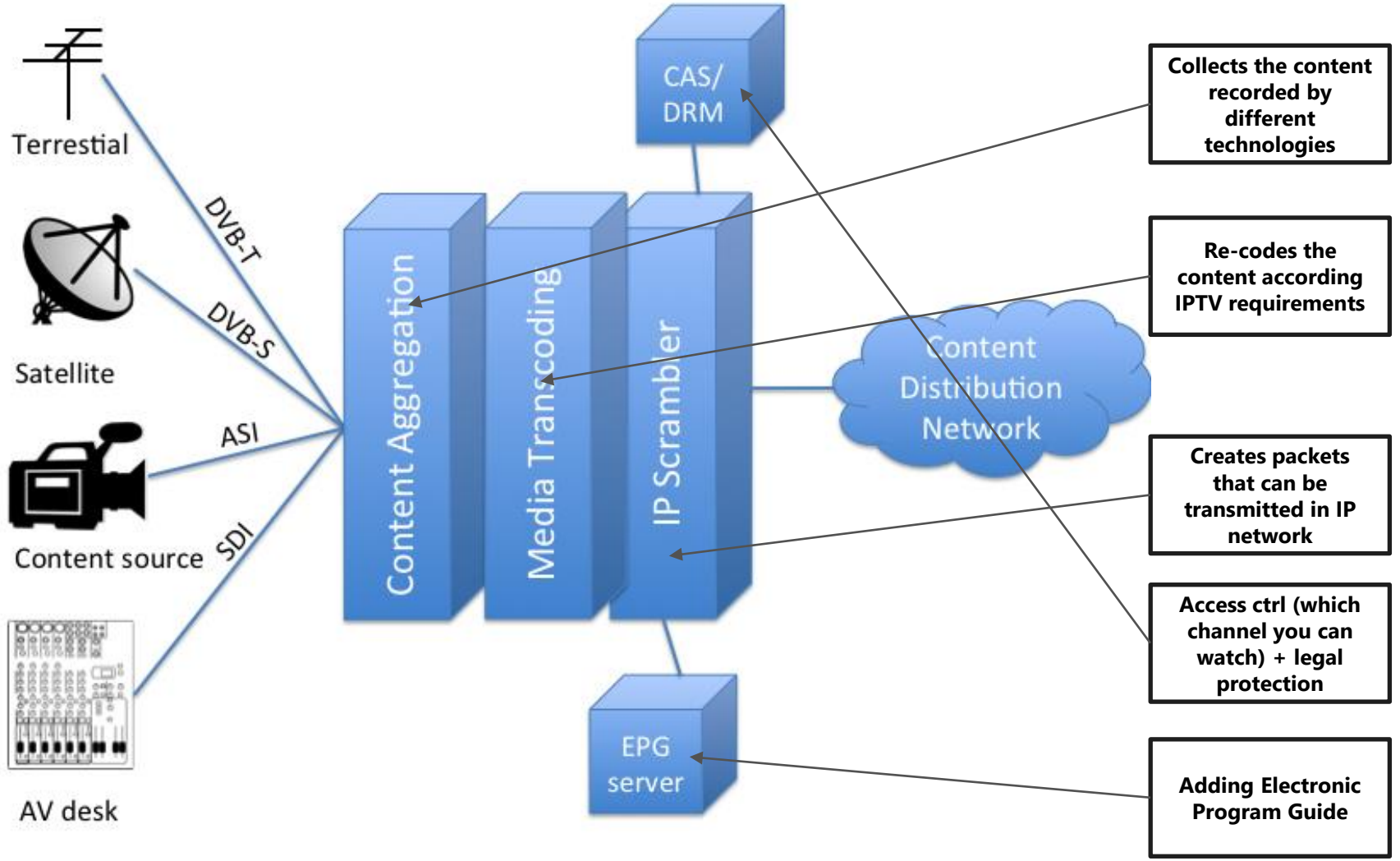
IPTV

Distribution on a managed network

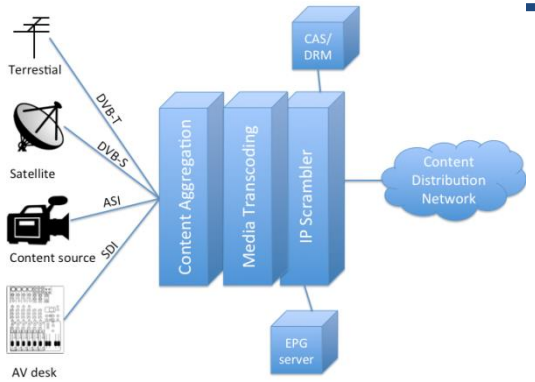


- The service provider provides quality of service (QoS) guarantees for the transmission of the media stream in its own network
- Scalability: multicast forwarding model

Schematic structure of an IPTV headend

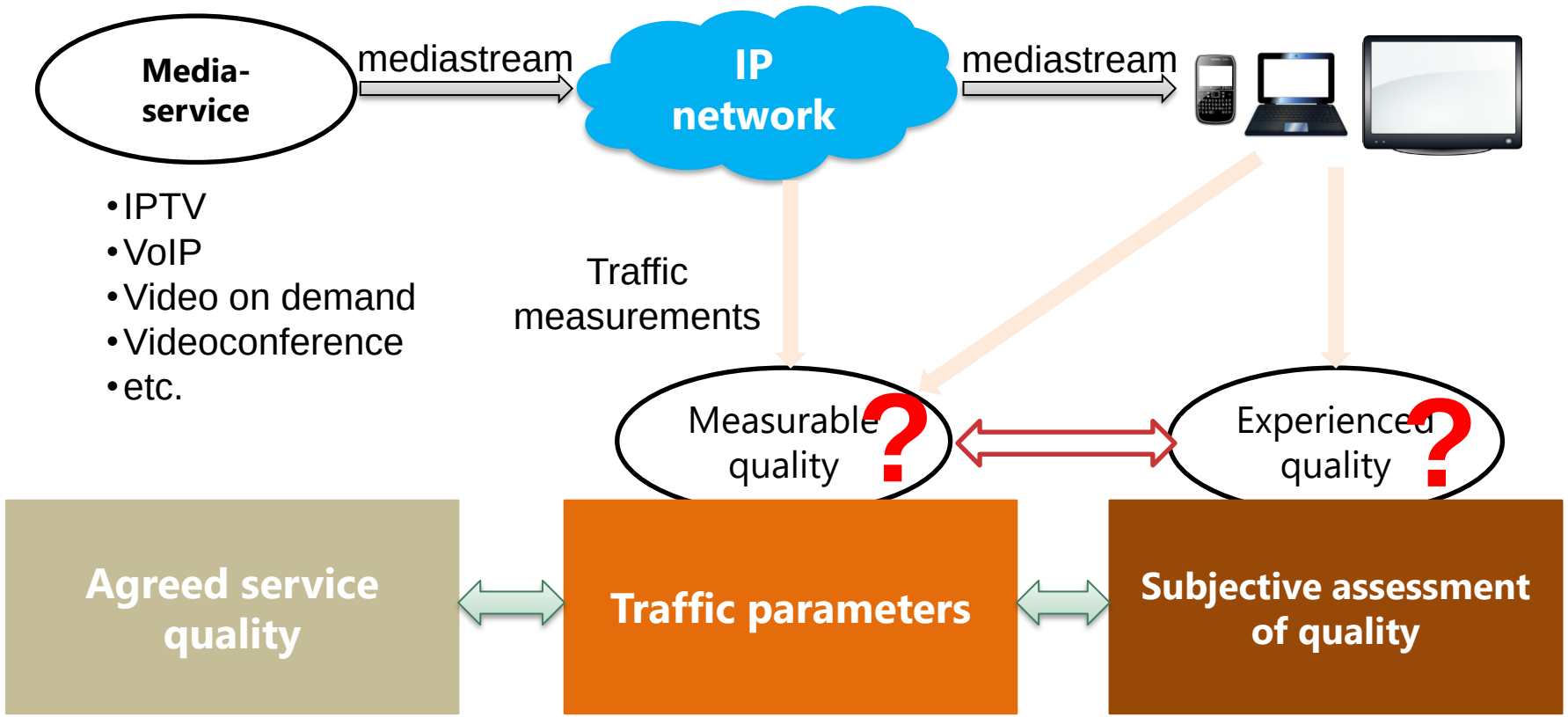


Headend: explanation of the slide



- Content aggregation: collection of program streams from different sources. The content provider (typically the TV company) can deliver the program to the IPTV provider using several technologies: DVB-T (Digital Video Broadcasting - Terrestrial), DVB-S (Digital Video Broadcasting - Satellite), SDI (Serial Digital Interface), ASI (Asynchronous Serial Interface)
- Media transcoding: re-coding of the media stream (image and sound) according to the requirements of the IPTV system: bandwidth requirements, frequency of key video frames, resolution, etc.
- IP Scrambler: Creating data packets that can be transmitted over an IP network from the re-coded media stream: IP/UDP/RTP/MPEG-2 TS/ES embedding
- CAS/DRM (Conditional Access System/Digital Rights Management): access control (to which channel or on-demand content a given subscriber can access), as well as subsystems implementing the legal and technical protection of digital content

Quality of Service



Measuring service quality

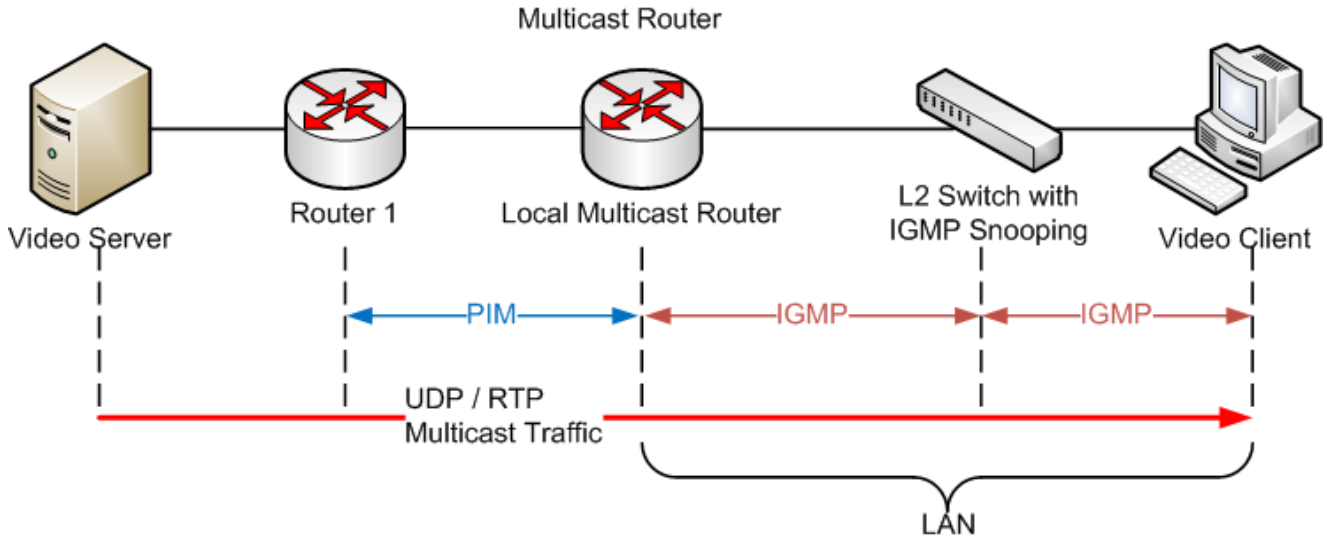
- IP-level elementary network metrics affecting the experienced quality of transmitted media:
  - Delay (ms)
  - Jitter (Delay fluctuation, ms)
  - Packet loss (%)
  - Packet reordeing (%)
  - Throughput, bitspeed, bitrate (bps)
- Complex network metrics affecting the experienced quality of transmitted media:
  - MDI (media delivery index) – double metric
    - Delay factor
    - Media loss rate
- Other metrics affecting service quality:
  - Channel switching time (zap time)
  - EPG loading time

Multicast IP addresses

- For **multicast** packet transmission: 224.0.0.0/4 **(class D) address range** (MSB: 1110xxxx)
- A **multicast address** identifies a well-defined group of nodes: **multicast group**
- Group **membership** is **dynamic**
- In IPTV systems, a multicast IP address identifies the **active viewers of a** given **TV channel** on the network
- The media stream of a TV program is forwarded by the network to the members of the group with the assigned class D IP address

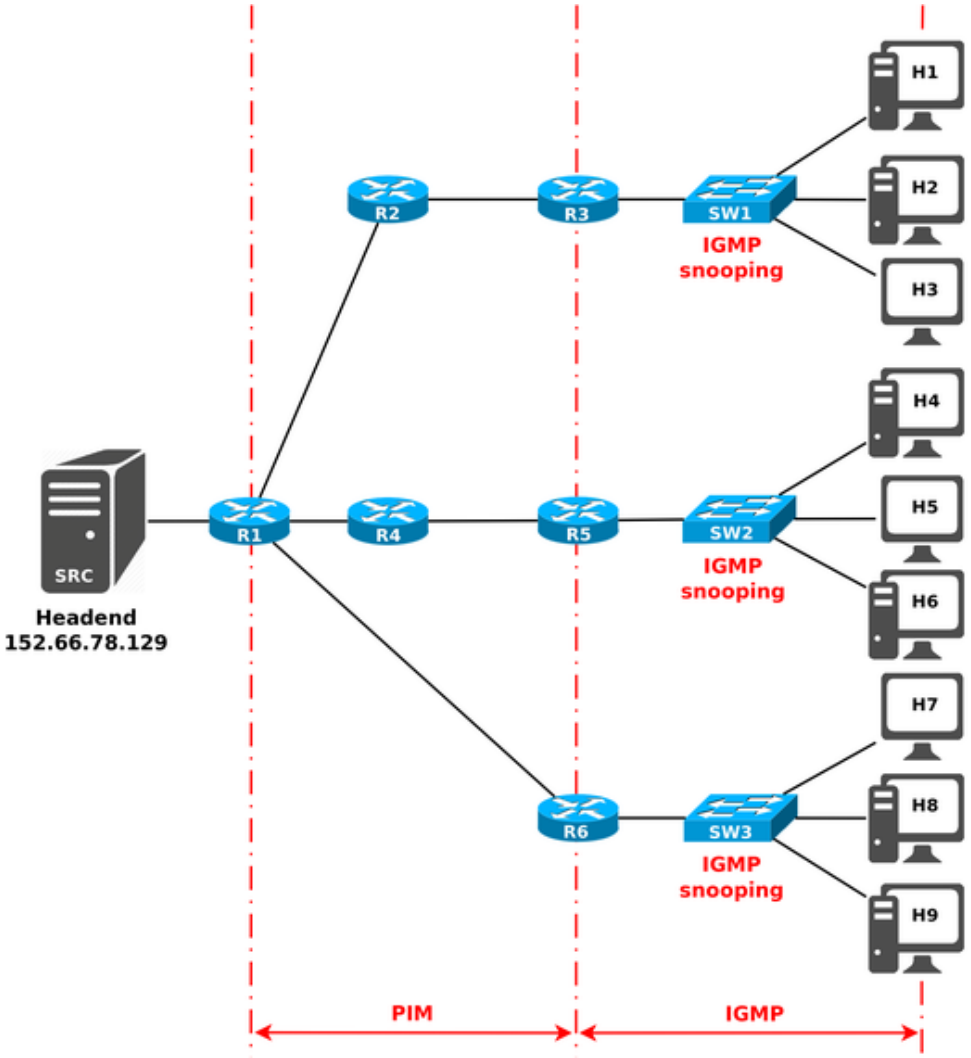
IGMP - Internet Group Management Protocol

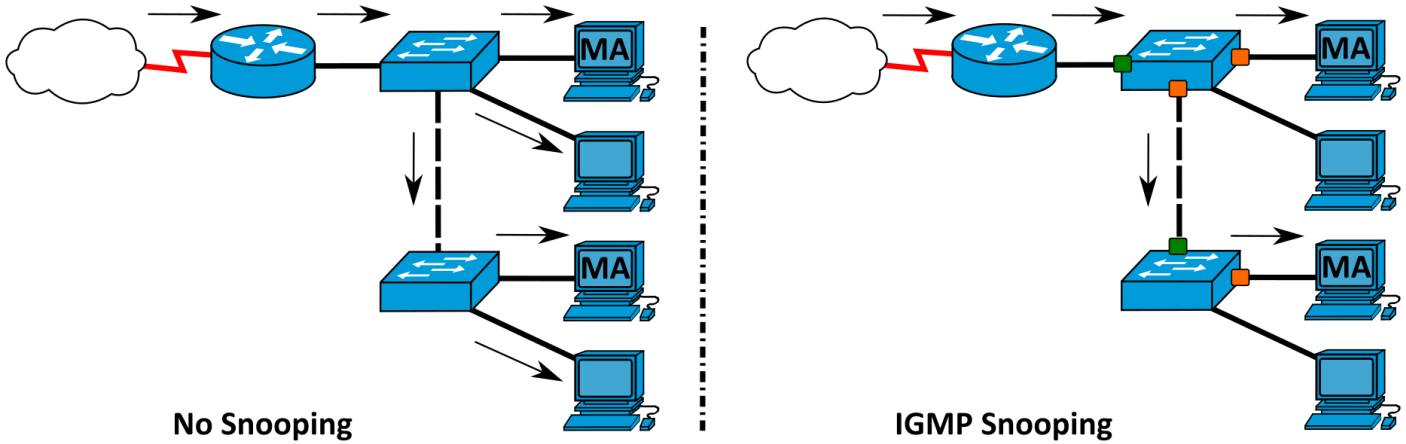
- Management of multicast group memberships within a network
- IGMP message exchange: between IP endpoints and the local router
- OSI classification: part of the IP protocol set, belongs to the network layer
- Roles: IGMP host – IGMP querier

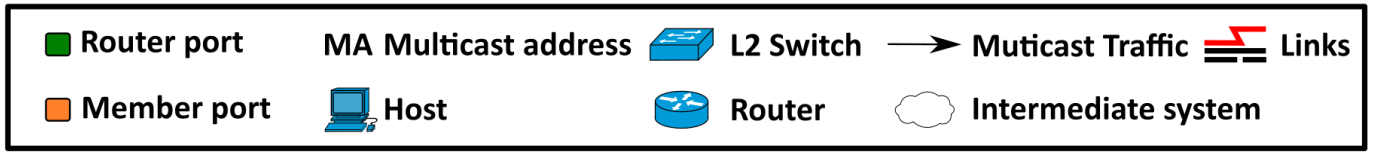


wikipedia.org

IGMP Snooping





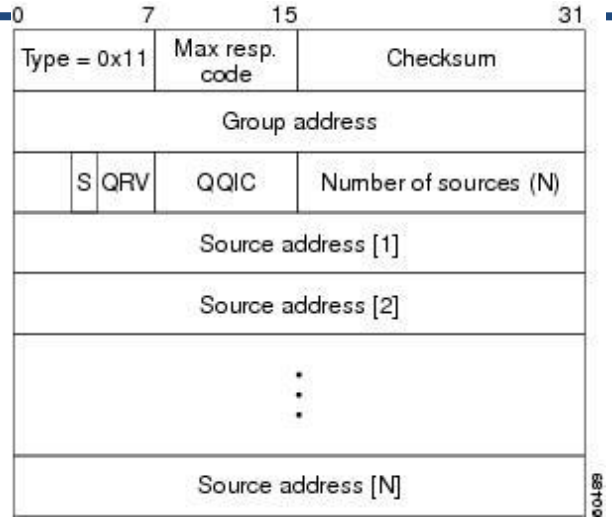


Without snooping: to all *potential* viewer
With snooping: only to *active* viewers

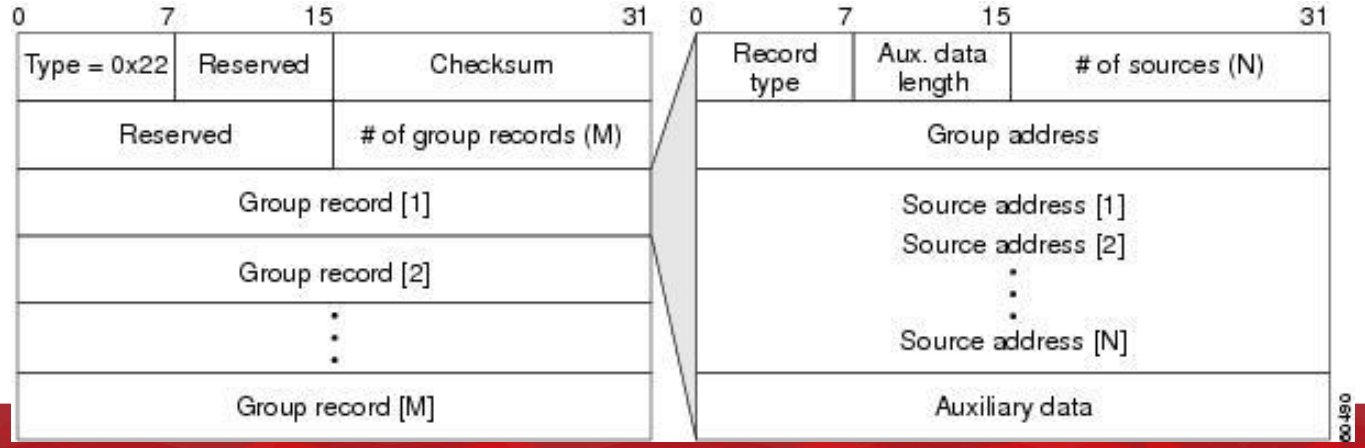
IGMP: Query/Report

IGMP message structure is not an exam material

- **Membership Query** - The multicast group membership of nodes is queried by the router connected to the network (IGMP querier)



- **Membership Report** – Membership Report / Join – Membership(s) reporting to the multicast router (IGMP querier). The node can subscribe to a multicast group with the same message.



IGMPv2/v3

- v2: explicit leave group
  - thus, if there are no more members, the traffic sent to the multicast address in the given network segment can be stopped quickly
- v3: source-specific multicast groups
  - We can specify restrictions (filters) on the address of the source node:
    - Include mode
    - Exclude mode
- Denial-of-service attacks from unauthorized sources can be prevented by filtering the source

Multicast IP routing

- Protocol Independent Multicast (PIM) protocol family
  - PIM is a routing protocol and is therefore used for communication between routers.
  - Three versions of it have spread
    - PIM Dense mode (DM)
    - PIM Sparse mode (SM)
    - PIM Source-specific multicast (SSM)

In IPTV systems: SM and SSM

PIM, SM and SSM - official

- PIM-DM (PIM Dense Mode) – In the first step, the multicast traffic is forwarded by flooding the entire network, and later, based on IGMP messages, the forwarding is suspended to the network segments in which there are no active receivers. The protocol builds the multicast tree implicitly with this method. Due to the intensive bandwidth requirements of the flooding transmission implemented in the first phase of operation, the protocol cannot be scaled to large networks and is therefore recommended for use only in small networks.
- PIM-SM (PIM Sparse Mode) – The multicast tree, the root of which is the so-called Rendezvous Point (RP) router, is built by the protocol based on explicit IGMP requests. The operating model is well suited to serving physically dispersed multicast receivers in a WAN (wide area) network environment.
- PIM-SSM (PIM Source-specific Multicast) – The root of the multicast tree is a predefined source node, for example an IPTV headend. The multicast group is specified with the source identifier/group identifier pair: (S,G), where S is the unicast IP address of the source and G is the multicast IP address identifying the group. Denial-of-service attacks from unauthorized sources within the group can be prevented by explicitly specifying the address of the source

PIM, SM and SSM - practical

- PIM-DM (PIM Dense Mode) – Supposes that most of the group members are interested in + in a locally restricted range (not too many routers)
  - Sent to all, those who are not interested in indicates it
  - D: flooding generates too large traffic + packets can arrive several times
- PIM-SM (PIM Sparse Mode) – Those who want to see must explicitly connect and/or large area
  - Subscription must be learned by the routers – extra task
    - But what happens, if one of these routers die?
    - For long distance connections, what happens if one intermediate service provider does not implement Multicasting?
      - Only in private networks operates well
- PIM-SSM (PIM Source-specific Multicast)
  - Problem with above methods: Possibly anyone in the group can transmit its own content
    - But what if it is an intruder?
    - In SSM the source can be specified

Real-time Transport Protocol (RTP)

Dedicated transport protocol to support real-time media transmission: IETF RFC 3550

Application areas:

- VoIP
- Videoconferencing
- IPTV
- Streaming audio and video

RTP services

- Manages network packet **reordering**
- Provides **timing** for playing message pieces at the right time (jitter compensation)
- Suitable for transmitting both audio and video content
- Supporting IP multicast

RTP header

Only the sequence number and timestamp fields of the RTP header need to be learned for the exam.

RTP packet header

Bit offset[b]	0-1	2	3	4-7	8	9-15	16-31
0	Version	P	X	CC	M	PT	Sequence Number
32	Timestamp						
64	SSRC identifier						
96	CSRC identifiers ...						
96+32×CC	Profile-specific extension header ID						Extension header length
128+32×CC	Extension header ...						

IPTV mediastream

Tulajdonságok	Standard Definition	High Definition
Bandwidth	1,6 - 2,2 Mbit/s	7 - 9 Mbit/s
Image resolution	720 x 576 px	1920 (1440) x 1080 px
Video codec	ITU-T H.264 / ISO/IEC 14496-10 (MPEG-4 Part 10, AVC)	
Voice codec	MPEG AAC/AC3	
Media container	ITU-T H.222.0 / ISO/IEC 13818-1 (MPEG-2 Transport Stream)	
Transport protocol	IETF RFC 3550 Real-time Transport Protocol (RTP)	
Transmission model	IP multicast	

IPTV protocol stack

- IPv4/IPv6
- IGMPv3/PIM
- UDP
- RTP/RTCP
- MPEG-2 Transport Stream
- PES (Packetized Elementary Stream)

Media encoding:

- H.264/H.265 (video codec)
- MPEG AAC (voice codec)

MPEG-2 Transport Stream

- Standard container format for multiplexed transmission of moving images, sound and program and service information messages (Program Guide, teletext, subtitles, etc.).
- Official name of a standard : ISO/IEC 13818-1
- Application areas: DVB, IPTV
- *Several elementary media streams* in one Transport Stream
  - E.g. simultaneous transmission of several audio tracks (different languages)
- *Synchronization* between elementary streams (e.g. sound, image, subtitles)

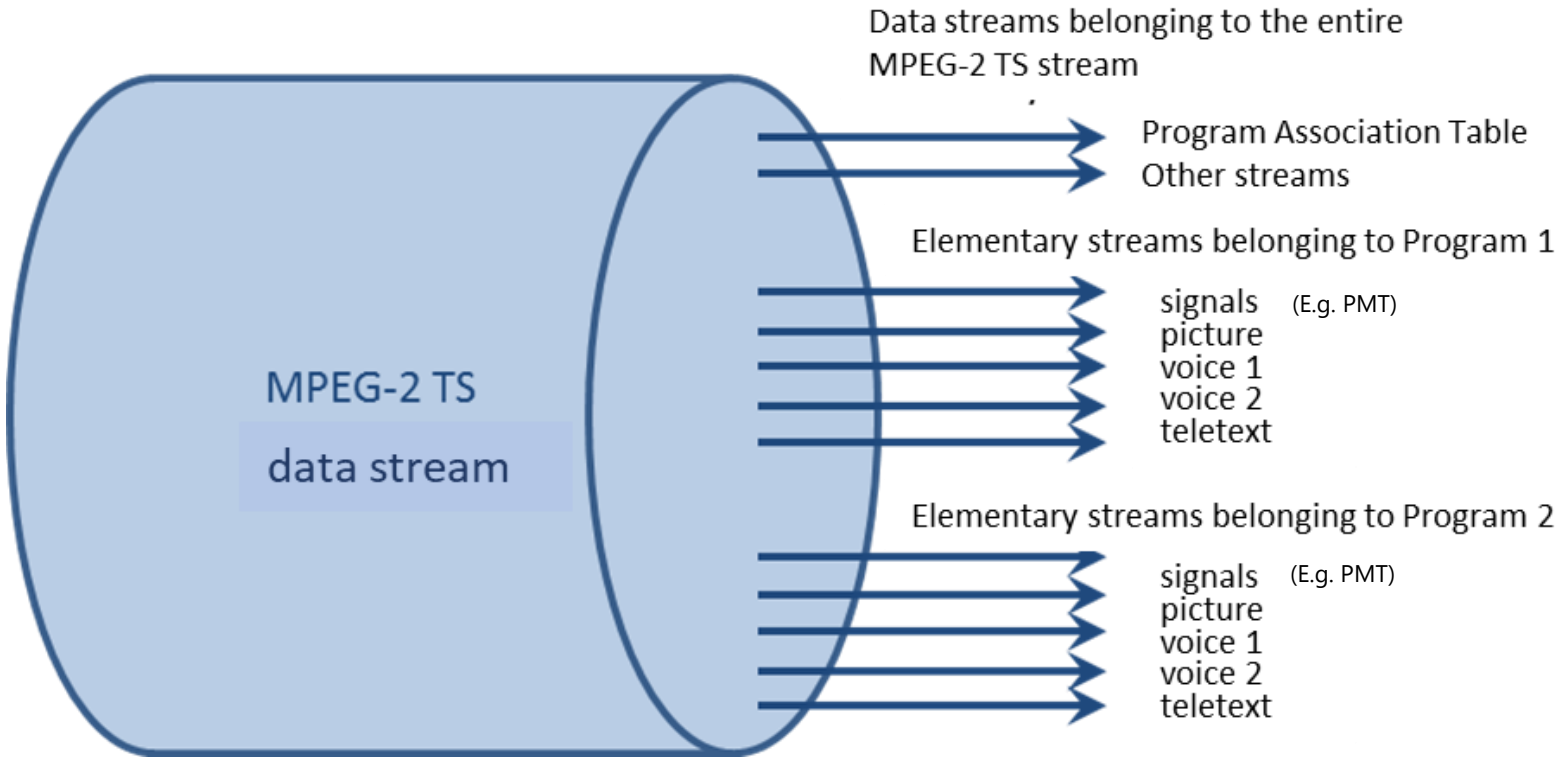
MPEG-2 Transport Stream

- Program = TV channel
- Elementary Stream
  - Video bitstream
  - Audio bitstream(s)
    - several soundtracks in different languages are possible
  - Teletext
  - Subtitles (in different languages)
- Packetized Elementary Stream, PES
  - Large packets of several kB in length
- TS (Transport Stream) packets
  - Small, 188 bytes
- IP packets
  - More (typically 7) TS packet in one IP packet

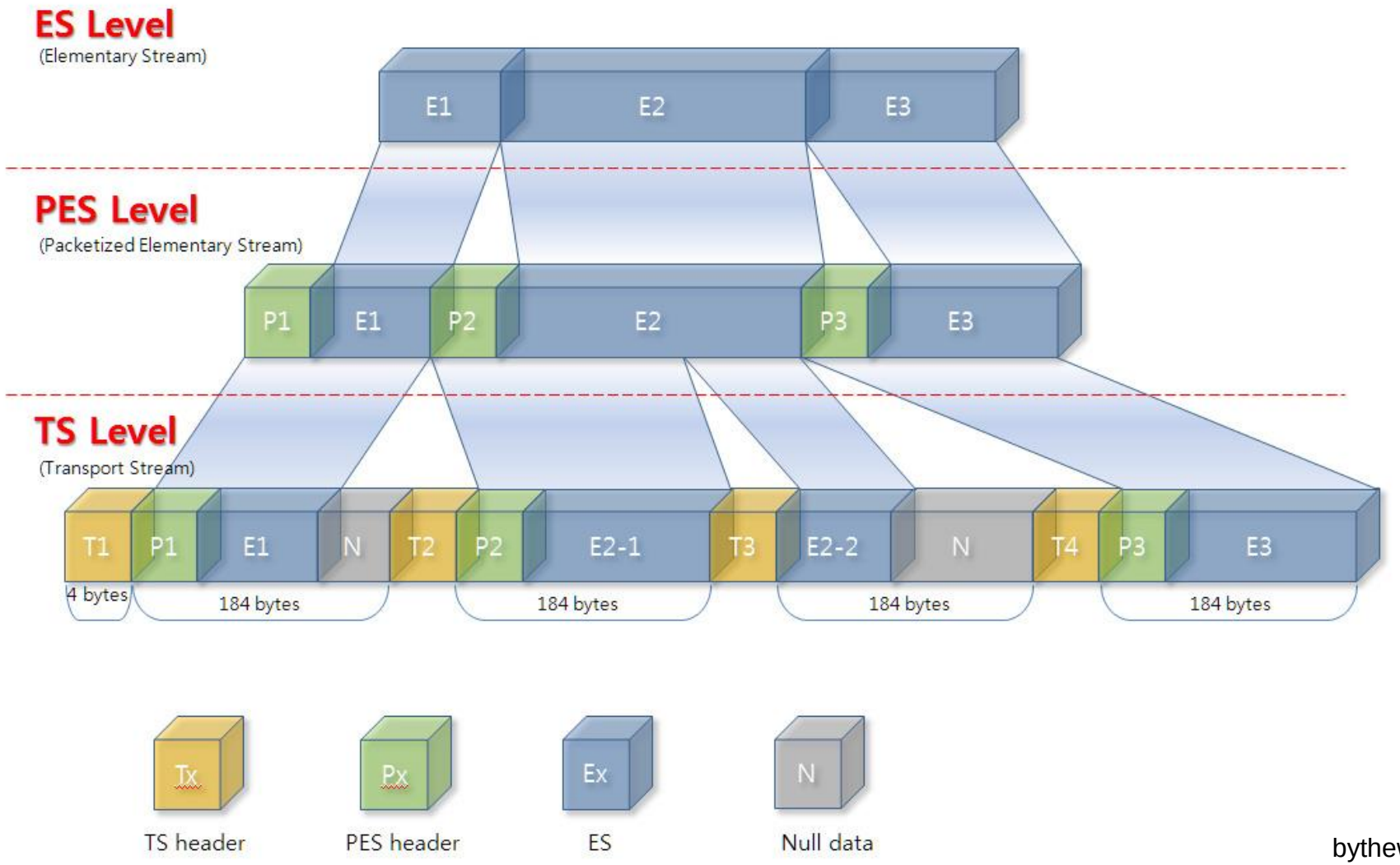
MPEG-2 Transport Stream information tables

- Program Association Table (PAT)
  - table containing the identifiers (PID) of the channels (programs) transmitted in the transport stream
- Program Map Table (PMT)
  - a table containing the PID identifiers of the elementary streams belonging to each program
- Event Info Table (EIT)
  - the EPG subsystem prepares the program guide based on the information of the EIT table
- Program Clock Reference (PCR)
  - the reference clock used to synchronize the elementary streams
  - (This is not actually an information table anyway, just an adaptation field that is included in some of the TS packets)

Structure of MPEG-2 Transport Stream

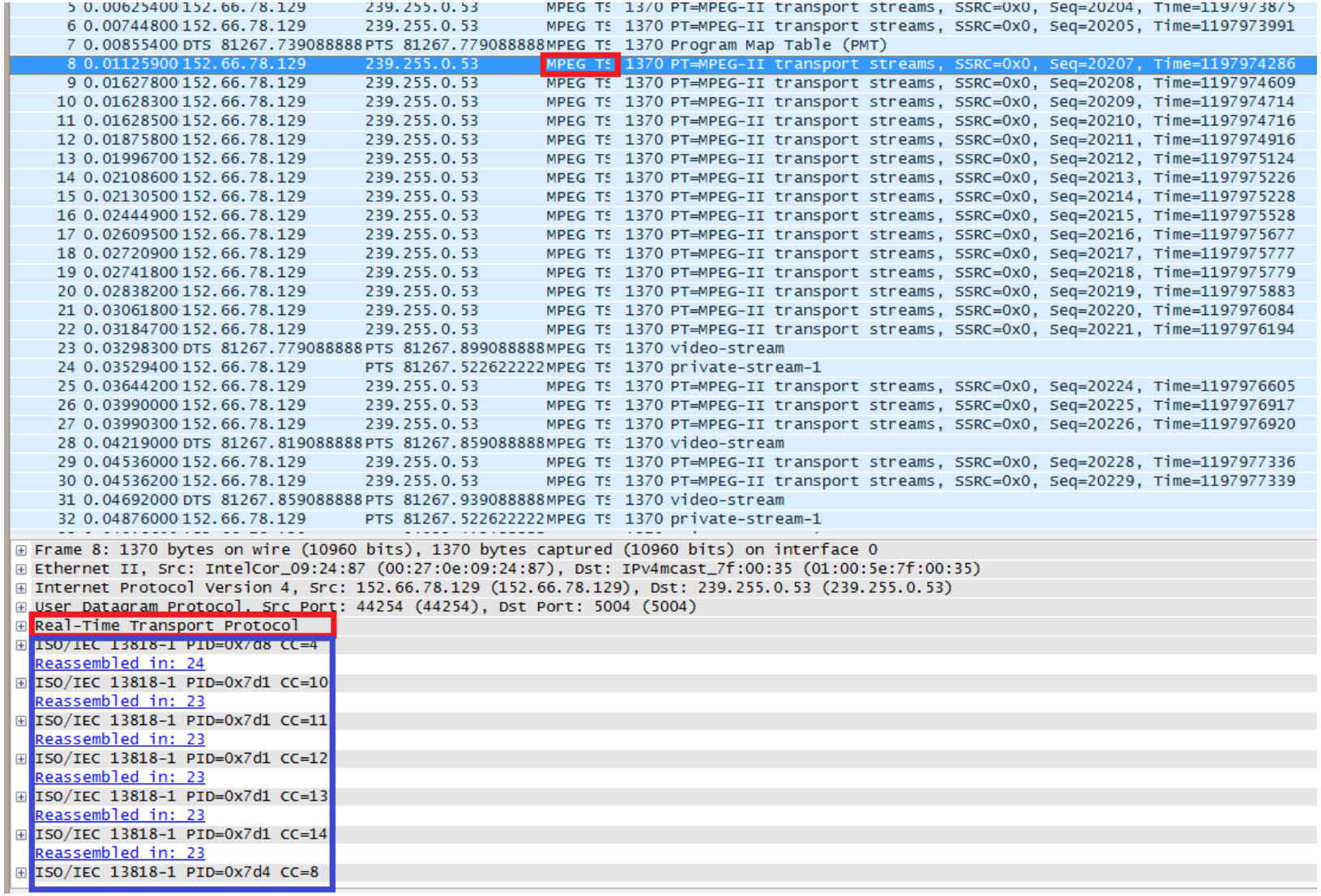


MPEG-2 Transport Stream



bytheway.wordpress.com

Embedding MPEG TS packets into IP packets



Media encoding

- **Constant Bitrate (CBR) mode**

  Produces the compressed media stream with a constant bitrate

  In this mode, the current compression rate is independent of the current complexity of the media content.

- **Variable Bitrate (VBR) mode**

  Produces the compressed media stream at a variable bitrate proportional to the complexity of the media content.

Advantages, disadvantages?

- CBR:
  - Fixed, plannable bandwidth
  - Which may be too large if not too much changes
  - Or may be too small (and producing worse quality) if much changes
- VBR:
  - Constant quality
  - Required bitrate (bandwidth) depends on content -> cannot be estimated

H.264

- H.264, Advanced Video Coding (AVC), MPEG-4 Part 10
- Lossy compression process
- Constant or variable bit rate (CBR/VBR)
- Group of Pictures (GOP)

H.264 frame types + Stream Structure

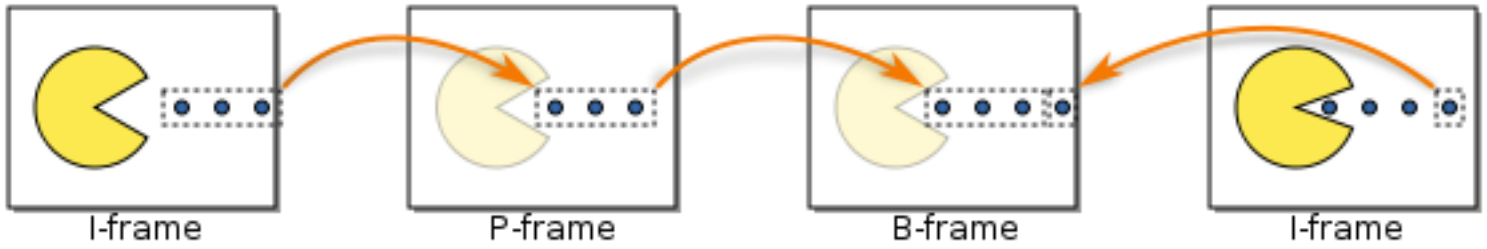
- I-frame (intra coded frame, key frame)
  - No additional frames are required to produce the frame. The starting frame of the group of pictures (GOP)
- P-frame (predictive coded picture)
  - It uses the previous I or P frame as a reference, and describes the change with motion vectors
- B-frame (bi-predictive coded picture)
  - It uses both previous and following I or P frames as a reference

- An H.264 encoded data stream can be divided into Group Of Pictures (GOP) blocks
  - Each GOP starts with an I frame, followed by P and B frames in a predefined order
  - A new I-frame indicates the start of the next GOP

H.264 GOP-structure

Playing order

Encoding/Decoding order



Packet loss involving a keyframe (I-frame)

A video sample containing two groups of pictures (GOP)

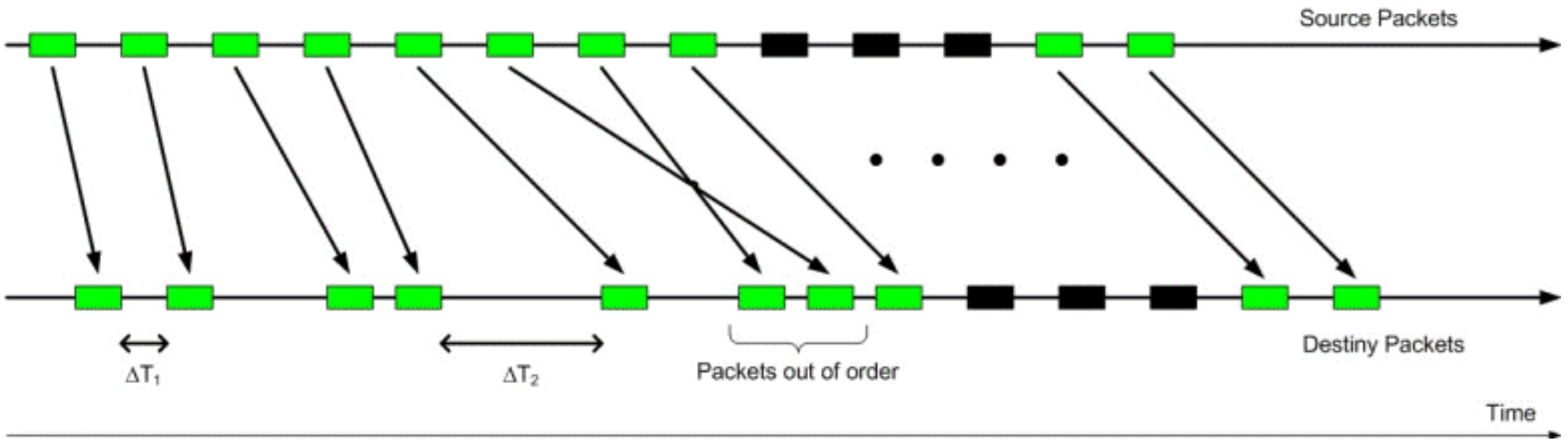
The keyframe of the second GOP is lost

The motion vectors appearing in the second GOP move irrelevant image elements

The Internship (Regency Enterprises, Wild West Picture Show Productions, 21 Laps Entertainment)

Jitter and rearrangement

The media encoders send the media packets at a specific speed and in a given order, on the other hand, the IP packets of the video and audio streams do not arrive from the source to the recipient with a fixed delay and not necessarily in a guaranteed order due to the operation of the network



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De-jitter buffer

- Packets arriving in the buffer with variable time intervals are read out from there at a fixed speed: delay fluctuation (jitter) compensation
- Its negative effect: it increases the delay
  - which should be kept low for real-time applications
- An important design aspect is the correct choice of the buffer size, knowing the network jitter and the expected delay
- Typical IPTV set-top-box buffer size: 80-200 ms

Channel switching (Zap Time)

Factors determining the time of channel switching:

- Group change (IGMP leave and join)
- Extension of multicast distribution tree (PIM routing)
- Playout buffering (De-jitter buffer)
- Waiting for the next key video frame (I frame).