

NGN (Next Generation Networks) – Selected Topics

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Title: NGN (Next Generation Networks) – Selected Topics
Published by: Czech Technical University in Prague
Compiled by: Faculty of Electrical Engineering
Contact address: Technická 2, Prague 6, Czech Republic
Phone Number: +420 2 2435 2084
Print: (only electronic form)
Number of pages: 126
Edition: 1st Edition
ISBN 978-80-01-05294-5
Reviewed by: Stanislav Marchevský

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This project has been funded with support from the European Commission.

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EXPLANATORY NOTES



Definition



Interesting



Note



Example



Summary



Advantage



Disadvantage

ANNOTATION

Mutual cooperation of the various types of networks and their gradual integration into one universal broadband multimedia network (next generation network, NGN) creates conditions for a transmission of all types of media and provision of broad spectrum of the multimedia services and applications. NGN concept has been evolving for a number of years either within ITU or ETSI and this process still continues particularly based on a challenge to provide the new services such as e.g. television (IPTV). This course offers selected topics covering various technologies which can be integrated within next generation networks relating to new communication technologies as well as technologies for digital video delivery.

OBJECTIVES

Main objective of this course is to acquire basic knowledge in the area of Next Generation Network architectures and network components from point of current and future platforms. Participants also become familiar with mobile and optical communication technologies as well as with technologies for digital video delivery such as DVB and IPTV systems. Moreover, they will dispose with knowledge about state of the art technologies like content delivery networks (CDN) and hybrid broadband broadcast television (HBB TV) systems.

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Index

1	NGN	11
1.1	Introduction	11
1.2	Requirements for NGN	12
1.3	Next Generation Network Concept	13
1.4	NGN Softswitch Based Architectures	17
1.5	IMS Based Architecture	19
1.6	IMS Based Architecture – P-CSCF	21
1.7	IMS Based Architecture – I-CSCF	22
1.8	IMS Based Architecture – S-CSCF	23
1.9	IMS Based Architecture – Other Entities	24
1.10	IMS Based Architecture – Self-SIP in IMS	25
1.11	SIP in Service SS	27
1.12	TISPAN	28
1.13	Multimedia Services in NGN Environments	30
1.14	NGN Service Capabilities	33
1.15	NGN Protocols	34
1.16	Fundamental NGN Protocols	36
1.17	Supporting NGN Protocols	41
1.18	Multimedia Services Control Protocol	44
2	Digital Video Broadcasting Technology	47
2.1	Introduction	47
2.2	DVB Services	48
2.3	DVB Standards	50
2.4	DVB System	52
2.5	DVB System - MPEG-2 systems layer	53
2.6	DVB System - Elementary streams	54
2.7	DVB System - Packetized elementary stream	56
2.8	DVB System - Stream multiplexing	57
2.9	Digital Video Broadcasting via Satellite	59
2.10	Digital Video Broadcasting via Terrestrial	61
2.11	Digital Video Broadcasting over IP (DVB-IPTV)	62
3	Mobile Access Network Technologies	63
3.1	Terrestrial Mobile Access Networks	64
3.2	Public Mobile Cellular Networks	65
3.3	GSM	66

3.4	HSCSD and GPRS	67
3.5	3G and 4G Mobile Radio Networks.....	68
3.6	Cordless Telephone (DECT).....	70
3.7	Private Mobile Networks.....	71
3.8	Ad-hoc Networks	73
4	IPTV (Internet Protocol Television).....	75
4.1	Introduction	75
4.2	Architecture of non-NGN Based IPTV	77
4.3	Architecture of NGN Based IPTV	79
4.4	NGN Integrated IPTV (non-IMS) Architecture	82
4.5	IMS Based NGN IPTV Architecture.....	84
4.6	NGN Based IPTV Services	86
4.7	Protocols Used for IPTV	88
5	HbbTV (Hybrid Broadcast Broadband TV)	89
5.1	Introduction	89
5.2	Services	92
5.3	HBB-NEXT	93
5.4	Multiple-User Environment	94
5.5	Multiple Devices	95
5.6	Identity And Trust	96
5.7	Standardization.....	97
6	CDN (Content Delivery Network)	98
6.1	Introduction	98
6.2	Nowadays CDNs in World.....	100
6.3	Content Flow	101
6.4	Control Layer	102
6.5	Distribution Layer	103
6.6	Federated CDNs	106
7	Optical Technologies.....	107
7.1	Optical Networks.....	107
7.2	Optical Access Networks	108
7.3	Optical Access Networks – PON Concept.....	109
7.4	Metropolitan Optical Access Networks	113
7.5	Optical Transport Networks	115
7.6	All-optical Transport Network Architectures.....	117
7.7	Synchronous Optical Network	119

7.8	Burst Switched Network	121
7.9	The WDM Technologies	122
7.10	The WDM Architectures	124
7.11	Development Trends in Transport WAN	125

1 NGN

1.1 Introduction

Opinions on NGN definition may differ in some ways, but the main principles of the **NGNs** (*Next Generation Networks*) were formed when the idea of NGN itself emerged. The next two definitions from ETSI and ITU-T describe NGN in substance.



According to ETSI NGN is a concept for the defining and establishing of the networks, allowing a formal distribution of functionalities into separate layers and planes by using open interfaces, making it possible for the service providers and operators to create a platform which can be gradually developed thanks to creation, implementation and effective management of innovative services [1], [2]. ITU-T defines NGN as a network based on packet transfer, enabling to provide services, including telecommunication services, and is capable of using several broadband transmission technologies allowing guaranteeing QoS [1], [2]. The functions related to services are at the same time independent of the basic transmission technologies. NGN provides unlimited user access to different service providers. It supports general mobility providing the users with consistency and availability of services.

1.2 Requirements for NGN



That is what definitions say, but probably eventual NGN advantages are of bigger importance. Worth mentioning are some requirements for NGN it should conform to [3]:

- High-capacity packet transfer within the transmission infrastructure, however, with a possibility to connect existing and future networks (be it the networks with packet switching, circuit switching, connection-oriented or connectionless, fixed or mobile).
 - Separation of managing functions from transmission features. Separation of service provisioning from the network and ensuring the access via an open interface and thus a flexible, open and distributed architecture.
 - Support for a wide range of services and applications by using the mechanisms based on the modular and flexible structure of elementary service building blocks
 - Broadband capabilities, while complying with the requirements for **QoS** (*Quality of Services*) and transparency. Possibility of a complex network management should be available.
 - Various types of mobility (users, terminals, services). Unlimited access to a variety of service providers.
 - Various identification schemes and addressing which can be translated to the target IP address for the purposes of routing in the IP network. (Flexible addressing and identification, authentication).
 - Converged services between fixed and mobile networks (as well as voice, data and video convergence). Various categories of services with the need of different QoS and *classes of services* (**CoS**).
 - Conformance to the regulation requirements, such as emergency calls and security requirements in terms of personal data protection.
 - Cheaper and more effective technologies if compared to the current technologies.
-

1.3 Next Generation Network Concept



Within the NGN concepts the standardization institutions are solving the following issues and problems:

- existing networks migration towards NGN,
 - development in the field of access technologies,
 - connection of other networks to IP networks,
 - provision of services and development of new ones,
 - interworking in the area of addressing,
 - interworking of signaling systems,
 - roaming a mobility.
-

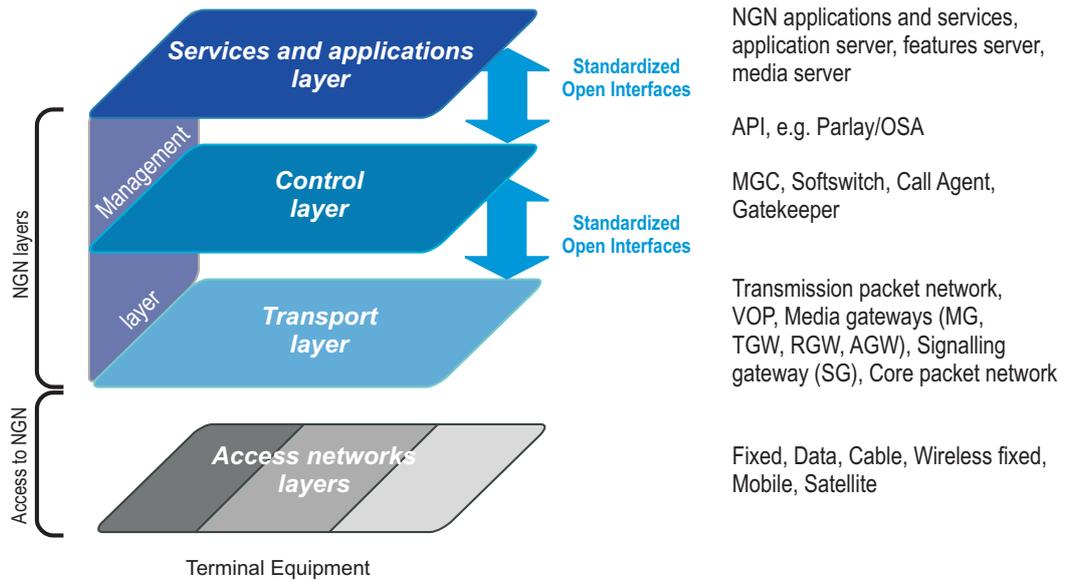
There are many conceptual models and reference architectures for both the converged networks and VoIP architectures. Therefore, we have tried to find common features and to define a suitable conceptual model for NGN.



An objective of the **conceptual model** [4] is to determine functional layers (covering similar functionalities), their entities, reference points (interfaces) and information flows between them. Such a model then can be mapped more easily into the physical reference architecture (and it is independent of the physical entities, i.e. components of the architecture).



In most analyzed cases the NGN conceptual model layers are from the point of view of functionalities divided into independent parts as follows: access (some reference architectures do not include it directly into the NGN model or replace it by the adaptation one), transport (transmission, switching), control (call/sessions control) and application (services).



NGN conceptual model and its functional layers

Conceptual model layers



The access layer provides the infrastructure, for example an access network between the end user and the transport network. The access network can be both wireless and fixed and it can be based on various transport media.

The transport layer ensures the transport between the individual nodes (points) of the network, to which are connected access networks. It connects physical elements deployed in the individual layers. It also enables the transport of different types of traffic, media (signaling, interactive data, real-time video, voice communication, etc.)

The control layer includes the control of services and network elements. This layer is responsible for set-up/establishing, control and cancelling of the multimedia session. It ensures the control of sources as well, depending on the service requirements. One of the fundamental NGN principles is the separation of control logic from the switching hardware.

The service layer offers the basic service functions, which can be used to create more complex and sophisticated services and applications. It controls the progress of the service based on its logic.



In the NGN it is required that the network control is not determined only by the terminal equipment applications, but that the network intelligence may carry out control over the network at all levels of the reference model. The network management reference model implies the following tasks for the network intelligence it has to ensure:

- Resource management (capacity, ports, and physical elements) and QoS in access to the network and in the transport network, as necessary.

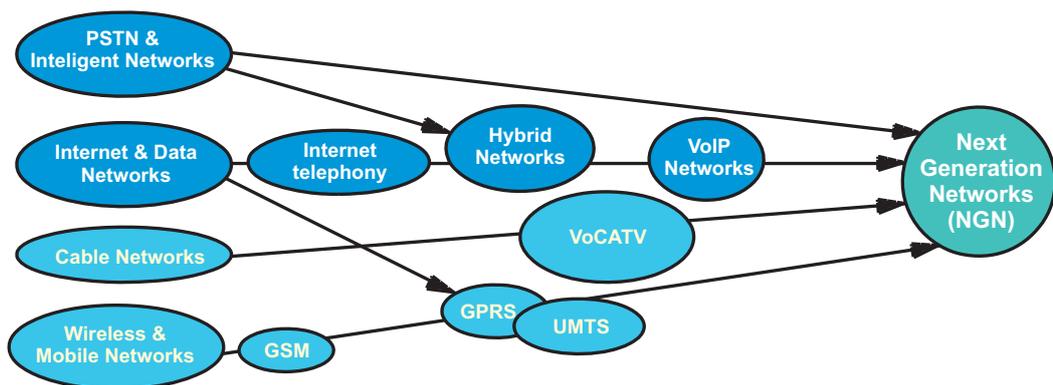
- Various media processing, encoding, data transfer (information flows).
- Management of calls and connection. Management and interworking of all elements of the reference architecture.
- Service control.

NGN evolutional scenarios and multimedia services

As it has already been stated, the next generation networks are a vision of a converged network, meeting all the requirements for a converged universal packet network of the future. The main aim is to explain the deployment and functions of the individual components within the network intelligence and to give a brief characteristic of the individual layers of an NGN conceptual model. After introducing the first real solutions, the next generation networks are becoming a reality, not just a concept. That is why it is appropriate to look into their evolution and to outline their future trends and the open issues to be solved as well. Migration scenarios of different types of networks platforms are based on the idea to integrate **TDM** (*Time Division Multiplexing*) and **IP** (*Internet Protocol*) platforms into one converged NGN platform (from the point of network infrastructure, as well as services, Figure below) [4], [5]. The separation of processes of service control and providing from the physical network architecture and extension of telephone and multimedia services are two different NGN aspects.

+

New concepts and architectures of new generation of **ICT** (*Information and Communication Technology*) based on converged ICT and NGN offer to operators new opportunities to implement and provide wide spectrum of multimedia services and applications [6].

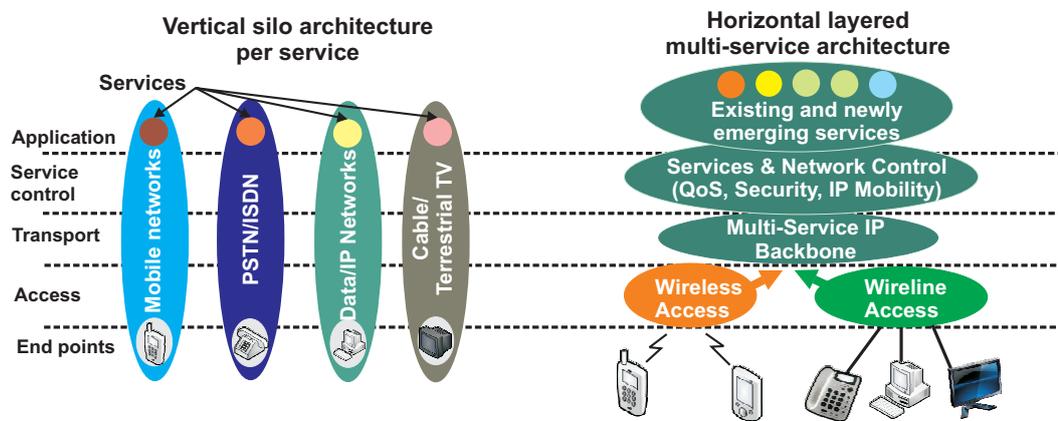


Migration scenarios



Therefore operators can move from vertical silo architecture where each type of service has dedicated access, transport, control and application infrastructure per service to horizontally oriented architecture more independent from provided services. The main idea of NGN based IPTV is to include functionalities and

infrastructure required for any of multimedia NGN services specially here the IPTV type of services to NGN architecture.



From vertical silos to horizontal NGN architecture

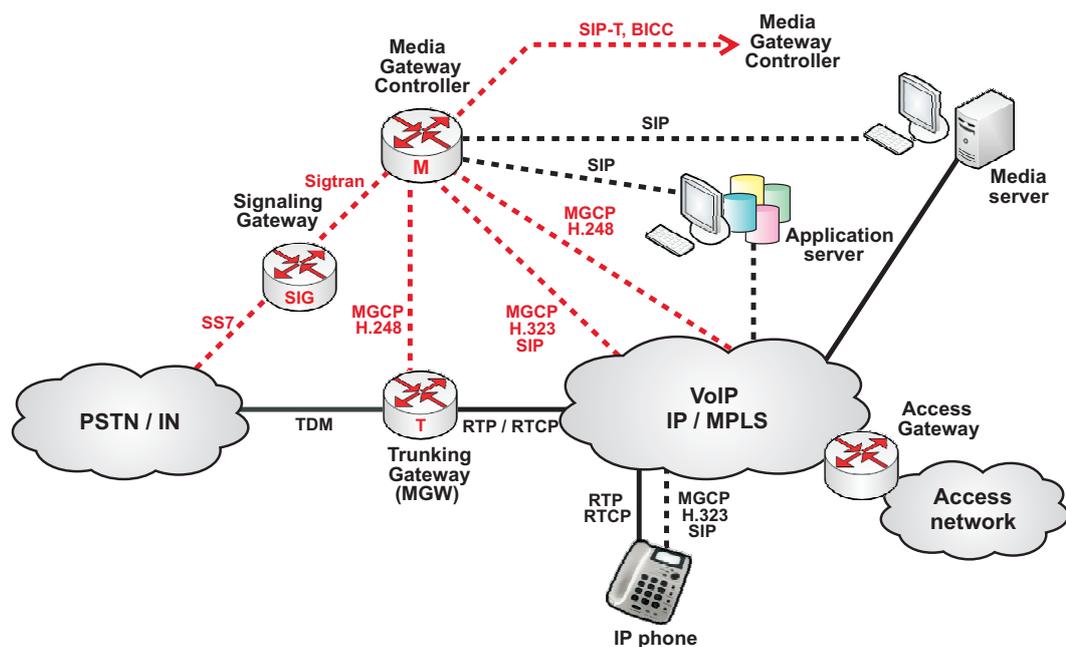
Table shows some of the main parameters and features of network concepts: NGN, PSTN/IN (*Public Switched Telephone Network/Intelligent Network*) and Internet (simplified and generalized interpretation).

Comparing the features of PSTN/IN, Internet and NGN

	PSTN/IN	Internet	NGN
Multimedia services	NO	YES	YES
QoS support	YES (Voice)	NO	YES
Network intelligence	YES	NO	YES
Intelligent terminal equipment	NO	YES	YES
Integrated supervision and control	YES	NO	YES
Reliability	high	low	high
Service creation	complex	ad-hoc	systematic
Simplicity of services use	medium	high	high
Modularity	low	medium	high
Time of service introduction	long	short	short
Openness of architecture	small	high	high

1.4 NGN Softswitch Based Architectures

NGN architecture based on software switching technology (softswitch) can be supposed as first and unique evolution step in NGN, although there are more modern architecture available nowadays (for example see chapter IMS based architecture). However, it has built up philosophy of building of new NGN networks and validated principles and features of NGN architecture and its components towards its next evolution [3]. This architecture was the first which drive was significantly motivated by telecommunication vendors, naturally reflecting on massive development of VoIP protocol family and by telecommunication providers demand to implement services more modern and more efficient way. Due to this fact it is not strictly standardized and we can see there several different attitudes of telecommunication vendors how to provide some features, how to distribute components across network including distribution of functional entities inside control element plane. Knowledge of this architectural approach is important for understanding of next evolved architectures and principles. Components of softswitch based architecture you can see in next Figure.



Components of softswitch based architecture

Media Gateway Controller/call agent/softswitch: generally serves as components for controlling of communication relations of users and other network components; provides call routing, network signaling, billing, and other logical functions.

Media Gateway: operate within transport plane, perform all function related to media physical transport between different networks, media processing functions (transcoding, echo cancellation, jitter managing), tones processing and management of information transport

- Trunking gateways: interface between the PSTN/PLMN and VoIP network
- Residential gateways: provide traditional analog (RJ11) interface to VoIP network
- Access gateways: provide traditional analog or PBX interface to VoIP network
- Signaling network: provide change of signalization systems between PSTN or **PLMN** (*Public Land Mobile Network*) network to VoIP network

Application Server: it is obviously implemented to perform functionalities specific to certain service, perform specialized service logic call control, also includes more functionalities in terms of user web interface, end-points management, etc. For example it can provide specific videoconferencing service, Call Centre service or IP Centrex service.

Media Server: provides functionalities allow interaction between calling party and application using end-point device. It provides Media Resource Functions (tones detection, speech synthesis and recognition, compressions, media mixing, etc.) and Media Control Functions – control of media functions (voice message play management, conference bridge, fax messages management, etc.).

1.5 IMS Based Architecture

The initiative of organization institutions 3GPP within the specifications of **UMTS** (*Universal Mobile Telecommunications System*) architecture (3GPP within the UMTS architecture 5/6) has defined two domains:

- Circuit switching domain,
- Packet switching domain.

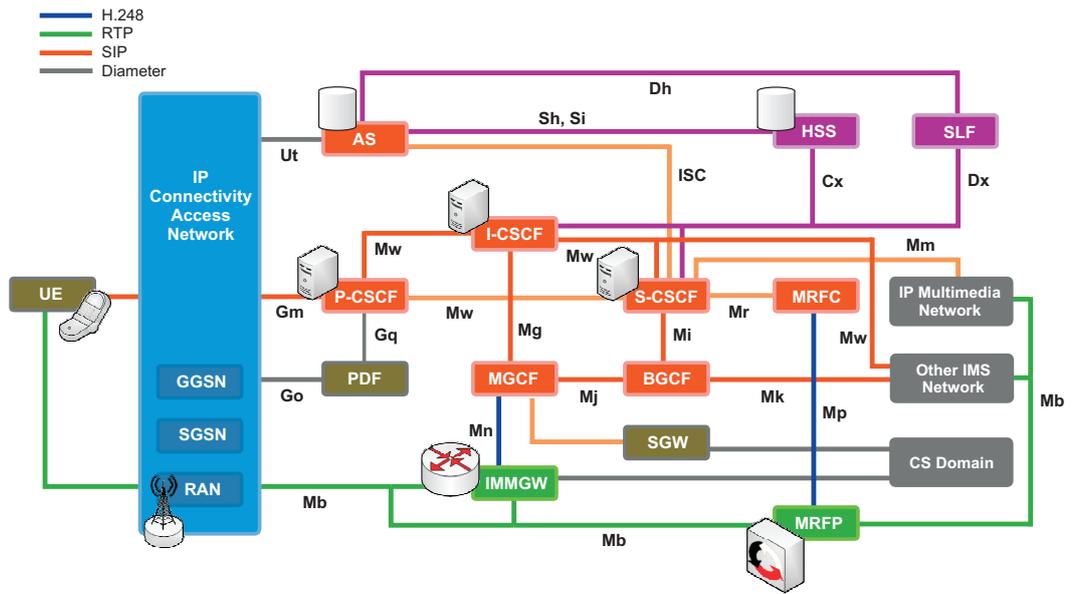
The packet switching domain extends the existing GSM network and other mobile *2nd generation (2G)* networks by the CDMA-based access, while the packet switching domain extends the abilities of the GPRS and other systems of 2.5 generation.



The subsystem for supporting multimedia services, telephony and IP-based message sending, designed in the framework of the packet switching domain is called **IMS** (*IP Multimedia Subsystem*). IMS is based on the IP architecture for multimedia and it was placed as a supporting network element to provide standardized and universal services for mobile users. As it was one of the first concepts on which all the standardizations institutions agreed and which conformed to the NGN principles, it is becoming one of the reference concepts for the fixed networks as well.

The 3GPP adopted **SIP** (*Session Initiation Protocol*) [7], which was originally standardized by the IETF. In time, the 3GPP discovered that there were gaps between the SIP, as initially defined by the IETF, and the features that were required to provide full support for IMS networks. Because SIP did not address all the requirements of IMS networks, the 3GPP subsequently defined dozens of new SIP extensions that are specific to IMS networks, e.g. [8]. Collectively, these extensions comprise the IMS SIP protocol, which is defined in the 3GPP TS.24.229 standard. IMS SIP extensions, such as extended call control, presence and instant messaging, extend the functionality of SIP on IMS networks.

By definition, SIP is not a protocol designed for a specific network or application. To use SIP, you can define the usage profile. Usage profiles work much like templates, and provide a varied, flexible environment for application development in which you can easily develop an application suited to your particular requirements can be easily developed. In effect, this is what IMS SIP did. The IMS SIP usage profile is the most important in the telecommunications industry, as it affects the entire telecom industry and not only mobile networks. The usage profile used by IMS SIP is actually the most appropriate for NGN networks. There are numerous IMS SIP extensions. The figure below illustrates a typical IMS network. Note that all SIP interfaces are shown in orange and specify the name of the interface between two adjoining entities. For example, the AS uses the ISC interface.



SIP in IMS

1.6 IMS Based Architecture – P-CSCF

Proxy Call State Control Function (P-CSCF) performs the following functions:

- is the first contact point for UE within IM CN subsystem, forwards the registration to the I-CSCF to find the S-CSCF and after that forwards the SIP messages between UE and I-CSCF/S-CSCF,
- behaves as like a proxy in RFC 2543, i.e. accepts requests and services the internally or forwards them possibly after translation,
- may behave also like a user agent (in RFC 2543), i.e. in abnormal conditions it may terminate and independently generate SIP transactions,
- is discovered using DHCP during registration or the address is sent with PDP context activation,
- may modify the URI of outgoing requests according to the local operator rules (e.g. perform number analysis, detect local service numbers),
- detect and forward emergency calls to local S-CSCF,
- generation of charging information,
- maintains security association between itself and **UE** (*User Equipment*), also provides security towards S-CSCF,
- provides the *policy control function (PCF)*,
- authorization of bearer resources, QoS management and Security issues are currently open in standardization.

1.7 IMS Based Architecture – I-CSCF

Interrogating Call State Control Function (I-CSCF) performs the following functions:

- is the contact point within an operator's network for all connections destined to a subscriber of that network operator, or a roaming subscriber currently located within that network operator's service area. It can be regarded as a kind of firewall between the external IMSS and the operator's internal IMSS network. There may be multiple I-CSCFs within an operator's network.
- assigns a S-CSCF to a user performing SIP registration,
- routes a SIP request received from another network towards the S-CSCF,
- obtains from HSS the Address of the S-CSCF,
- charging and resource utilization,
- in performing the above functions the operator may use I-CSCF to hide the configuration, capacity, and topology of the its network from the outside,
- additional functions related to inter-operator security are for further study.

1.8 IMS Based Architecture – S-CSCF

Serving Call State Control Function (S-CSCF) performs the following functions:

- performs the session control services for the terminal. Within an operator's network, different S-CSCFs may have different functionality.
- maintains session state and has the session control for the registered endpoint's sessions,
- acts like a Registrar defined in the RFC2543, i.e. it accepts Register requests and makes its information available through the location server (e.g. HSS),
- may also behave as a proxy or as a user agent as defined by RFC 2543,
- interacts with services platforms for the support of services,
- obtain the address of the destination I-CSCF based on the dialed number or SIP URL,
- on behalf of a UE forward the SIP requests or responses to a P-CSCF or an I-CSCF if an I-CSCF is used in the path in the roaming case,
- generates charging information,
- security issues are currently open in standardization.

1.9 IMS Based Architecture – Other Entities

Media Gateway Control Function (MGCF) provides the following functions:

- protocol conversion between ISUP and SIP,
- routes incoming calls to appropriate CSCF,
- controls MGW resources.

Media Gateway (MGW) provides the following functions:

- transcoding between PSTN and 3G voice codecs,
- termination of SCN bearer channels,
- termination of RTP streams.

Transport Signaling Gateway (TSG) provides the following functions:

- maps call related signaling from/to PSTN/PLMN on an IP bearer,
- provides PSTN/PLMN <-> IP transport level address mapping.

Multimedia Resource Function (MRF) provides the following functions:

- performs multiparty call and multimedia conferencing functions.

The S-CSCF, possibly in conjunction with an application server, shall determine that the session should be forwarded to the PSTN. The S-CSCF will forward the Invite information flow to the *Breakout Gateway control function (BGCF)* in the same network. The BGCF selects the network in which the interworking should occur based on local policy. If the BGCF determines that the interworking should occur in the same network, then the BGCF selects the MGCF which will perform the interworking, otherwise the BGCF forward the invite information flow to the BGCF in the selected network. The MGCF will perform the interworking to the PSTN and control the MGW for the media conversions.

1.10 IMS Based Architecture – Self-SIP in IMS

SIP and SDP as a protocol has been selected to some and IPv6 as the only solution to all of the IP Multimedia Subsystem interfaces.

As shown by the figure below the basic SIP has been selected as the main protocol on the following interfaces:

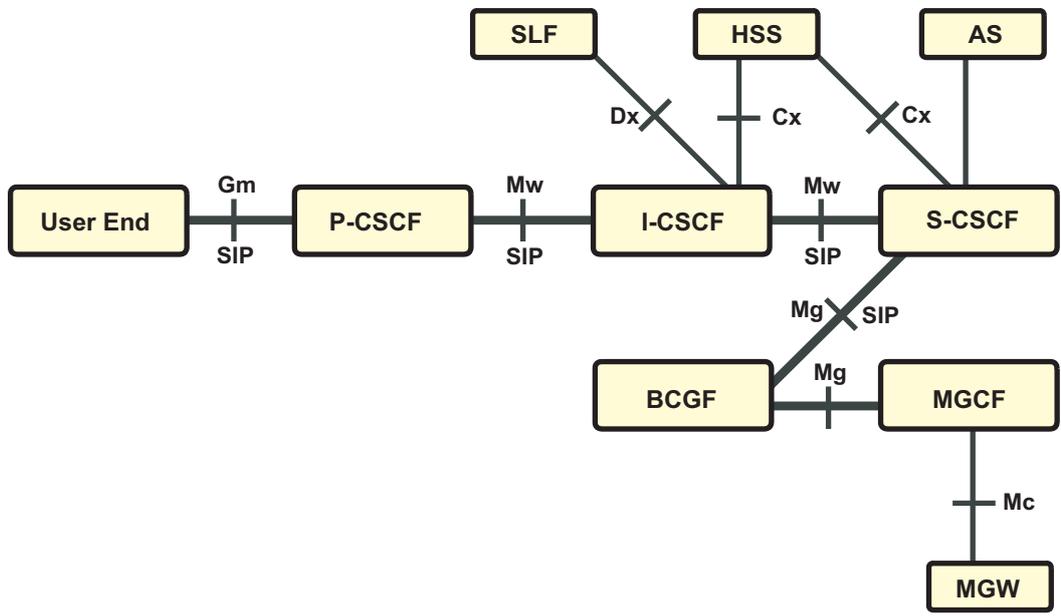
- Gm: P-CSCF – UE
- Mw: P-CSCF – S-CSCF and P-CSCF – I-CSCF
- Mm: S/I-CSCF – external IP networks & other IMS networks
- Mg: S-CSCF – BCGF Mk: BCGF – external IP networks & other IMS networks

Eventually there may be differences in the SIP procedures of Gm and Mw reference points. This implies that there is a difference in UNI and NNI interfaces.

The following procedures have been defined for the 3GPP IM subsystem in:

- Local P-CSCF discovery: Either using **DHCP** (*Dynamic Host Configuration Protocol*) or carrying address in the PDP context
- S-CSCF assignment and cancel
- S-CSCF registration
- S-CSCF re-registration
- S-CSCF de-registration (UE or network initiated)
- Call establishment procedures separated for
 - Mobile origination; roaming, home and PSTN
 - Mobile termination; roaming, home and PSTN
- S-CSCF/MGCF – S-CSCF/MGCF; between and within operators, PSTN in the same and different network
- Routing information interrogation
- Session release
- Session hold and resume
- Anonymous session establishment
- Codec and media flow negotiation (Initial and changes)
- Called ID procedures

- Session redirect
- Session Transfer

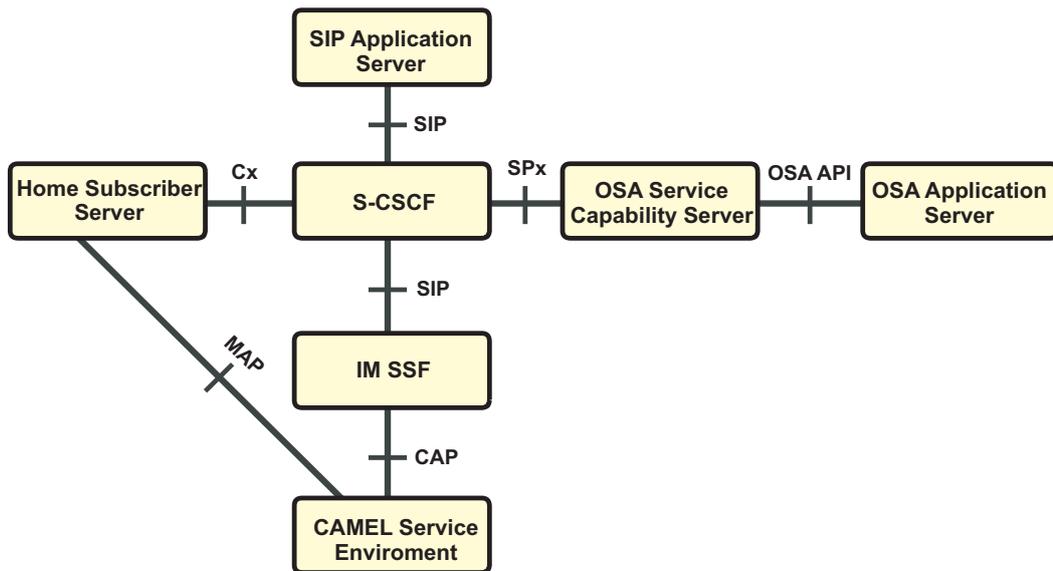


SIP protocol in IMS

1.11 SIP in Service SS



The service subsystem and its connections to IM subsystem is shown in the Figure. The S-CSCF interfaces the application development servers with SIP+ protocols. The SIP application server can reside either outside or within operator's network. The OSA capability server and Camel refer to already standardized 3G and GSM based service generation elements.



Service Subsystem connections with IMS

1.12 TISPAN



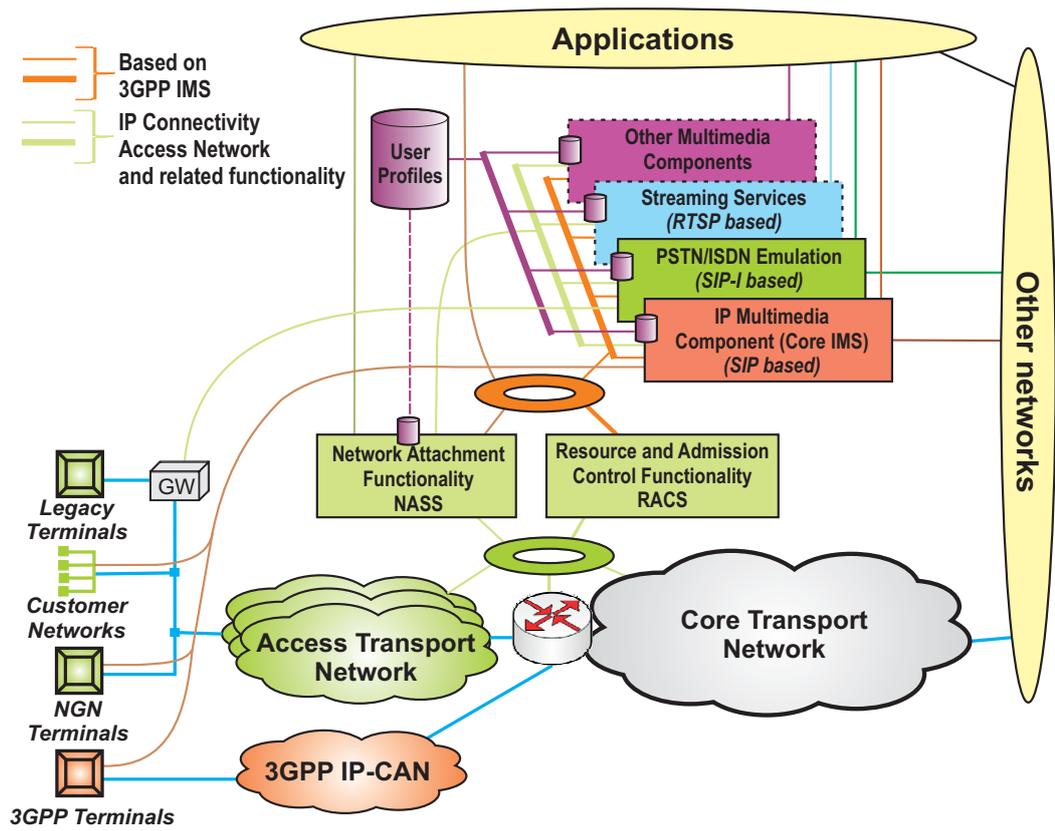
The TISPAN network architecture is based on 3GPP IMS, which is a basis for control and provision of the real-time conversation services (based on SIP protocol) [2], [9], [10]. 3GPP IMS architecture is extended in TISPAN NGN to support various types of access networks, such as xDSL, WLAN, etc.



TISPAN architecture is extended mainly by:

- Access networks control (QoS, access control and authentication),
 - Co-ordination of various control subsystems via one transport network to control resources,
 - Interworking and interoperability with public networks (legacy networks),
 - Separation of the application layer from the connection control layer and the transport layer,
 - Independence of access technologies from the call control layer and the application layer.
-

For services on other than SIP basis, the TISPAN NGN architecture can include other subsystems defined in TISPAN. Figure illustrates the NGN components and functionalities.



Architecture of a TISPAN NGN

1.13 Multimedia Services in NGN Environments

NGN Service Categorization



In parallel with standardization processes in the area of NGN architectures, the working groups of standardization institutions concentrate their work also to standardization activities in the area of new multimedia services and applications, like:

- categorization of NGN services and applications,
 - service development processes, service implementation, service control and provisioning in the NGN environment, etc.
-

ITU-T Service Categorization

In the FGNGN WG1 Services and capabilities document the NGN services and applications categorization structure is introduced [11]. Actual set of services defined by ITU-T is introduced in this document:



1. Interactive-based services

- Real-time Conversational Voice services,
- Point to Point interactive multimedia services, including interactive real-time voice, video, white board and other media,
- Collaborative interactive communication services – multimedia conferences with files and applications sharing, e-learning, games, etc.
- Push to talk over NGN – PoN,
- *Instant messaging (IM)* and Messaging services (SMS, MMS, etc.),
- Group Messaging,
- Existing PSTN/ISDN emulation and simulation services,
- Data communication services (data transmission, fax, e-mail box, etc.),
- Data retrieval applications – telesoftware,
- Online applications (e.g. E-business),
- Voice control services.

2. Non Interactive-based Services

- Content delivery services (audio video streams creation, digital TV distribution services, financial information distribution, distribution of professional and medical images, e-publishing, etc.),
- Sensor Network services,
- Push services,
- Remote control/tele-action services, such as home applications control, telemetry, alarms etc.),
- Broadcast/Multicast Services,
- Over-the-Network Device Management.

3. Both Interactive-based and Non Interactive-based Services

- *Virtual Private Network (VPN)* services,
- Hosted and transit services for enterprises) – IP Centrex, etc.,
- Information services (e.g. traffic services-situation on the road, train/buss tickets, advanced push services, etc.),
- Presence and general notification services,
- 3GPP Release and 6/3GPP2 Release A OSA-based services.

4. Network Services

- *Basic Transport Service (BTS)*,
- *Enhanced Transport Service (ETS)*.

5. Regulated Services

- Emergency Telecommunication Services – citizen-department, department-department, department-citizen,
 - Lawful Intercept Services,
 - Broadcast Emergency Alerting Services.
-

ETSI Service Categorization



1. IP multimedia services

- IP multimedia applications,
- PSTN/ISDN simulation services – 3rd class,
- Instant messaging service,
- Presence service,
- Location service,
- Video Telephony service.

2. PSTN/ISDN Emulation services

3. Regulated services for both IP multimedia and PSTN/ISDN emulation

- Lawful Intercept Services,
 - Emergency Call Services,
 - Malicious Communication Identity Services,
 - Anonymous Communication Rejection Services.
-

1.14 NGN Service Capabilities

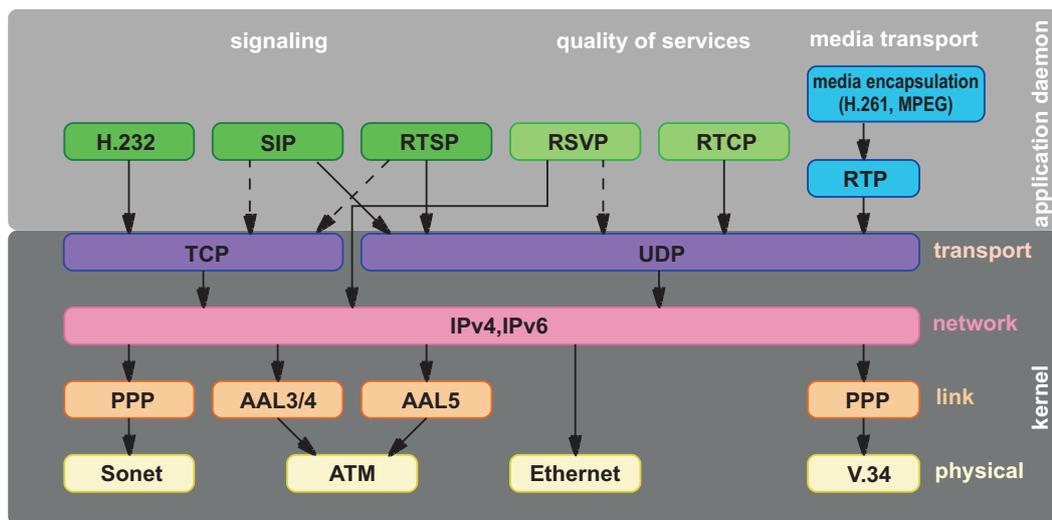
NGN platform should be able to provide the capabilities (infrastructure, protocols, etc.), so that it should be possible to develop, deploy/implement and manage all types of services (known and expected) [1], [12]. This involve the possibility to apply different types of media (audio, video, text, data), different types of encoding techniques, data services, conversation services, user and group transmission services, messaging services, real-time and Non-real time Conversational Voice services, data communication services, delay sensitive services, delay non sensitive services, etc. These types of services required the different speed of communication connection (from some kbps till hundreds of Mbps), which are required for the given service. These requirements have to be supported by the capabilities of the transport technologies.

-
- + One of the expected advantages of the NGN platform is the user comfortable and flexible access and control of multimedia services. At some time the NGN should provide the effective interface for service development, service providing and service management.
-

1.15 NGN Protocols

NGN protocol stack

The best way to show the functions of individual protocols in the hierarchy of NGN protocols platforms supporting voice transport over the packet networks or controlling of elements in NGN architecture is shown on Figure below with depicting the individual protocols and the OSI reference model layers they belong to.



Protocols for NGN

The protocols for the converged technologies and NGN platform can be divided into the following groups [4], [10], [12], [13], [14]:

- call control protocols (VoP signaling from the telecommunication point of view): SIP/SDP, H.323 [10], [12], [13],
- media gateway control protocols (components of the distributed VoP architecture): MGCP, Megaco/H.248 (protocol approved by both IETF and ITU-T),
- protocols for signaling transport: SIGTRAN, BICC, SIP-T, SIP-I,
- transport protocols: RTP, RTCP (in the sense of media transfer not RM OSI, as otherwise TCP/IP or UDP/IP is used for all),
- protocols for QoS support: RSVP, RTCP (RTCP is a transport one, but allows QoS support as well).

Other support protocols:

- DHCP, ENUM, DSN, COPS,

- **RTSP** (*Real-Time Streaming Protocol*) – protocol for creation of streams in the real time,
- IGMP/MLD.

1.16 Fundamental NGN Protocols

SIP protocol

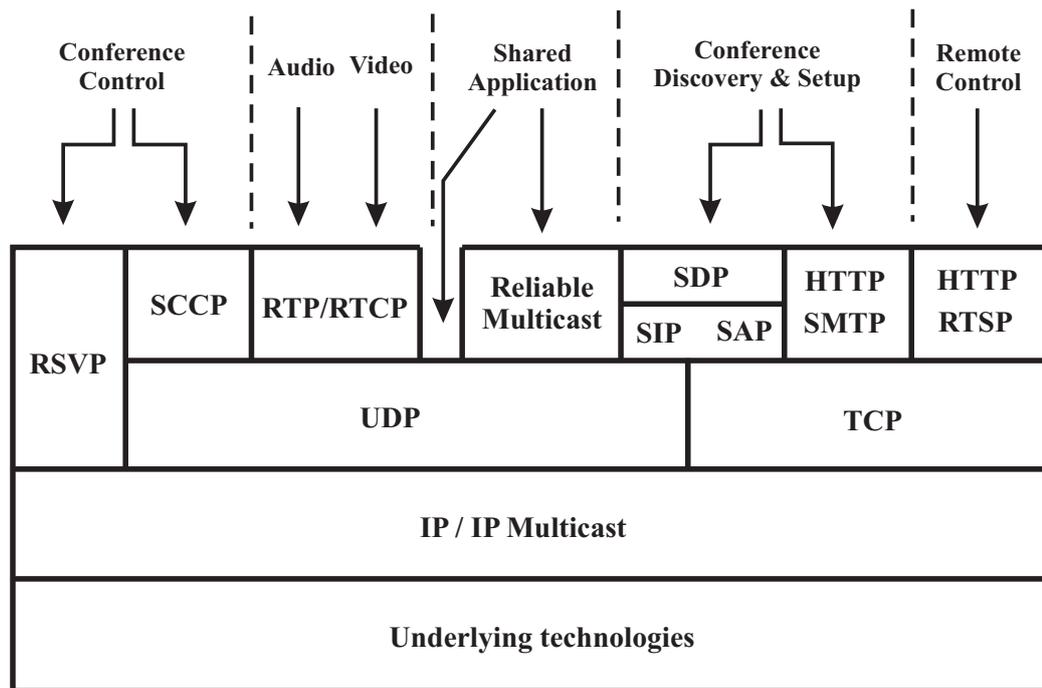


Session Initiation Protocol (SIP) is an application-layer control protocol that handles the setup, modification, and tear-down of multimedia sessions. Media can be added to (and removed from) an existing session. SIP is used in combination with other protocols to describe the session characteristics to potential session participants. SIP is based on a request and response transaction model similar to HTTP. Each transaction consists of a request that invokes a particular method or a function on the server and at least one response.

SIP supports five facets of establishing and terminating multimedia communications:

- User location: determination of the end system to be used for communication;
 - User availability: determination of the willingness of the called party to engage in communications;
 - User capabilities: determination of the media and media parameters to be used;
 - Session setup: "ringing", establishment of session parameters at both called and calling party;
 - Session management: including transfer and termination of sessions, modifying session parameters, and invoking services.
-

SIP is a text-based protocol suggested and standardized in RFC 3261. SIP has been proposed as a part of a unit based of following protocols.



SIP protocol stack

Consequential protocols



A lot of SIP functions depend from other protocols. SIP defines establishment, termination and call modification and SIP use other protocols as *Real-time Transport Protocol (RTP)* for transporting real-time data and providing QoS feedback, the *Real-Time Streaming Protocol (RTSP)* for controlling delivery of streaming media, the *Media Gateway Control Protocol (MEGACO)* for controlling gateways to the *Public Switched Telephone Network (PSTN)*, and the *Session Description Protocol (SDP)* for describing multimedia sessions. Therefore, SIP should be used in conjunction with other protocols in order to provide complete services to the users. However, the basic functionality and operation of SIP does not depend on any of these protocols.

SDP



SDP (*Session Description Protocol*) [4] is intended for describing multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation.

When initiating multimedia teleconferences, voice-over-IP calls, streaming of video, or other sessions, there is a requirement to convey media details, transport addresses, and other session description meta data to the participants. SDP provides a standard representation for such information, irrespective of how that information is transported. SDP is intended to be general purpose so that it can be used in a wide range of network environments and applications. However, it is

not intended to support negotiation of session content or media encodings: this is viewed as outside the scope of session description.



An SDP session description includes the following:

- Session name and purpose
- Time(s) the session is active
- The media comprising the session
- Information needed to receive those media (addresses, ports, formats, etc.)

As resources necessary to participate in a session may be limited, some additional information may also be desirable:

- Information about the bandwidth to be used by the session,
 - Contact information for the person responsible for the session.
-

RTP

The goal of this part is to present **RTP** (*Real Time Transport Protocol*) [14] in the way that it will be easily understandable also by a beginner. It contains deeper description of the main RTP protocol, but also protocols that stand next to RTP, cooperate with it and fill its gaps. The structure of RTP will be shown, to introduce the protocol body to the reader with a close view of its parts and their functions.



RTP provides end-to-end delivery services suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. Real-time means that not only correct results are required, but also a sufficient time in which the result is delivered. That is why the delivery of the audio or video data is typically delay sensitive. According to this RTP use timestamp and control mechanisms for synchronizing different streams with timing properties.

RTCP



RTCP (*Real-time Transport Control Protocol*) [4] is an application layer protocol designed to control of data delivery in real-time and to measure the QoS. It is defined in RFC 3550 published in July 2003. RTP protocol uses the RTCP protocol, which transports the following additional information for the management of the session. RTCP is based on the periodic transmission of control packets to all participants in the session. The underlying protocol must provide multiplexing of the data and control packets, like UDP protocol that allows the multiplexing of RTP data packets and RTCP control packets. RTCP protocol requires the sending of information periodically by the participants of the session.

RTP packets only transport user's data, whereas RTCP packets only transport in real time the supervision.



Protocol RTCP performs these principal functions:

- Provide the information about the quality of the session (QOS) by means of feedback, which include the number of lost packets, the time return ticket and the gigue.
- Keep a trace of all the participants by a persistent transport-level identifier called CNAME (Canonical Name). Because SSRC (Synchronization Source Identifier) may change if a conflict or program restart occurs.

Control the media flow and adapt it to all the participants of the RTP session. By having each participant send its control packets to all the others, each can independently observe the number of participants. This information is used to calculate the rate at which the packets are sent.

DIAMETER



DIAMETER [4] is a member of "AAA" protocols collection, derived from its predecessor RADIUS protocol. It is a peer to peer protocol , used for handling service requests such as user validation, network resource control, connection and session management, wireless or roaming charging, billing applications etc.



Diameter sessions consist of exchange of commands and AVPs between servers and clients and unlike Radius, uses peer to peer architecture rather than more classic client/server scheme. Each node may initiate a message (request) at any time, as example, server may abort a service to specific user. Diameter is defined in terms of base protocol and a set of applications. This design allows protocol to be extended for new access technologies. The base protocol provides basic mechanism for reliable transport, delivery and error handling.

MEGACO/H.248



This protocol has been established to cover the need of IP networks and services to interoperate with traditional networks (e.g. PSTN) and provide the same services over both types of networks (IP, Traditional). This enables separation of call control from media conversion. Megaco/H.248 is defined as master / slave architecture based protocol which is used for communication between **MGC** (*Media Gateway Controller*, sometimes called a call agent or softswitch, which dictates the service logic of that traffic) and one or more decomposed **MGs** (*Media Gateways*), which converts circuit-switched voice to packet-based traffic.



Megaco/H.248 instructs an MG to connect streams coming from outside a packet or cell data network onto a packet or cell stream such as RTP. Megaco/H.248 is similar to MGCP from an architectural standpoint and the controller-to-gateway relationship, but Megaco/H.248 supports a broader range of networks, such as ATM.

SIGTRAN

In view of functionality and performance the user make high demands on modern telecommunication networks. Using **IP** (*Internet Protocol*) signaling messages will be transmitted over **TCP** (*Transmission Control Protocol*) or **UDP** (*User Datagram Protocol*). These transport protocols are not designed to meet the requirements given by a signaling system used in a circuit switched network like **PSTN/ISDN** (*Public Switched Telephone Network/Integrated Services Digital Network*). So the working group SIGTRAN was founded by the **IETF** (*Internet Engineering Task Force*) to develop a new protocol, based on IP, in consideration of given requirements by the existing switched telephone network.



This protocol, named **SCTP** (*Stream Control Transmission Protocol*) has some advantages in comparison to TCP. The SCTP offers a fundament to initiate and run secured transport connections using IP networks to transmit signaling information. Based on SCTP, several adaptation layers enable the transmission of upper layer protocols, i.e. **ISUP** (*ISDN User Part*), **SCCP** (*Signaling Connection Control Part*) and **DSS1** (*Digital Subscriber System No. 1*).

1.17 Supporting NGN Protocols

DHCP

DHCP (*Dynamic Host Configuration Protocol*) evolved from the BOOT protocol (BOOTP). Both protocols are described in RFC 2131 (DHCP) and RFC 951 (BOOTP).



DHCP includes all features known from BOOTP, that means an *Internet Software Consortium (ISC)* DHCP includes a BOOTP server and additional features along with a dynamic address assignment. Both protocols act for IP address assignment to nodes.

Therefore, the un-configured IP node sends a request for an IP address to a DHCP server. Then this DHCP server assigns an IP address to the client. Furthermore, this answer includes e.g. domain-name, IP address of the name-server or IP address from a router. The transmission of all configuration parameters will be proceeded automatically, depending of the chosen method.

DNS



The **DNS** (*Domain Name System*) protocol is used to link IP addresses with domain names. Usually, it is more convenient for people to remember names (ngnlab.eu) than IP addresses (147.175.103.213). IP addresses are required by the third layer of the network model to deliver the application data through networks. This chapter will highlight SIP/IMS specific use of DNS and not introduce the protocol itself.

Besides only storing a mapping between IP address and domain name, DNS contains additional information using various record types. DNS can handle for example certificate records, location information records, service information records and much more.

HTTP



The *Hypertext Transfer Protocol (HTTP)* is application protocol using the request/response mechanisms and is one of the most used protocols on Internet for web services.

A client sends a request to the server in the form of a request method, URI, and protocol version, followed by a **MIME**-like message (*Multipurpose Internet Mail Extensions*) containing request parameters and body content over a TCP connection with a server (HTTP session). Server is replay with response containing status line including the message's protocol version and a success or error code, followed by a MIME-like message containing server information, metadata and body content.

XML



In the following sections, **XML** (*Extensible Markup Language*) and its concept will be introduced. The protocol has been standardized by the **W3C** (*World Wide Web Consortium*). This section will not go into the very details of the protocol itself. It will rather provide the basics to understand the protocols which are based on XML. XML is a main mechanism for representing structured data. The data in XML documents is represented by a tree with nested elements. Each node in the tree represents an element.

Elements can have attributes, but they are not mandatory. Furthermore, so called “leaf” elements can contain text content. XML documents require a declaration with version and encoding mandatory at the beginning. After this declaration, the elements and the XML encapsulated data itself follow. The elements, which can be used, are defined by the XML schema or **DTD** (*Document Type Definition*). As different and more definitions can be used within one document, XML is extensible.

XCAP



XCAP (*XML Configuration Access Protocol*) allows clients to read, write and modify data stored in XML format on a server. This can be done by mapping XML document sub-trees and element attributes to HTTP URIs which grants direct access, as for the reason that all content discussed so far is held in XML “containers”. The XML files are stored on a so called **XDMS** (*XML Document Management Server*), which is usually a normal HTTP server. The standard describes the interface between client application and the server managing the XML data (e.g. presence resource lists or authorization data for presence management).

SOAP



SOAP (*Simple Object Access Protocol*) is also an application layer protocol. The protocol is used for the communication between applications over the internet. SOAP uses HTTP as lower transport layer protocol.



The advantage using HTTP is its support by many applications (browsers, servers, mobile phones) and its easy and cheap implementation. Other protocols for remote communication do not join this advantage.

CORBA



CORBA (*Common Object Request Broker Architecture*) is also a standard that defines a protocol for remote procedure communication. It is defined by the **OMG** (*Object Management Group*). The core of CORBA is the so called **ORB** (*Object Request Broker*). The ORB is the middle-ware that describes the client/server relationship for the communication. The ORB is responsible to:

- Intercept a call from the client
 - Find the correct object
 - Pass it the parameters
 - Invoke its method
 - Return the results to the client
-

Thus, the process seems transparent to the client. It uses only the communication via CORBA. The realization of the actual distributed application does not need to be specified any further. The only standard required for the actual communication is the communication standard.

VoiceXML



VoiceXML is designed for creating audio dialogs that feature synthesized speech, digitized audio, recognition of spoken and **DTMF** (*Dual-Tone Multi-Frequency*) key input, recording of spoken input, telephony, and mixed initiative conversations. Its major goal is to bring the advantages of Web-based development and content delivery to interactive voice response applications.

The top-level element in the XML description file is the `<vxml>` tag. It can contain two types of dialogues:

- Forms – To present information and gather input
- Menus – To choose the next step

1.18 Multimedia Services Control Protocol



MPEG is a standard for "the generic coding of moving pictures and associated audio information. The function of MPEG is to take analogue or digital video signals and convert them into packets of digital information that are more efficiently transported on modern networks. This process of converting audio and video signal to digital form is called compression (or coding). Opposite process is called decompression (or decoding) when digital audio and video signal is converted to analog form. MPEG is a system for compression and encoding of digital multimedia content. MPEG standard compresses the video and audio into much less information as it needed before, consuming less transmission bandwidth. Level of compression depends on bandwidth requirements and also on level of quality.

MPEG 1



The default size for an MPEG-1 video is 352x240 at 30 fps for NTSC (352x288 at 25 fps for PAL sources). These were designed to give the correct 4:3 aspect ratio when displayed on the rectangular pixels of TV screens. For a computer-based viewing audience, 320x240 square pixels give the same aspect ratio. MPEG-1 delivers roughly VHS quality at 30 frames per second at 1.5 Mbps. It can be scaled up or down in size or bit rate, but range between 1.2 – 1.5 Mbps is the optimal bit rate.

MPEG 2



MPEG-2 needs about 6 Mbps to provide the quality movie on DVDs, although data rates up to 15 Mbps are supported. 720x480 is the typical 4:3 default resolution, while 1920x1080 provides support for 16:9 high-definition television. MPEG-2 is now arguably the most successful new consumer standard ever relative to its acceptance in the marketplace. Presently it is the predominant standard for existing digital video equipment worldwide. Based on MPEG-2, digital television (both *standard definition*, **SD**, and *high definition*, **HD**) has become the norm for broadcasting, replacing analog broadcast in all but standard terrestrial and cable television.

MPEG 4



MPEG-4 and H.264 have a common heritage within the ISO and ITU standards committees. As a result the overall coding approach is quite similar. Both algorithms are based on a common heritage of DCT based, hybrid image coding, first used in H.261 and MPEG-1. A number of comparison tests have been performed between H.264 and MPEG-2/MPEG-4 on standard MPEG test material. For standard resolution (704x480, 60 Hz interlaced) video sequences to

achieve a PSNR level of 28, NBA must be coded at a rate of 5 Mbps using MPEG-2, but only 1.8 Mbps using H.264.

MPEG Audio Compression



MP3 is actually part of the MPEG-1 standard. The audio portion of the MPEG-1 specification contains three different compression schemes called layers. Of the three, Layer 3 provides the greatest audio quality and the greatest compression. At 8 kbps, MP3 will sound like a phone call – intelligible, but nothing that would ever be called high-fidelity. Good-quality music starts at about 96 kbps, but generally 128 or 160 kbps to would be closer to "CD quality".

RTSP



RTSP (*Real-Time Streaming Protocol*) is an application-level protocol for control over the delivery of data with real-time properties and its goal is streaming of multimedia over multicast and unicast in "one to many" applications. RTSP provides an extensible framework to enable controlled, on-demand delivery of real-time data, such as audio and video.

Sources of data can include both live data feeds (e.g. Live TV channels) and stored clips (e.g. Video On Demand). RTSP establishes and controls single or several data delivery time synchronized media sessions, provide a means for choosing delivery channels such as UDP, multicast UDP and TCP, and provide a means for choosing delivery mechanisms based upon RTP and control mechanism of streams upon RTCP. RTSP is not tied to RTP and RTCP. There is no notion of an RTSP connection – instead, a server maintains a session labeled by an identifier. An RTSP session is in no way tied to a transport level connection such as a TCP connection.

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During an RTSP session, an RTSP client may open and close many reliable transport connections to the server to issue RTSP requests. Alternatively, it may use a connectionless transport protocol such as UDP. The streams controlled by RTSP may use RTP, but the operation of RTSP does not depend on the transport mechanism used to carry continuous media. The protocol is intentionally similar in syntax and operation to HTTP/1.1 so that extension mechanisms to http can in most cases also be added to RTSP.

IGMP



Internet Group Management Protocol (**IGMP**), allows Internet hosts to participate in multicasting. IGMP allows users to announce their intention to join particular multicast groups. These groups are identified by their unique Class-D IP addresses.

When a workstation wants to participate in a multicast group, it sends an IGMP “join” message to its local router. If multiple routers exist on a single segment, they can mutually elect a “*Designated Router (DR)*” to manage all of the IGMP messages for that segment. After a router receives one or more “joins” for a specific group, the router will forward any packets destined for that group to the appropriate interface. The router should only forward one copy of the data packet per interface.

If multiple receivers exist on a single interface they will all receive the same information by monitoring common multicast MAC and IP addresses. If the multicast group has receivers spread over several router interfaces, the router must replicate the packet and deliver a copy to each interface that contains registered users. This type of transmission activity can be immense, which is why IGMP is highly useful as a stateful protocol. The designated router regularly verifies that the attached workstations want to continue to participate in their respective multicast groups. The designated router sends periodic “queries” to the receivers. These queries are transmitted to a well-known multicast address (224.0.0.1) that is monitored by all systems. If the receivers are still interested in that particular multicast group, they will respond with a “membership report” message. When the router stops seeing responses to queries, it will delete the appropriate group from its forwarding table.

MLDv2



MLD (*Multicast Listener Discovery*) is de facto derivative of IGMP used in IPv6 networks. It is control layer protocol used to control multicast stream flow in IPv6 network. MLD and MLDv2 protocols are designed for use only in IPv6 network. This protocol is follower of IGMP in IPv4 networks used for joining some network node to multicast group. Multicast groups are separated by IP addresses. Procedure of joining some multicast group on network is to announce next routing point, which must be directly attached to some of the requester interfaces.

Node asks it need packets provided for some particular group. If router point have not access to some multicast group data, it must also join it on other interface(s) and forward this data to requester. Router point is sending request to next router point. This process must recursively continue to router point, where source of multicast group data is connected. Note that a multicast router may itself be a listener of one or more multicast addresses; in this case it performs both the "multicast router part" and the "multicast address listener part" of the protocol, to collect the multicast listener information needed by its multicast routing protocol on the one hand, and to inform itself and other neighboring multicast routers of its listening state on the other hand.

2 Digital Video Broadcasting Technology

2.1 Introduction

Television (TV) has undergone a lot of important milestones throughout the years of its evolution starting with a primitive mechanical television (1884) through electronic analog (black and white, color) televisions to digital television (in standard and high definition resolutions). Digital television provides a new way of video distribution and broadcasting. It is a new media that offers a lot of innovations with a new operation model. The advent of digital television significantly contributes to a convergence of computers, television and Internet. Benefits for customers are noticeable: a treat from a picture in high definition resolutions, audio in **CD** (*Compact Disc*) quality, hundreds of TV channels and plumbless access to a wide range of new services. These digital technologies allow various companies, operators, providers and distributors to offer a variety of useful and profitable services such as a high data rate Internet access, offline as well as online games, video on demand, video and audio (songs) streaming, electronic newspapers and others.

+

Digital television utilizes a big advantage it offers a high speed data transmission enabling it to provide a rich multimedia content. In comparison to analog television one analog TV channel can carry a group of digital TV channels including radio and data channels. How is it possible? Thanks to digital video compression techniques and modulations.

In order to allow digital television come into being cooperation among a several companies was required. Some of the best-known companies are: the *European Telecommunications Standards Institute (ETSI)*, *Digital Video Broadcasting (DVB)* consortium (project), the *Advanced Television Systems Committee (ATSC)* and other.



Digital video broadcasting project originated in years 1991 to 1993 and grouped about 80 members. Currently, about 300 organizations and companies form this project (consortium) in more than 30 countries. Project member portfolio covers electronic device producers, network operators, broadcasters, software companies and regulatory bodies [18].

2.2 DVB Services

From the end user point of view every new technology is interesting when it offers a lot of new more quality services in comparison with existing technologies. DVB technology is not more the passive medium. It enhances an analog TV technology by a picture quality, data services and interactivity.



The DVB technology provides three basic (core) services:

- video broadcasting (television in standard or high definition resolutions),
 - audio broadcasting,
 - data broadcasting.
-

However, except a TV program distribution the DVB technology also allows to distribute to users varied data services. These services can be divided into two groups: interactive and pseudo-interactive. In case of pseudo-interactive (or one-way interactive) services data for such services are transmitted via transmission channels along with TV channels and after decoding they are stored in a memory of the end user device (set-top-box). The user has options to select and browse information in a memory. In case of interactive (fully or two way interactive) services a return channel is used by the user to choose and control data services provided by operator. The return channel is realized by a separate transmission medium.

Among classic pseudo-interactive services we can include:

- electronic news (news agencies, text and graphical information),
- webcasting (data service providing information from selected Internet sites; usually public interest information),
- weather forecasts,
- information from reservation systems (hotels, time-tables, city public transport, etc.) without options of an active access,
- information from betting systems,
- software and games distribution,
- exchange information systems (stock, commodity and option exchanges, financial inter-bank markets, etc.),
- auction systems,
- advertisement information,
- distance learning and trainings.

The interactive services cover:

- Internet access,
- interactive distance learning,
- electronic business, interactive advertisements,
- reservation systems with active access (control),
- video-services (video on demand, video rental services),
- games,
- electronic banking,
- betting, quizzes, contests, voting.

2.3 DVB Standards



DVB represents a set of open standards maintained by DVB Project (consortium) covering broadcasting of digital video or in general digital TV.

These standards are published by a *Joint Technical Committee (JTC)* of standardization organizations ETSI, **CENELEC** (*European Committee for Electrotechnical Standardization*) and **EBU** (*European Broadcasting Union*). Therefore, they are internationally accepted. DVB was first adopted in Europe (United Kingdom).



Currently, except Europe DVB is also used in Australia, Asian, African and American countries. In addition to DVB, there are other standards for digital television, such as **ATSC** (*Advanced Television Systems Committee*) employed e.g. in Canada, US or Mexico or **ISDB** (*Integrated Services Digital Broadcasting*) implemented in Japan [18].

DVB standards cover all aspects related to broadcasting and processing of digital video and audio at a physical and data link layer of a communication model. In details, these standards define modulations and forward error coding, multiplexing of several services into one transport stream, interfaces, etc. A lot of these aspects closely relate to transmission media used for broadcasting (terrestrial, satellite or cable). Some of standards that can be found within DVB are as follows [19]:

- DVB-S – broadcasting digital TV via satellite (S)
- DVB-S2 (satellite 2nd generation) – broadcasting high definition TV via satellite
- DVB-SH – broadcasting IP based media to handhelds (mobile, PDA) via satellite
- DVB-T – broadcasting digital TV via terrestrial (T) environment
- DVB-T2 (terrestrial 2nd generation) – broadcasting high definition TV via terrestrial
- DVB-C – broadcasting digital TV via cable (C) systems
- DVB-RCS/RCT/RCC – return (interaction) channel via satellite/terrestrial/cable
- DVB-H – broadcasting digital TV to handhelds (H) via terrestrial
- DVB-MC/MS – broadcasting digital TV via microwave systems
- DVB-Data – transmission of high speed data services
- DVB-SI – defines a *service information (SI)*, i.e. data structures (so called metadata)

- DVB-CSA – defines a *common scrambling algorithm (CSA)*
- DVB-CI – defines a *common interface (CI)* between a removable conditional access module and receiver
- DVB-NIP – defines *Network Independent Protocols (NIP)* to support interactive services
- DVB-MHP – definition of a java-based *Multimedia Home Platform (MHP)* for development of end user applications

There are also other DVB standards not mentioned in the list above covering a subtitling, measurement, multiplexing, 3D-TV, IPTV, source coding, etc. but most of them are out of scope of this chapter.

2.4 DVB System

DVB-S, DVB-T and DVB-C are best known as well as most used technologies for accessing the digital content. As was already mentioned, those standards define the physical and data link layer of an entire distribution system. All multimedia content is grouped and transmitted through MPEG *transport streams* (TS).



If we talk about the multimedia content namely its video and audio content it is good to realize that source video and audio signals are analog signals and they have to be converted into a digital form (analog-to-digital converter) to utilize benefits of a digitization. However, the analog video signal that needs a bandwidth of 5 MHz in case of a standard European 625-line TV signal with 720 pixels per line amounts to 414,720 (576 x 720) pixels per picture (frame). After digitization a black and white video signal (with 25 pictures per second) would require a rate of about 83 Mbps (or about 250 Mbps for color video). Those bit rates are too high and almost inapplicable in real communications (e.g. over satellite). Fortunately, video signals as well as audio signals contain a lot of redundant information that can be removed via suitable compression technique.

Using the compression the original rate can be decreased (based on quality and resolutions) to several Mbit/s. For this purpose the *Moving Pictures Experts Group* (MPEG) was formed with a task to develop efficient compression techniques for a work with moving clips in computers and their transport between computers or other devices. DVB technology adopted an MPEG-2 compression standard [20]. It supports several video qualities and resolutions as well as it provides high flexibility.



As was already said above DVB is set of standards covering not only video/audio compressing but all functions of an entire DVB system for digital video delivery to end users or other providers. Such DVB system has to multiplex all input streams (video, audio, data signals) into one final transport stream and send it via given transmission medium in a proper form. Next parts of this chapter will deal more with this stream processing.

2.5 DVB System - MPEG-2 systems layer

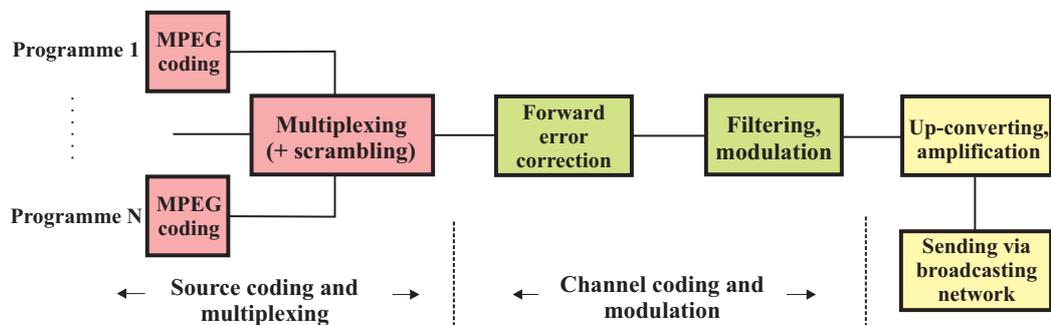


MPEG-2 systems layer defines how various elementary streams representing one or multiple programmes are multiplexed together. The elementary streams can carry video, audio, data, and other information. This multiplexing process creates a single (multi-programme transport) data stream that can be stored or transmitted via a physical medium. In general, the MPEG-2 systems layer performs more functions:

- multiplexing,
- packetization,
- timing and synchronization,
- conditional access.

First two functions are described below the characterization of other two can be found in [22]. In this chapter we use a term “programme”. It has a lot of meanings but we will think of it as a single broadcasting service or channel.

Figure below shows a block diagram illustrating all main operations that have to be done at a transmitter side to broadcast a digital content to users [20]. At first all programmes have to be encoded and multiplexed. The resulting transport stream is equipped by error protecting codes and modulated to a carrier. At a last phase the signal is amplified and sent to the transmission medium.



Main operations performed at transmitter side

2.6 DVB System - Elementary streams



The elementary streams can carry the MPEG-2 compressed video and audio, data, timing and system information, conditional access information and other programme related data. They represent components of the programme.



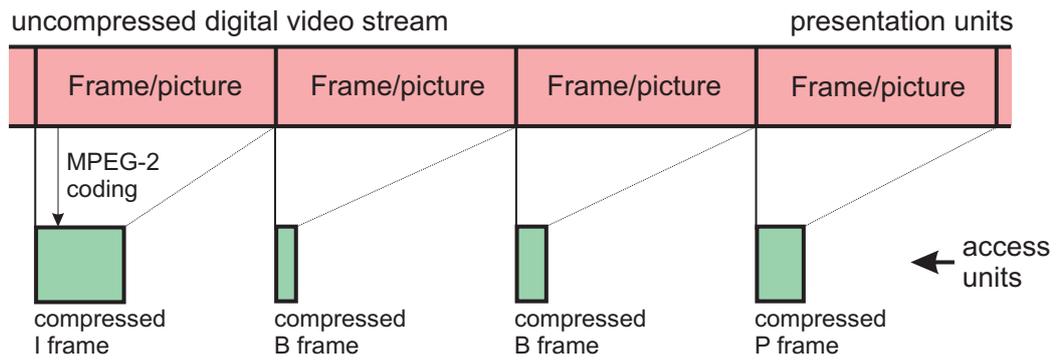
The simplest type of a programme is a radio service that consists of a single elementary audio stream. On the other hand a classical television service consists of three elementary streams: one stream carries coded video, second stream carries coded stereo audio and third contains teletext.

However, there is no problem to offer a television service containing one stream with video in the standard definition, one stream with high definition video, a several audio streams in different languages and even more streams for teletext in different languages [21].

Let's consider an uncompressed digital video stream that consists of a sequence of frames. Each frame (e.g. 830 kB for 625 lines) representing an uncompressed video picture is called a presentation unit. MPEG-2 coder encodes and compresses every presentation unit making an access unit. The access units as can be seen in Figure below are not of the same size. Their size depends on original picture complexity and a type of each frame whether it is an I, P or B frame [20]:

- I (Intra) frames/pictures are coded in similar way like JPEG pictures without any reference to other video pictures. They contain all information needed to reconstruct original pictures.
- P (Predicted) picture is coded in reference to a preceding (I or P) picture. This picture only carries information about a change (motion) between preceding and actual picture.
- B (Bi-directional) picture is similar to the P picture but it is also coded in reference to a picture which follows.

An output of the MPEG-2 coder is a sequence of the access units and this sequence constitutes the elementary video stream. In the similar way an uncompressed audio stream of audio presentation units is encoded by the MPEG coder to a sequence of audio access units forming so the audio elementary stream.

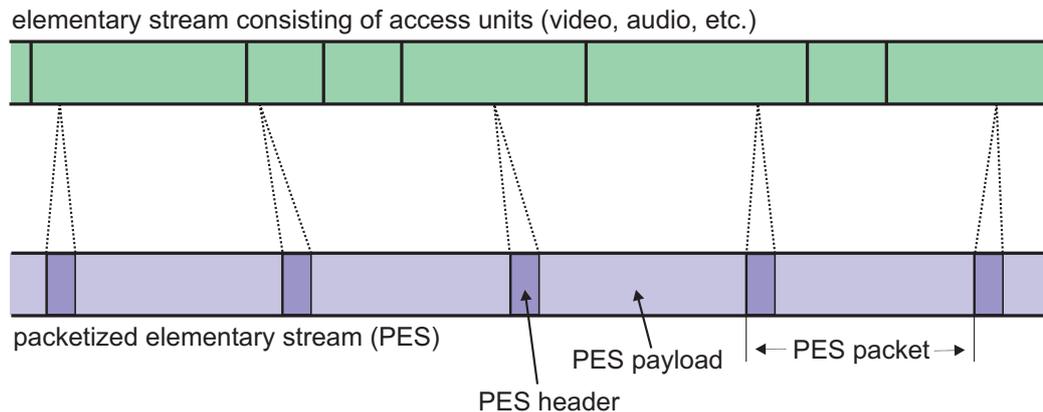


Principle of video sequence coding

2.7 DVB System - Packetized elementary stream



The MPEG-2 multiplexer does not directly multiplex sequences of the access units from its inputs. All elementary streams consisting of the access units are transformed into so called *packetized elementary streams (PES)*. Each PES consists of PES packets as can be seen in next Figure [21].

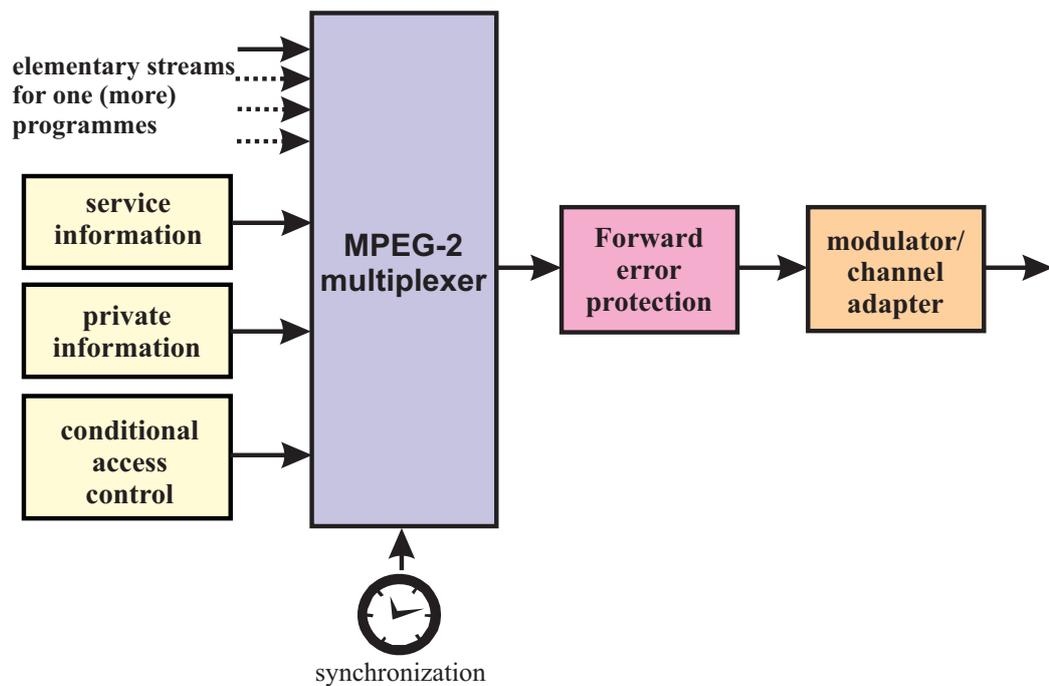


Principle of forming the packetized elementary stream

Every PES packet contains a header and a payload. The payload is a field where data of the original elementary stream are grouped in one after another. There is no limitation on synchronization of PES starts and access unit starts. That means a starting byte of the access unit can occupy any place (byte) of the PES packet payload. Several small-sized access units can even be put in the payload of one PES packet. PES packets can have a variable length but up to 64 kB in size. There is certain freedom for designers of the MPEG-2 multiplexer to utilize this flexibility. They can decide to apply PES packets of a fixed size or variable size to ensure that first byte of any PES packet payload will carry first byte of some access unit. Every PES packet header contains a start code (to identify a start of the PES packet), an identifier of stream (within programme), a length of the PES packet and its header, optional header subfields and flags (indicators of subfields).

2.8 DVB System - Stream multiplexing

When the elementary streams are in a form of packetized elementary streams they are multiplexed (grouped) by the MPEG-2 multiplexer with all other special information to form a resulting contiguous byte data stream. Figure below depicts a principle of multiplexing in the MPEG-2 systems layer.



The principle of multiplexing



As can be seen from Figure an output multiplexed signal contains a lot of other information [21]:

- Time stamps – time stamps allow to ensure a synchronization among elementary streams (e.g. video and audio streams of one programme have to be synchronized).
- Tables – tables contain service information (SI) data which identify programmes within a multiplexed signal (multiplex) as well as their elementary streams, network parameters, etc. There are also tables for a conditional access related to scrambling but this type of control was not defined by MPEG (or DVB).
- Other support data – support data for additional data streams whose content is not specified by MPEG. These data streams can carry contents of various data services, other system information (e.g. modulation) or e.g. teletext.

The MPEG-2 multiplexer can produce two types of multiplex streams: a programme or transport stream. The *programme stream (PS)* is intended for the

storage and retrieval purposes of digital content from storage medium (e.g. DVD) and it relies on error-free environments. Unlike the programme stream the transport stream (TS) enables to multiplex more programmes and is not so much susceptible to errors because it is protected by **FEC** (*Forward Error Correction*) code. Therefore, TS is suitable for broadcasting via terrestrial or satellite environments. The other difference is that TS consists of transport packets with fixed length of 188 bytes.

2.9 Digital Video Broadcasting via Satellite

- + Satellite systems provide the operators with a lot of benefits in offering services to end users. A natural ability of satellites to distribute signals to large areas of the Earth surface has been utilized for broadcasting of analog television and radio for decades. This ability mainly relates to geostationary satellites that are placed in a geostationary orbit i.e. in latitude about 36000 km (over the equator).
-

Each geostationary satellite appears for Earth user fixed in the sky so there is no need for an antenna tracking system.

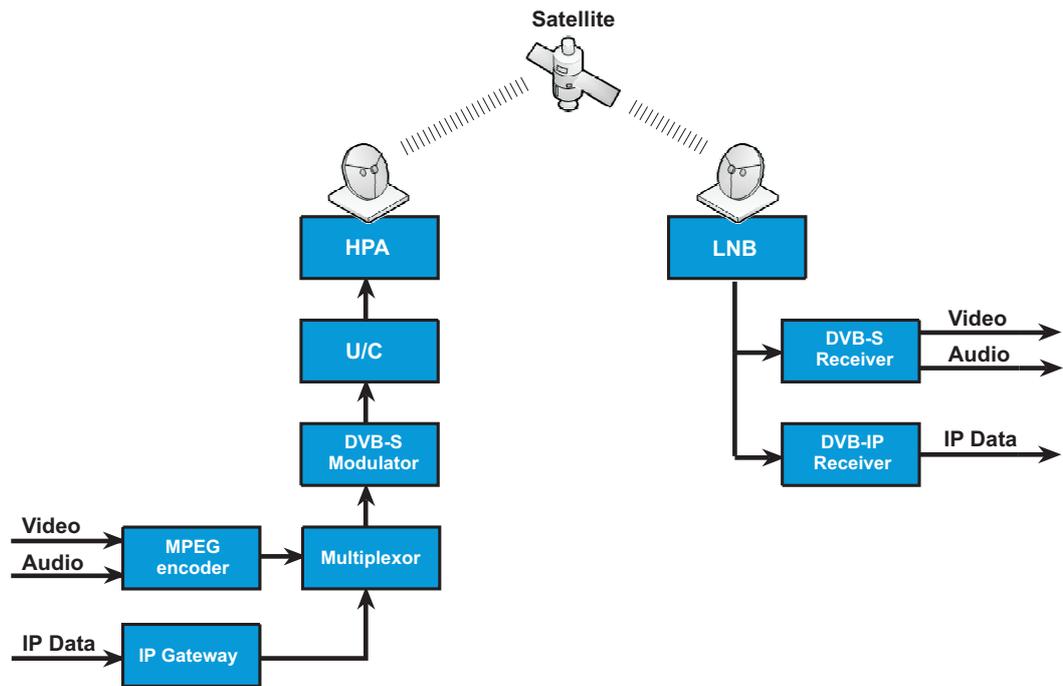
- On the other hand satellite transmissions suffer from error prone satellite links therefore every signal before transmitting has to be adapted for such difficult propagation conditions [20].
-

A communication payload of satellites consists of transponders. Their function is to receive, restore, amplify, process, re-modulate and sent signal back to Earth.



Currently, the conventional geostationary satellite contains about 20 to 30 transponders and a single transponder can most often have a bandwidth ranging from 26 to 72 MHz (e.g. 36 MHz on the ASTRA 3A satellite).

In case of the satellite analog television a single transponder took care of one TV channel. Applying the DVB technology to the satellite systems a single 33 MHz satellite transponder can carry 4 to 8 TV channels or 150 radio channels. A noisy satellite channel required from DVB project to define for the DVB-S systems efficient modulation techniques and error correction codes [22].



Simplified block diagram of DVB-S system (platform)

Figure above depicts a block diagram of a general DVB-S system. The MPEG-2 multiplexer multiplexes video and audio PES streams from the MPEG-2 coder with data in PES data streams coming from an IP gateway in the same way as was written in previous section. The multiplexed transport stream is randomized to spread its spectrum within its bandwidth. The randomized transport stream is equipped by an outer code (Reed Solomon code with a coding rate 188/204), interleaved (to make it more resistant to block errors) and encoded by an inner FEC code (convolutional code with a coding rate from 1/2 to 7/8). In the next phase the encoded transport stream is modulated on a carrier. DVB-S uses **QPSK** (*Quaternary Phase Shift Keying*) modulation, DVB-S2 uses 8-PSK, 16-APSK or 32-APSK (*Amplitude and Phase Shift Keying*) modulation. Afterwards, the signal is up-converted to a carrier from Ku band, amplified by a **HPA** (*High Power Amplifier*) and radiated by an antenna system to the satellite. The receiver performs opposite actions to demultiplex particular video, audio and data streams and to provide them to users using some end device (e.g. TV set, PC).

2.10 Digital Video Broadcasting via Terrestrial



The first commercial DVB-T service was implemented by the Digital TV Group in United Kingdom in 1998. Currently, digital TV broadcasting replaces classical analog TV broadcasting all over the world.

The DVB-T technology utilizes the same signal processing as was already described above to prepare the multiplexed transport stream. Differences that are relating to DVB-T standard are due to terrestrial transmission medium used for broadcasting and can be summarized as follows [18]:

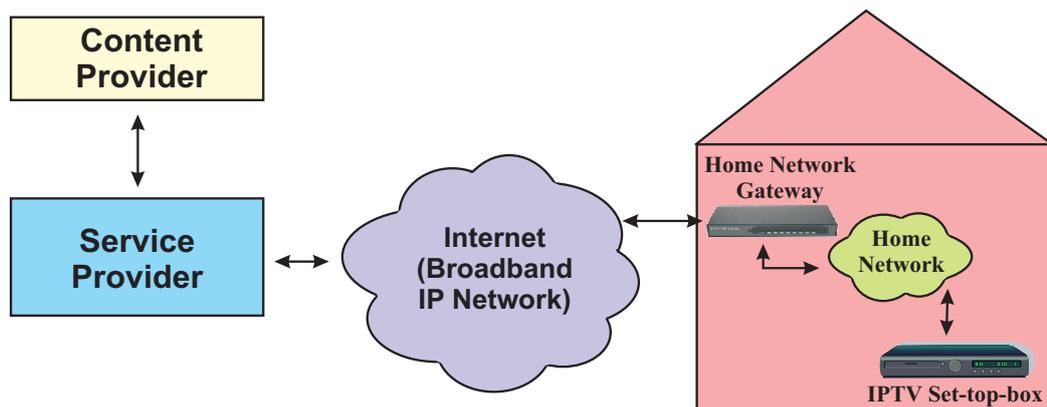
- Transmission medium – DVB-T services are aired terrestrially within the *ultra-high frequency (UHF)* band covering frequencies in a range from 300 MHz to 3 GHz. DVB-T shares the same band with analog TV therefore its implementation is based on releasing of frequencies occupied by analog television channels. An 8 MHz channel carrying single analog TV channel can carry within DVB-T several digital TV and radio channels with other information.
- Transmitting and receiving platform – DVB-T technology can reuse the same infrastructure used by analog terrestrial television (the existing broadcasters and transmitters). At the receiving side users have to buy a new end receiver that can be in the form of a standalone device – set-to-box – or as an *integrated receiver decoder (IRD)* in TV set.
- Modulation schemes – a DVB-T transmitter (as well as a receiver) realizes almost the same operations on the transport stream so as was already described in chapter about the DVB-S system. The transport stream is also randomized, protected by error correction codes (RS and convolutional), interleaved, modulated (mapped into a base band). Valid modulation schemes are QPSK, 16-QAM, 64-QAM (*Quadrature Amplitude Modulation*). Because a terrestrial environment is characterized by a multipath propagation, i.e. transmitted signals travel to destination (a receiver antenna) via several paths (due to reflections on various geographical objects: buildings, trees, hills, ground, etc.) another modulation/multiplexing method was defined for DVB-T. It is **OFDM** (*Orthogonal Frequency-Division Multiplexing*). This method transmits a signal on a number of carrier frequencies [20].

2.11 Digital Video Broadcasting over IP (DVB-IPTV)



DVB-IPTV means a delivery of DVB-Services over IP-based networks [23]. DVB-IPTV standard (formerly **DVB-IPI** as *DVB – Internet Protocol Infrastructure*) provides a set of technical specifications to cover the delivery of DVB MPEG-2 based services over bi-directional IP networks, including specifications of the transport encapsulation of MPEG-2 services over IP and the protocols to access such services.

Another important issue is the specification of the *Service Discovery and Selection (SD&S)* mechanism for DVB MPEG-2 based audio/video services over bi-directional IP networks to define the service discovery information and its data format and the protocols.



Basic IPTV architecture

A basic IPTV architecture can be seen in Figure above. The IPTV architecture and technology in detail will be characterized in IPTV section.

3 Mobile Access Network Technologies

In this part, there is introduced basic technology specification of the mobile wireless access networks, which provide access to a communication network by radio channel.



In term of used technology and radio channel, mobile access networks can be divided into two main units:

- Terrestrial access networks – access points are situated in the surface of the Earth and the communication with them is realized over radio channel in the lower part of the atmosphere.
 - Satellite access networks – access points are either terrestrial stations or satellites in their orbits. Communication is realized by passing of radio signal through different layers of the atmosphere.
-



Mobile communication networks were taken as a part of wired telecommunication networks since the beginning of the mobile communication networks development. For this reason communication protocols and interfaces were solved to satisfy requirements of networks interconnection. Characteristic element of the public and private radio communication networks is mobile switching center, which is the gateway to the wired telecommunication network [24], [25].

3.1 Terrestrial Mobile Access Networks



In the present there are used for this type of the networks so called trunking radio networks – the pack of channels is conjugate into common pool. The pool of channels is used together by several users [26], [27]. Trunking principle is the basic principle of all modern radio communication networks. These networks are divided into:

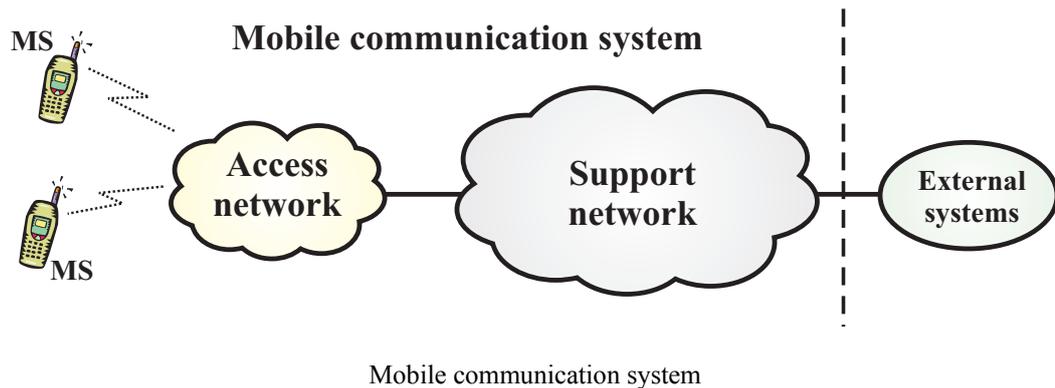
- Public radio networks (e.g. GSM, UMTS).
- Private radio networks (e.g. MPT 1327, SMARTNET, TETRA, TETRAPOL).

Besides this division, radio networks can be divided according to their coverage:

- Large coverage radio networks (full-area public networks),
 - Medium coverage radio networks (local private networks),
 - Small coverage radio networks (local ad-hoc networks).
-

3.2 Public Mobile Cellular Networks

Public mobile cellular networks are typical representatives of large coverage radio networks. The basic communication principle is the individual connection (dispatching type of network traffic is not considered). This type of network is designed as a network with a big throughput (a lot of parallel connections). The representatives of these networks are cellular mobile networks, e.g. GSM and UMTS. Mobile communication system is created of two basic networks types.



Radio Access Network (RAN) performs basic functions of connection control with a mobile terminal. RAN consists of fixed base stations (access points), which are interconnected by fixed or radio relay links.



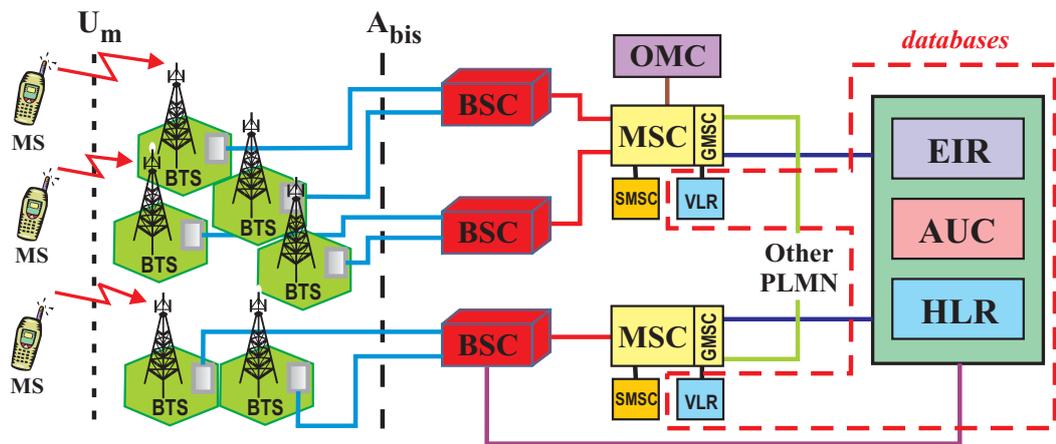
Support networks of the mobile communication systems are divided into two classes:

- **Core Network.** This is the part of the system, which provides an interconnection. Support network of the first and the second generation is created only by this core network.
 - **Backbone Network.** This network provides the interconnection of several networks and consists of equipment and transmission paths to offer high-speed data transfer rates. For this reason, backbone network is often called as high-speed data network. In 3G-support network, core network and backbone network are usually used together.
-

3.3 GSM



GSM standard (*Global System for Mobile Communication*) is, at the present, the most extended public mobile communication system of the second generation (2G), which has almost global coverage [28], [29]. GSM standard uses for creation of multiple accesses the combination of **FDMA** and **TDMA** technology (frequency division and *time division multiple access*) where one carrier signal is divided in time domain to eight time slots. Combination of frequency duplex and time duplex (FDD/TDD) is used for assurance of the duplex transmission and for uplink (from mobile to base station) and downlink (reversal way) directions separation. The width of channels is 200 kHz. Two frequency bands are devoted to GSM system, with width of 25 MHz (890 – 915 MHz for downlink and 935 – 960 MHz for uplink), i.e. the system provides up to 124 FDMA channels or 992 transmission channels. Between upper and lower band is the guard band, but this band is usually not used. Besides GSM 900 standard, the standard called GSM 1800 (before DCS 1800) is also used in Europe. GSM 1800 is a derivate of GSM 900 standard and it is used as a complement system of GSM 900 for providing the huge intensity traffic in hot spots (shopping centers, administrative centers, bus and railway stations, airports and moreover).



GSM network architecture

3.4 HSCSD and GPRS



GSM common data services are based on the circuit switching technology with maximum transmission data rate 9.6 kbps. In the upgrade of GSM network (phase 2+) ETSI defined new and faster style of data transfer. One of the main characters of GSM phase 2+ is GPRS and HSCSD standard. While the HSCSD standard represents application of circuit switching in data transfer, GPRS standard is the technology, which uses packet switching. Both standards represent 2.5-generation standard.



In **HSCSD** (*High Speed Circuit Switched Data*) standard is the data transfer realized without error correction code. This makes possible to increase data rate from 9.6 kbps to 14.4 kbps per channel. HSCSD standard also supports time slots combination and final data rate is a combination of 9.6 kbps or 14.4 kbps channels. Operators will be able to provide variant data rates from 9.6 kbps to 57.6 kbps. Operators can achieve data rates up to 200 kbps with the data compression. HSCSD standard has possibility to provide an asymmetric and symmetric data traffic. In term of resource allocation, HSCSD standard is not effective for packet data transfer, because resources are allocated only in time when the packet transfer is required.



ETSI made a standardization of a new service in GSM phase 2+. This service is based on a packet switching and it is called **GPRS** (*General Packet Radio Service*). GPRS speeds up data rates in GSM network, provides a better compatibility with LAN and WAN networks and with Internet. GPRS network uses radio resources only in a case when the data are received or sent. GPRS network provides immediate connection and high level of throughput. Whereas GSM system was originally designed for voice services, the main goal of GPRS network is to offer the access to standard data networks, working under TCP/IP protocol. GPRS network is a sub network of such networks.

3.5 3G and 4G Mobile Radio Networks

The main reason for 3G systems application is the tendency of worldwide standard, which can provide convergence of the wired and mobile networks. The idea lies in elimination of 2G systems incompatibility by using of existing wired and mobile networks infrastructure. We expect, that system should be global, which could be achieved by using enhanced cellular system. All types of networks (satellite – global coverage, terrestrial – macro-, micro- and picocell) will be covered by mentioned 3G system [27], [28].



The development of UMTS standard tended to such a solution, that revolution tendency of UMTS network development was abandoned. Evolution steps from 2G to 3G systems were preferred, especially the use of a high developed GSM standard [30], [31], [27]. There were three crucial decision accepted with creation of infrastructure:

- To use the multiple accesses based on **CDMA** (*Code Division Multiple Access*) for radio interface.
- To create the UMTS terrestrial mobile access network by ATM transfer mode.
- To use the enhanced GSM 2+ network elements for UMTS core network.

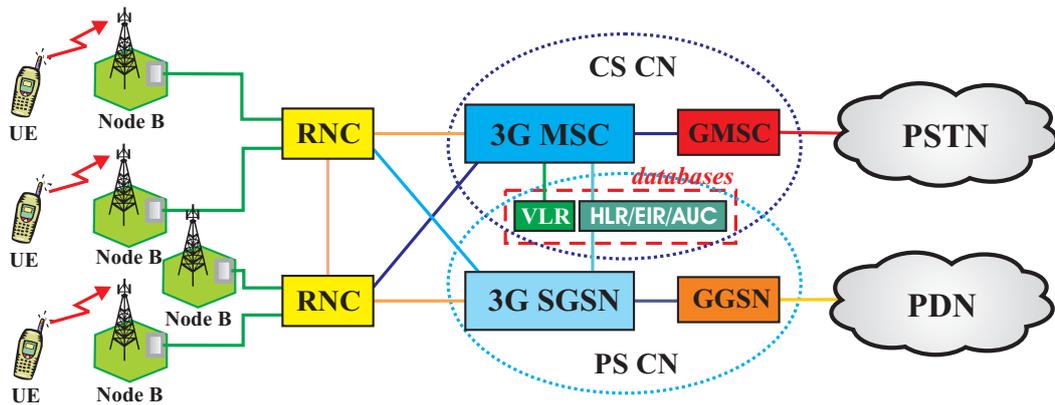
Because the GSM standard is widespread global world standard of 2G networks, in the first step it is expected to use a connection of *UMTS Terrestrial Radio Access Network (UTRAN)* to existing but newly regenerated core GSM network. That means that GSM network will serve users of both technological standards in parallel. Radio subsystem of GSM network and UTRA network will work as two different, but cooperating and subsidiary access networks in generic network infrastructure, which can be regarded as GSM and also UMTS core network. The difference between them lies in the modernization of the some nodes of GSM network for UMTS network. In the second step the core network will be changed to IP network. 3G network architecture is on Figure below.



In different prognosis of 2G to 3G networks evolution, there exists standard, which is often marked as intermediate step between GSM and IMT-2000 networks – standard **EDGE** (*Enhanced Data Rates for GSM Evolution*). EDGE belongs to 3G standards, but on the contrary to UMTS standard, it needs essentially minor changes in radio access network and core network. For this reason, EDGE represents a simpler solution for operators of existing GSM networks. The EDGE actually represents 2nd generation of HSCSD and GPRS systems. This standard is not able to provide transfer rates up to 2 Mbps. The advantage of this standard lies in its simpler application without requirement to change the infrastructure. For this reason a lot of operators consider IMT-2000 like ideal standard for newly built 3G networks.



The 4th generation of mobile communication systems (4G) probably won't be based on the creation of a new standard, because it is clear, that neither 2G nor 3G networks were successful in enforcing the worldwide global standard. The main motive for building a 4G system, which is often called as system beyond 3G (B3G), will be probably tendency of economic success based on users' requirements for new and advanced services with high security and reliability. Also, there will be a tendency for budget-priced and easy terminals with long lifetime of batteries.



G3 network architecture

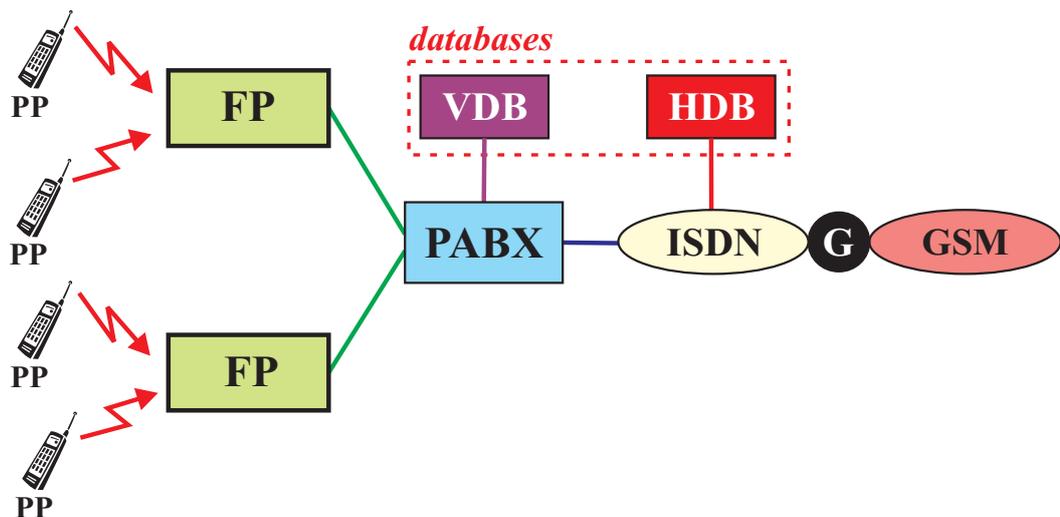
3.6 Cordless Telephone (DECT)



DECT (*Digital Enhanced Cordless Telephone*) standard is a boundary milestone of the long-term development. Since from beginning, this system was designed for wireless telephony in consideration of GSM standard and with connections to other networks. It is not proper to consider this system as a replacement of already existing networks, but as creation of a bridge between wireless and cellular technologies.



DECT is full digital system, transparent for PSTN services, providing mobility in user' premise (home, work). Coverage is realized by picocells with small cells (radius approximately 50 m), channel selection and channel allocation is dynamic (DCS/DCA), handover of calls is without interruption, provides roaming. Data transmission rate is 24 to 522 kbps (2 Mbps in the future). System has big capacity (10 000 Erl/km²). This system is very flexible, because of predefined profiles, which enable cooperation with other networks. Primarily, DECT system was prepared for radio coverage of small areas and for cooperation with GSM network, which is not constructed for such small areas.



DECT network structure

3.7 Private Mobile Networks

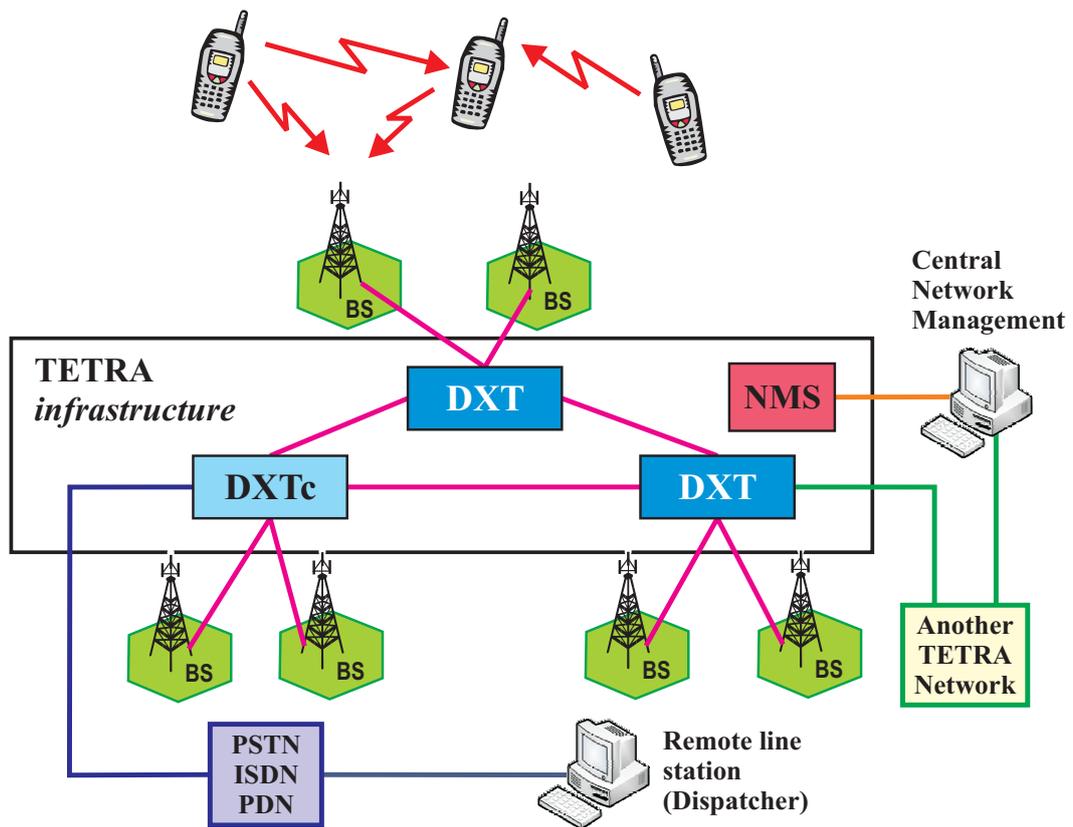


A lot of financial resources for building of an infrastructure by means of base stations and antenna towers were often required to invest for necessity of radio communication coverage in the case of the classical radio networks by little and major users. Public cellular networks have a certain specifications, which support team works of subscribers, but with certain restrictions. This absence is mostly sensitive in work of some organizations like Police, Customs Service, emergency teams [27], [28].



The solution of these problems lies in private radio trunking networks, which are able to use the same network for several organizations with preserving of secrecy and guarding of data and voice transfer. These networks provide an access to radio channel without great investment because they are able to join one or more systems and to invest to necessary mobile and portable terminals.

Standard for analog private radio networks in Europe represents the group of standards MPT 1317. This group consists of four standards, from which the best known is MPT 1327 one. **TETRA** (*Trans-European Trunked Radio*) is the first European opened digital radiotelephony standard defined by ETSI in 1995 [32]. The same way, as public mobile networks were progressively substituted by GSM network in a mobile radiotelephony, digital networks based on new standard will substitute today's analog terrestrial mobile private networks. TETRA network architecture is in following Figure.



TETRA network architecture



Among basic services of the network belongs the distribution of information for specific group or for each network user (analogy to paging networks). The other services include: e-mails, fax and SMS messages, transfer of data files, and safe access to databases or transport of information from GPS system.

From technological aspect, TETRA consists of two base standards:

- TETRA Voice + Data,
- TETRA Packet Data Optimized (TETRA PDO).

TETRA V + D is a radio trunking network standard for voice and data transfer. TETRA PDO is special version of packet transfer in radio channel, permitting of high effective using of limited radio spectrum. TETRA PDO can realize quite high transfer rates (36 kbps), and with superior methods of compression, system can transfer video sequences (for example: police can send signal of video record from the place of accident to police station, where it can be analyzed). System TETRAPOL, like TETRA is a digital private communication network, but it works on different multiple access method (TETRA – FDMA/TDMA, TETRAPOL – FDMA). Standards are not compatible.

3.8 Ad-hoc Networks



Ad-hoc network is a network created without any central control or management. This network is based of mobile nodes, which use wireless interface to transfer of data packets. Nodes in network are able to work as routers and they can route packets for other nodes. Ad-hoc connection is based on peer-to-peer type of communication. To provide a connection among mobile units it is not used any cable infrastructure and there is no central control to manage a creation of connections and to support the coordination and communication. Furthermore, there is no intervention from operators.

In general, in ad-hoc networks all devices, which share a common space, will also share common channel and they will be equivalent in this sharing each other.



These networks can be applied for network creation in such areas, where there is no infrastructure available, for example by emergency operations in far-away areas and for wireless public access in metropolitan areas – access nodes can serve as fixed relay stations for packet routing among each other [27]. On local level they are used for connection of notebooks, palmtops, e.g. at the conference, creation of home network, creation of personal networks and also for surroundings monitoring and realization of WLAN networks.



Wireless Local Area Network (WLAN) as a main representative of Ad-hoc networks can operate in two configurations, either as independent configuration (ad-hoc) – stations communicate directly and there is no necessity to install any supporting infrastructure, or as distributive system configuration – configuration expects existence of *access point (AP)*, which at the same time works as a base radio station and as a data bridge. Wireless local networks can be divided into radio technology networks and *infrared (IR)* technology networks. IEEE 802.11 (WiFi), HIPERLAN and Home RF standards belong to radio technology networks [28].

Bluetooth



Bluetooth technology represents the next radio technology of a short range, it has to be able to work in ad-hoc networks, which can be either independent, or works like a part of IP networks in all over the world, eventually as combination of both possibilities. The goal of this technology is to replace the cable connection among electronic devices by radio channel by means of cheap radio chip. The key characteristics of this technology are robustness, small complexity, low power and small price. Bluetooth works in **ISM (Industrial-Scientific-Medical)** band of 2.4 GHz and uses frequency hopping to eliminate of interference and fading. Coverage is about 10 meters (possible connection through walls of building), transmission rate is 780 kbps (one-direction transmission is 721 + 57.6 kbps, symmetric transmission is 432.6 kbps).

The following table presents bit rates survey, which networks technologies of particular generations can ensure.

Mobile technologies and their bit rates

Distribution of technology		Maximal bit rate downlink/uplink [Mbit/s]
2G/2.5G/2.75G	GPRS	0.080/0.040
	EDGE	0.236/0.236
	EDGE Evolution	1.9/0.9
3G/3.5G/3.9G	UMTS	0.384/0.384
	HSPA	14.4/5.75
	HSPA+	56/22
	LTE	360/80
	Flash – OFDM	15.9/5.4
4G	WiMAX	144/35
	LTE Advanced	1 Gbit/s fixed connections and 100 Mbit/s mobile
	WiMAX IEEE 802.16m	
Others	WiFi 802.11b,g	54/54
	WiFi 802.11n	600/600

4 IPTV (Internet Protocol Television)

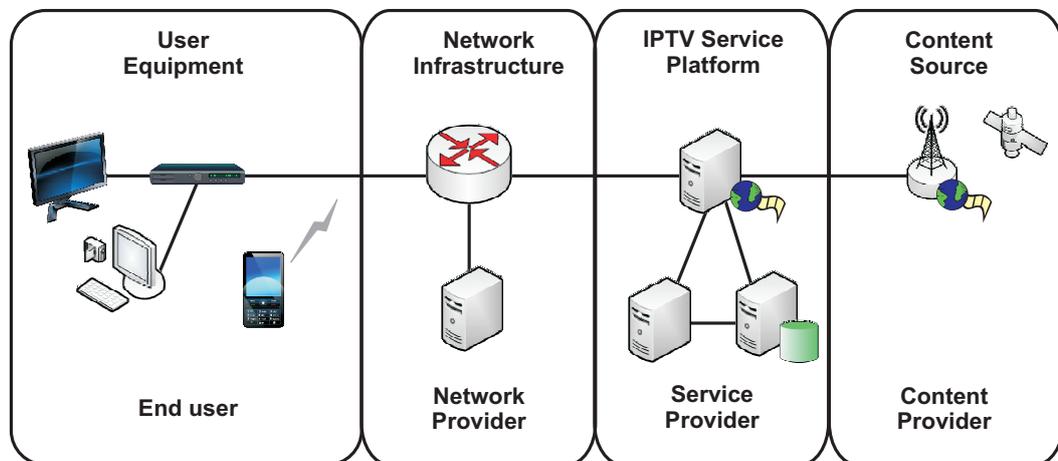
4.1 Introduction

The end user finally perceives the quality and **IPTV** (*Internet Protocol Television*) service portfolio, as well as the usability in order to satisfy his requirements. Several actors are responsible for the delivery of the content from their originators such as TV stations and studios but probably also from other users. In this section the IPTV domains and services are described together with the explanation of how the standardization develops from requirements to architecture.



End to end chain for delivery of the IPTV content to the end user usually contains these 4 main domains that are involved in the provision of an IPTV service (Figure below):

- Content provider,
- Service provider,
- Network provider,
- End-user.



IPTV Domains



The four IPTV domains definitions could be provided by the ITU-T or ETSI TISPAN specification [34], [35], [9]. Most of the standardization bodies follow the same schema to produce end to end solution specifications that apply also to the IPTV. First of all it is necessary to specify all requirements for the service but also from the UE and network capabilities point of view (stage 1). Secondly it is the specification of the functional architecture, functional entities and their task, relevant reference point among the functional entities as well as high level procedures for services (this is done usually in stage 2). In the final stage 3 it is

required to conclude all details needed from the implementation perspective as for example the protocol models and detailed protocol procedures.

There are two main aspects of the IPTV. First one is technological one resulting to the IPTV architecture and second one is the user's perspective aspect which can be seen from the provided IPTV services and user experience.



From the user's perspective is not really important what architecture the IPTV service provider selects, but it is surely more important which services are provided. Most of the existing non-NGN solutions provide only basic set of services like linear TV (live TV channels), *video on demand (VoD)*, and some of them also **PVR** (*Personal Video Recording*).



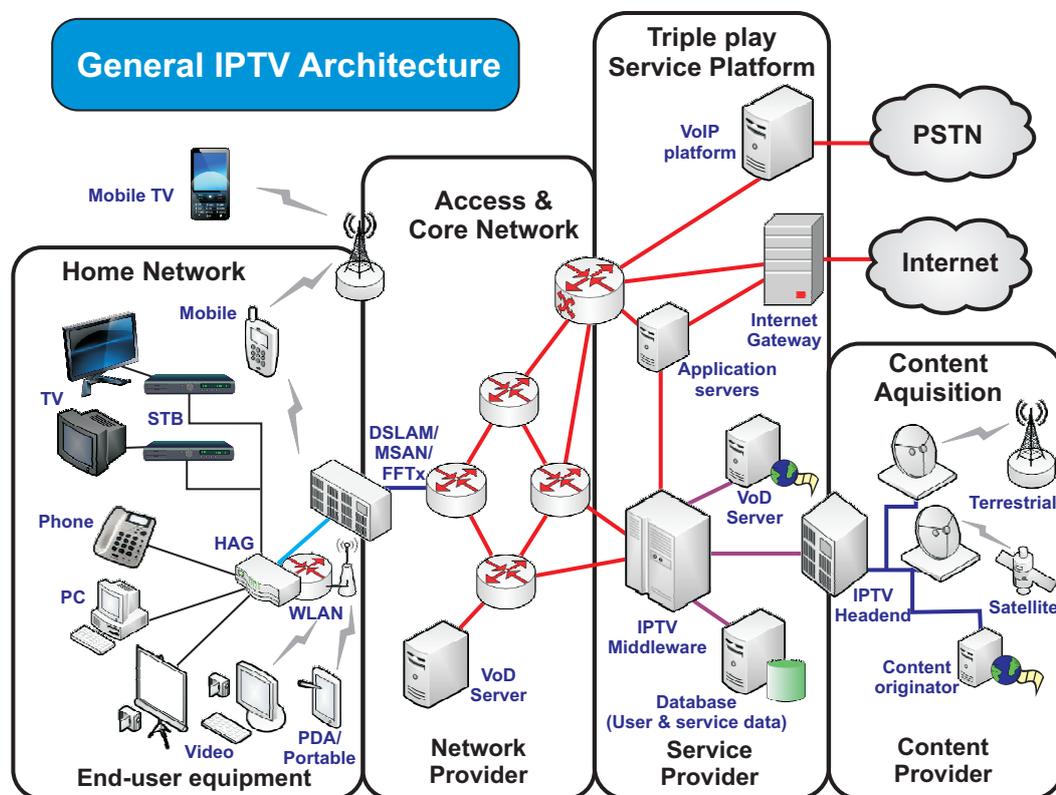
New NGN based IPTV solution should therefore provide much more services, features but most important also new user experience in watching TV with more interactivity, personalization, mobility and last but not least comfort in consumption of the right content in the right time and right way.

4.2 Architecture of non-NGN Based IPTV



The general Triple Play architecture (Figure below) usually consists of the following parts [4]:

- Service platform domain including IPTV middleware (non-NGN)
- Transport network
- Access network
- Home network and CPEs



General IPTV architecture



The Triple Play service platform usually contains several less independent parts of complex service architecture:

- Content acquisition subsystem which allows to receive, process, and encode content from external sources to defined media coding and encapsulation (receiver and decoders infrastructure, IPTV headend, VoD import and pre-processing).
- Content distribution subsystem responsible for retrieving, protecting, distributing, storing and delivering of the content by preferred way to the end user's system (user equipment).

- IPTV middleware contains the application servers which control and manage the whole IPTV infrastructure (servers, databases, frontend, backend systems, interfaces to external systems e.g. OSS/BSS), users and services. Part of the application platform could also be additional IPTV applications or gateways allowing limited interaction with other systems (e.g. VoIP, NGN).
 - Service selection and discovery subsystem which allow the user to browse and find via user TV portal an appropriate content or service information (metadata) which he would like to watch (could be part of IPTV middleware).
 - VoD, nPVR or other subsystems – specialized subsystem infrastructure required for dedicated services (Video on Demand or network based personal video recording service).
-

For the Triple Play contains tree type of services – video, voice, data – the connection to internet services and voice service platform is required (e.g. over VoIP gateway).

There is no single approach to the IPTV service provisioning. Due to huge costs involved in the network equipment, operators usually follow incremental approaches to network upgrading, always relying on existing premises and procedures. Therefore the way a new NGN service is provisioned, it clearly depends on the history of the operator. Therefore there are a lot of differences from solution to solution and also to operator specific transport, access and home network design.

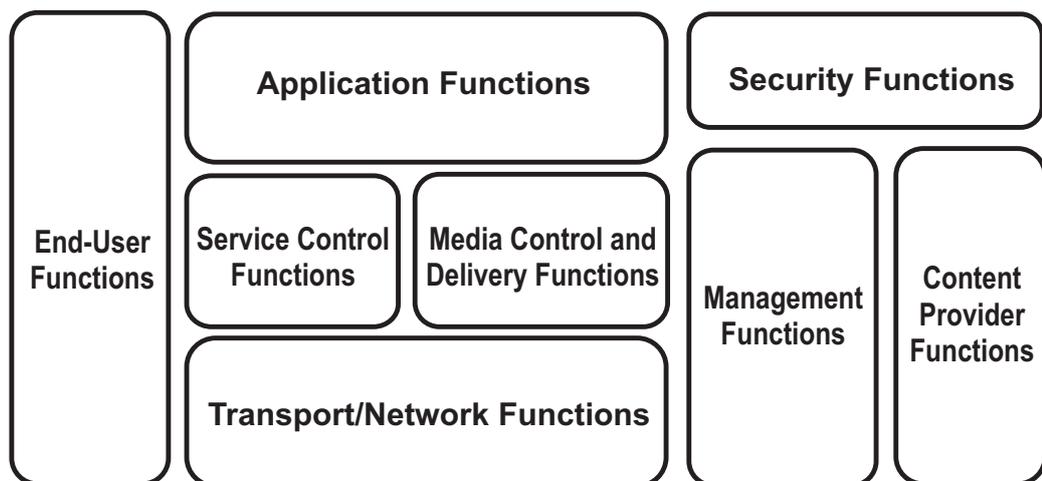
4.3 Architecture of NGN Based IPTV

The major players in any IPTV delivery chain consist of content providers, service providers, network providers, end-users. Content provider is a source of content as for example TV stations, studios, content aggregators, etc. The IPTV platform usually own by service provider has to provide all functions necessary for control and delivery of IPTV services over network infrastructure (network provider) to end user.



Main blocks NGN based IPTV platforms are following (Figure below):

- Application functions
- Service control functions and User profiles
- Media control and delivery functions
- Supporting, management and security functions
- End user function



High-level architecture of NGN based IPTV functional



Application functions can include several service logics of the IPTV services, mechanisms for service discovery and selection to find right services and content, also help to interact with other application and external systems.

Service Control Functions provides functionality for authentication, authorization of service requests. This function is also responsible for the setup and control of all the IPTV services. It can also reserve resources towards transport control functions.

User Profiles contain user data and user profiles related to user's services.

Media Control and Delivery functions has received content and media streams from content provider and then control and provide media processing, media delivery, content storing, transcoding and relaying of content.

End-User Functions represent home network and user equipment as for example end devices (e.g. TV with set-top-box, mobile, etc.) but also home networking part including Home Access Gateways.

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The greatest advantage of NGN based IPTV architecture is possibility integrate IPTV services with other NGN services, reused existing NGN capabilities, better utilized resources, personalization of services and mobility.



NGN/IMS function which can be re-used for providing IPTV:

- User registration and authentication,
 - User subscription management,
 - Session management, routing, service triggering, numbering,
 - Interaction with existing NGN service enablers (presence, messaging, group mng.,etc.),
 - QoS and bearer control,
 - Mobility, FMC capability,
 - Charging and billing,
 - Security and management mechanisms.
-



There are IPTV specific functions which have to be additionally described:

- Service discovery & selection, presentation, e.g. EPG,
- Service & Content protection, e.g. DRM and CAS,
- Service & Content management, managing the services and contents in the Content Provider domains and/or the Service Provider domains,
- Content distribution, delivery and locating control,
- Multicast support and control,
- VCR control, e.g. play/pause/fast-forward/rewind.

The producing of specification is usually defined by standardization bodies in 3 stages:

- Collect Service and system requirements, service use cases,

- Define functional entities and architecture, reference points, service procedures,
 - Specify the implementation, signaling flows, protocols details.
-



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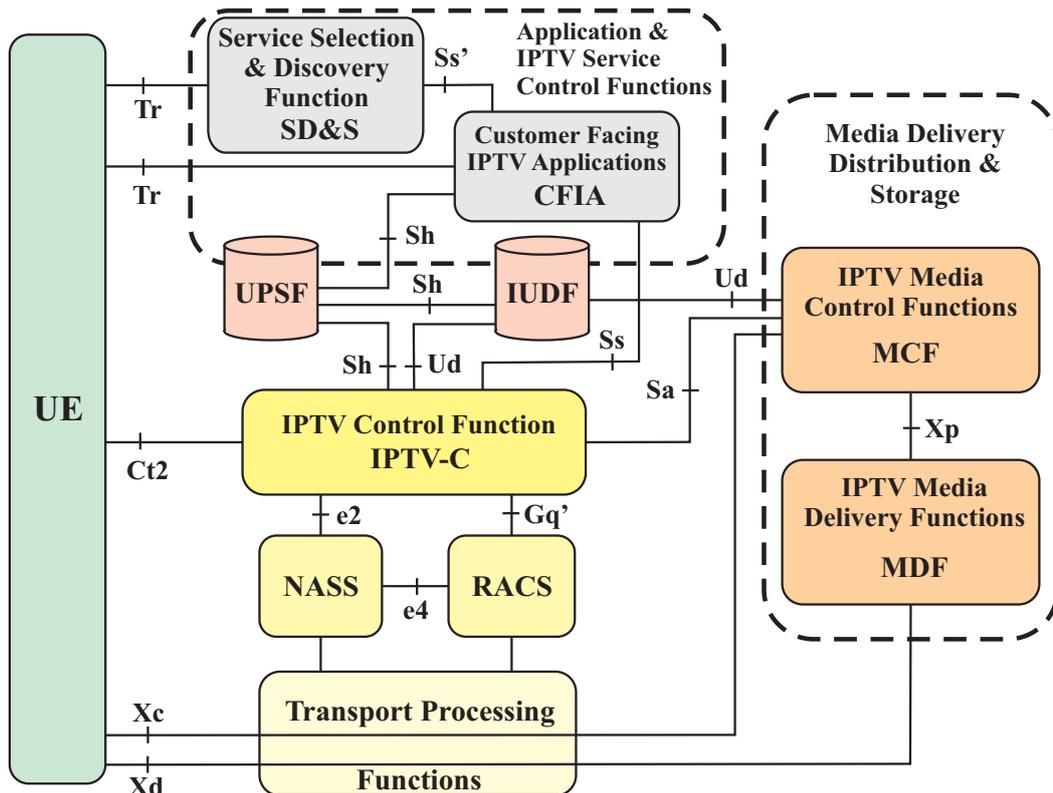
4.4 NGN Integrated IPTV (non-IMS) Architecture

The Concept of NGN integrated IPTV subsystem architecture (formerly in Release 2 called NGN dedicated IPTV subsystem) describes how to integrate IPTV functions into the NGN architecture. TISPAN NGN Integrated IPTV subsystem is what ITU-T or ATIS call non-IMS NGN based IPTV. The proposed architecture focuses on closer integration between IPTV services and features with NGN network and its subsystems (NASS, RACS, UPSF) but also migration scenarios from existing solutions (i.e. DVB-IPI, ATIS-IIF) into TISPAN NGN and common components (Figure below). Several parts of system are complied and based on existing standard like those from DVB-IPTV and in fact DVB recognizes this solution as potential architecture for easier implementation and interworking as with IMS based IPTV [36].



Core IPTV function:

- *Service Discovery & Selection (SD&S)*
 - *IPTV Control (IPTV-C)*
 - *Client Facing IPTV Application (CFIA)*
 - *IPTV User Data Function (IUDF)*
 - *Media Control Function (MCF)*
 - *Media Delivery Function (MDF)*
 - *User Equipment (UE)*
-



Simplified TISPAN NGN dedicated IPTV subsystem functional architecture

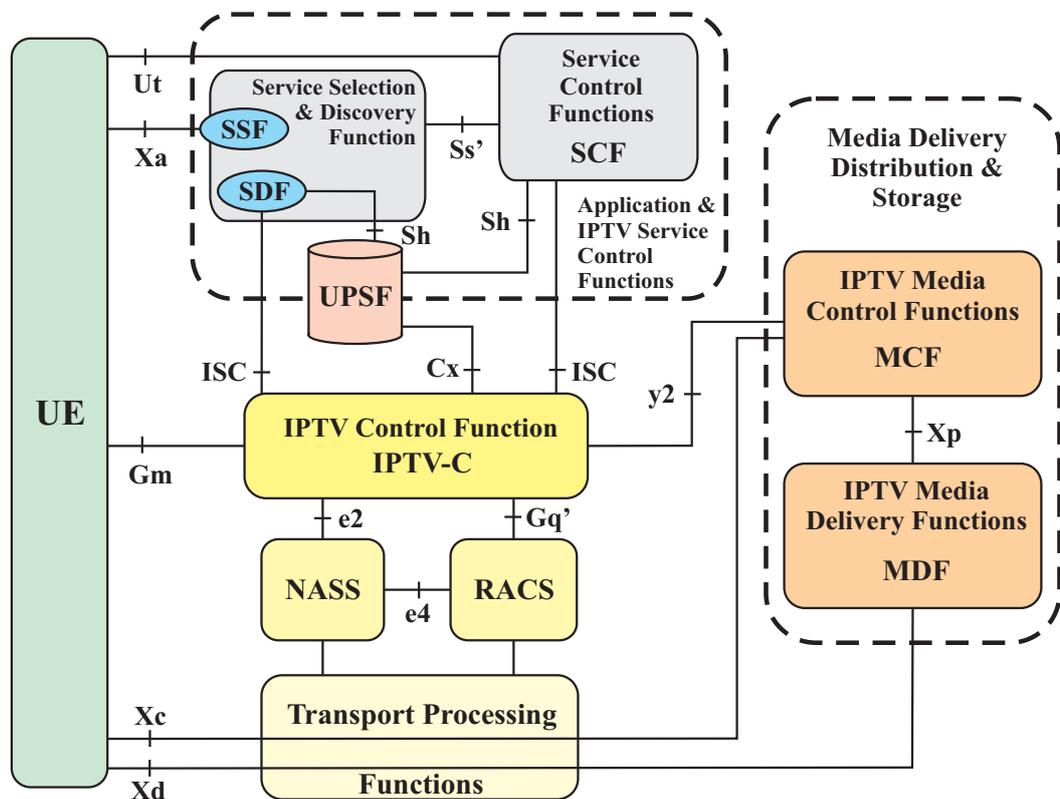
The user can have (using his user equipment – UE, like a set-top-box STB) access to the service description (e.g. Electronic Program Guide) via SD&S service selection and discovery procedures follow DVB IPTV specification and use http protocol over the Tr reference point. The same Tr interface can be used by UE for accessing the user interface and service selection over Customer Facing IPTV application (CFIA). CFIA provides over Tr http based interface IPTV service provisioning, selection and authorization. IPTV control (IPTV-C) is enabled over Ct2 interface (http or RTSP control). Media (e.g. *content on demand* – **CoD**) can be streamed by unicast or multicast over Xd from Media Delivery Function (MDF). For media control, as for example the trick play command, RTSP protocol is used via Xc by Media Control Function (MCF).

4.5 IMS Based NGN IPTV Architecture



Second concept for providing IPTV services over NGN architecture is described in TISPAN IMS based IPTV [36], [1]. Main difference as name said that IP Multimedia Subsystem is used to service and session control of IPTV services. Main advantage is reusing existing capabilities (IMS registration, authentication, session management, routing, service trigger, identity management, personalization, mobility, charging) of IMS and possibility to integrate service control layer to unified service control platform by utilization for IMS.

- Disadvantage of IMS based IPTV is higher complexity and less backward compatibility with existing standards (beside that several part of concept reused existing standard and protocols).



Simplified TISPAN IMS based IPTV functional architecture



IMS based IPTV functional entities (see Figure above):

- *Service Discovery Function (SDF)*
- *Service Selection Function (SSF)*
- *Service Control Function (SCF)*

- Core IMS elements (P-CSCF, S-CSCF, I-CSCF)
 - *Media Control Function (MCF)*
 - *Media Delivery Function (MDF)*
 - *User Equipment (UE)*
-

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The IMS based IPTV has number of advantages because IMS can act as unified service control subsystem for all NGN services instead of establishing an additional specialized subsystem (case of NGN dedicated IPTV subsystem). Additionally IMS can more naturally support mobility, interaction with NGN service enablers (like messaging or presence), service personalization or quadruple play services (voice, data, video, mobile).

4.6 NGN Based IPTV Services

Main goal of NGN based IPTV is to provide to end user a comprehensive list of converged IPTV services [2]. New generation IPTV services have to provide service personalization, interactivity, blending of services, user targeting, enhanced accessibility and mobility.



IPTV services can be split to three groups:

- Basic IPTV services
 - Advance IPTV services
 - Converged IPTV services
-



Basic IPTV services consist of minimal set of IPTV services which are expected from NGN based IPTV service provider:

- Broadcast TV (with or with trick modes) – delivery of linearly broadcasted TV channels.
 - Trick Modes – enable control playback and pause, forward, rewind content.
 - *Pay Per View (PPV)* – user pays for example only for particular show or time period not whole TV channel or TV package.
 - *Content on Demand (CoD)* – user requests content consumption on demand (e.g. Video on Demand or Music on Demand).
 - *Personal Video Recording (PVR)* – user can record content in network (network or n-PVR) or locally in STB (client or c-PVR).
 - *Electronic Program Guide (EPG)* – provides service selection information required by the viewer to find and select the programmes to be watched.
 - Parental Control – protection mechanism to limit access to television content for children's in age below rating of programme.
-



Advance IPTV services:

- Profiling and personalization – feature which enable personalized IPTV services based on user preferences and user profile. Provider can also use the information about user's behavior and content consumptions.
- *User Generated Content (UGC)* – content produced by the end user with the intention to share it with other users.
- *Content Recommendation (CR)* – service advisory for favorite shows based on user's preferences and behaviors.

- *Time Shift TV (TsTV)* – user can browse and play from past broadcasted content pre-recorded by provider.
 - *Personalized channel (PCh)* – user specific list of programs that are scheduled as playlist for personalized preview.
 - *Targeted advertising (TAI)* – advertising mechanism which is targeted to specified group of user based on his user profiles.
 - *Interactive TV (iTV)* – service providing interactivity between provider/broadcaster and end user or between several users.
-



Converged IPTV services:

- Convergence of the IPTV and other NGN Service and their interaction (e.g. presence based game, incoming call notification, sharing the remote control).
 - Interaction with 3rd Party application (e.g. Parlay) enable interworking with 3rd party developed applications
 - Interaction with Internet Services (e.g. convergence of IPTV with web 2.0 services and social media)
 - Service Continuation between IPTV UEs allows park service on one device and pick up and continue with service consumption on other one.
 - Service Continuation between Fixed-Mobile and support for mobility/roaming, service accessibility cross different access networks and terminals
 - Remote control of IPTV services for example control of recordings or content recommendations.
 - Hybrid IPTV services (combination of Satellite/Terrestrial delivery with IPTV)
-

4.7 Protocols Used for IPTV

In IPTV networks are used standard protocols **SIP**, **SDP**, **RTP**, **RTCP**, **HTTP** and **DIAMETER** for NGN and their combinations [14], [37], [38]. Their performance charts are listed in first section (NGN). Specific protocols **IGMP**, **MLD** and **FLUTE** are used too.



The Internet Group Management Protocol allows hosts or UE to participate in multicasting (by joining the stream) and describes the basic of multicasting IP traffic, including the format of multicast IP addresses, multicast Ethernet encapsulation, and the concept of a host group.

MLD and **MLDv2** protocols are designed for use only in IPv6 network. This protocol is follower of IGMP in IPv4 networks used for joining some network node to multicast group. Multicast groups are separated by different IP addresses.

The *File deLivery over Unidirectional Transport* (**FLUTE**) protocol (RFC3926) is a protocol for the unidirectional multicast delivery of files over the Internet.

5 HbbTV (Hybrid Broadcast Broadband TV)

5.1 Introduction



HbbTV (*Hybrid Broadcast Broadband TV*) is a new industry standard providing an open and business neutral technology platform that seamlessly combines TV services delivered via broadcast with services delivered via broadband and also enables access to Internet only services for consumers using connected TVs and set-top boxes [40].



You can find a full range of connected TVs nowadays. Every major brand has its own IPTV platform, such as Panasonic has VieraConnect or Samsung has Smart TV, they are offering a mixture of catch-up services and additional content, such as movie trailers, YouTube access, and other applications.

HbbTV is similar to **MHEG** (*Multimedia and Hypermedia Expert Group*) it means, that an application can be delivered to the TV using a data carousel, too. That application can later work with or load additional content from the internet. HbbTV is also an ETSI standard (TS 102 796).



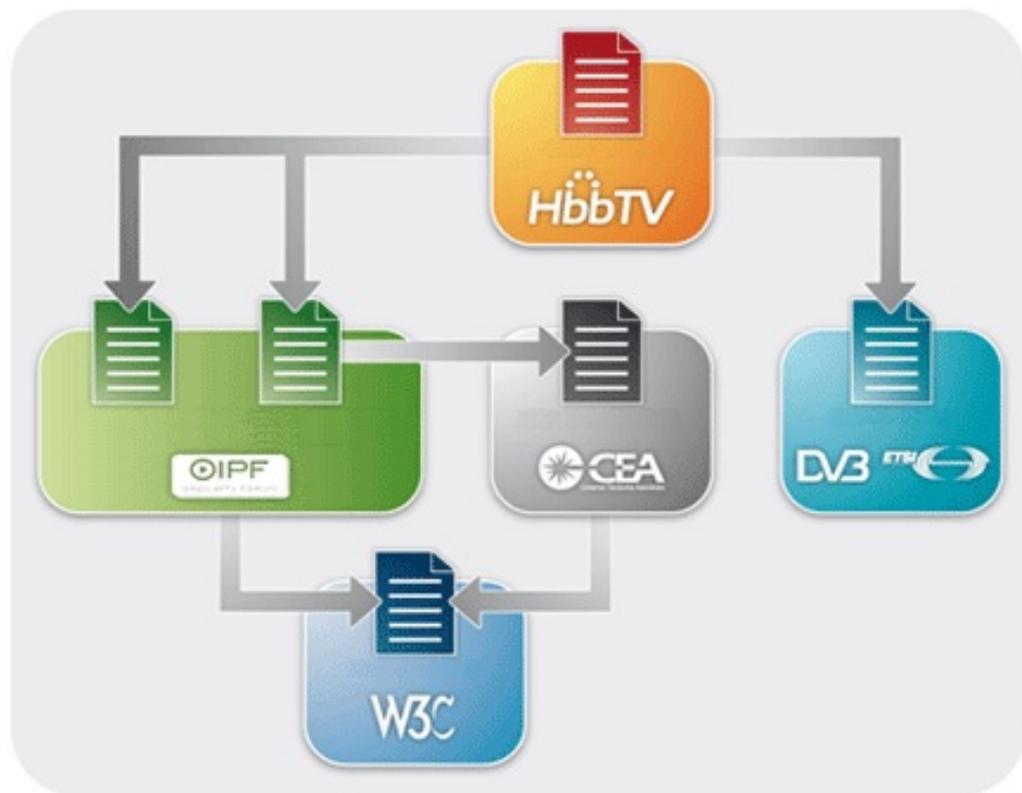
HbbTV doesn't depend on a particular broadcast link or on a particular IP link – it'll work with either [40]

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HbbTV applications, however, can be downloaded from an application portal in the TV and delivered solely over the internet. So it can support content services from providers who don't own broadcast channels.

Other advantage is that, it is based on technologies that are well known to web developers, mainly CE-HTML, a specification that includes XHTML, Ajax, CSS, and Javascript (jQuery). Because of that, content providers can build and release their apps more quickly.

HbbTV's Javascript API was extended to provide TV functionality, such as handling channel changes. This standard expects that host TVs have a minimum display resolution of 1280x720.



The HbbTV specification corrals standards and technologies, including CE-HTML, DVB and JavaScript, from many existing organizations, such as the Open IPTV Forum and the W3C [40]



The HbbTV specification also contains a contribution from the Open IPTV Forum, which has defined a set of audio and video formats that should be supported. HbbTV relies on the AVC (H.264) codec for both standard and high-definition video, with either E-AC3 or HE-AAC for audio. Audio streaming services use either MP3 or HE-AAC.



5.2 Services

Services delivered through HbbTV include

- enhanced teletext,
- catch-up services and video-on-demand (VOD),
 - Catch up TV is a term used to describe VOD in which TV shows are available for a period of days after the original broadcast.
- Electronic program guides (EPG),
 - provide users of television, radio, and other media applications with continuously updated menus displaying broadcast programming or scheduling information for current and upcoming programming
- interactive advertising,
- personalisation
- personal video recording (PVR),
 - similar to a VCR but records television data in digital format
- voting and games,
- social networking, other multimedia applications

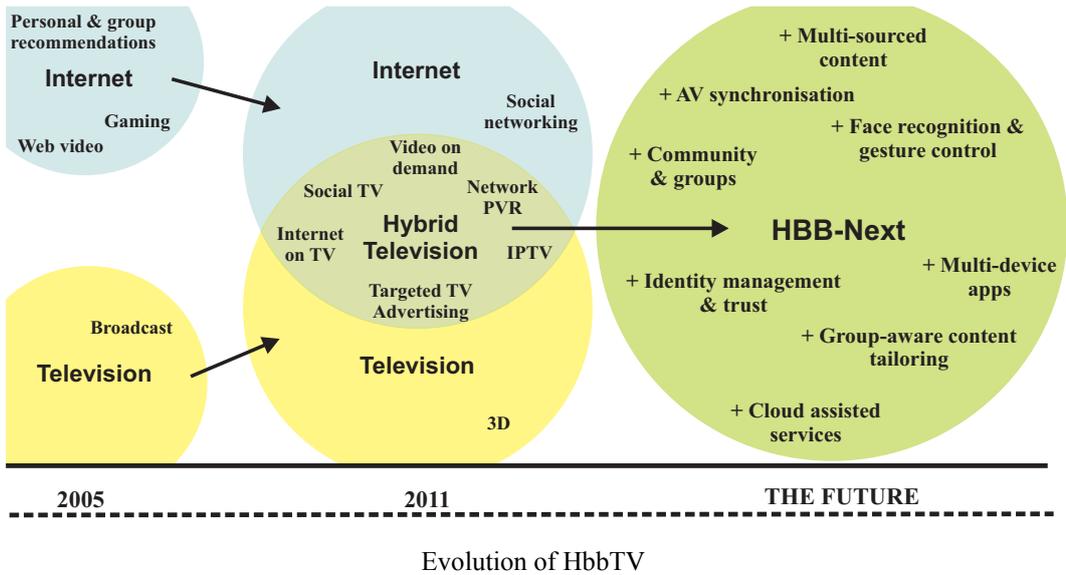


Example of HbbTV service

5.3 HBB-NEXT



HBB-NEXT seeks to facilitate the marriage of the broadcast and Internet world by researching user-centered technologies for enriching the TV-viewing experience: Multi-user tailored content recommendations, access via multiple devices, social media features, or user generated content [41].



5.4 Multiple-User Environment

In the area of catering for multiple users, the HBB-NEXT Service Personalization Engine will be developed. It will provide multi-user content recommendations, multimodal interaction and content awareness. The concept foresees a front-end functionality with a multi-modal user interface for a multi-user environment and a back-end providing context-aware multi-user content recommendations.

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Multimodal interfaces promise to allow users to enter requests and information using any of input modes such as a mouse, keyboard, handwriting, speech, and gestures. Speech and gestures are great candidates to fulfil many operation tasks which have been previously possible only by using controller. Multi-speaker identification aims to identify more speakers based on a recorded signal that may contain utterances of more individual.



Using multilevel authentication the user can be authenticated by face (2D, 3D, Iris), speech, gestures and not only by using personal code (pin code, password, etc.). 3D face detection has more possibilities than 2D detection. In 3D detection face curves are very important as the technology is virtually "touching your face". It is far more difficult to cheat the 3D technology.

5.5 Multiple Devices

The synchronization and presentation of different audio and video sources on single and multiple devices are required. HBB-NEXT will provide solutions to this.



Furthermore, resource/computing limitations of mobile devices need to be overcome. This will be done by employing an offloading mechanism and cloud computing solutions so that low performance video devices may serve all HBB-NEXT features also on heterogeneous devices.

5.6 Identity And Trust

HBB-NEXT will enhance multimodal and multilevel user identification technologies. The project will also focus on developing a trust management framework for HBB applications. This framework will be defined by a comprehensive trust and reputation management solution. All these objectives aim to maintain a high level of security and protection for end-user, service developers and the platform itself.

5.7 Standardization

The founding members of the HbbTV consortium together with a large group of supporters jointly developed the HBBTV specification to create a global standard for hybrid entertainment services. Version 1.1.1 of this specification has been approved by ETSI as ETSI TS 102 796 in June 2010.

The HbbTV specification is based on existing standards and web technologies including OIPF (Open IPTV Forum), CEA, DVB and W3C. The standard provides the features and functionality required to deliver feature rich broadcast and internet services. Utilizing standard Internet technology it enables rapid application development. It defines minimum requirements simplifying the implementation in devices and leaving room for differentiation, this limits the investment required by CE manufacturers to build compliant devices. HBB-NEXT is the next step that will enhance existing standard by multiple – user environment, use of multiple devices (even mobile devices), new ways of interactions and security.

6 CDN (Content Delivery Network)

6.1 Introduction



Content Delivery Network (CDN) is a network, which consists of huge number of distribution points (called nodes). This network is running over the Internet network and can defectively deliver huge amount of content to huge amount of recipients.



The most popular usage of these networks is distribution of software updates.

CDN is owned by a CDN provider, which is responsible for content distribution, which is ingested by a CDN customer.



The advantage of CDN is in resource sharing, because if every software developer wants to distribute updates over his customers in Internet, he must build system of servers, maintain and operate its. Then there will be numbers of parallel systems to the same point. This overlap and low utilization of nodes will increase the price/usability ratio. Distribution from one central point will use huge amount of capacity for transferring the same thing many times.

CDN network is organized in two layers:

- control layer – mostly central node, which is used for content ingestion and distribution over the distribution layer. It is also responsible for Authentication and Authorization functionality if implemented.
- distribution layer – forest of nodes, distributed over many locations.

CDN distribution is split in two methods:

1. on-line – is the same as multicast, all recipients get the same data at same time.
-

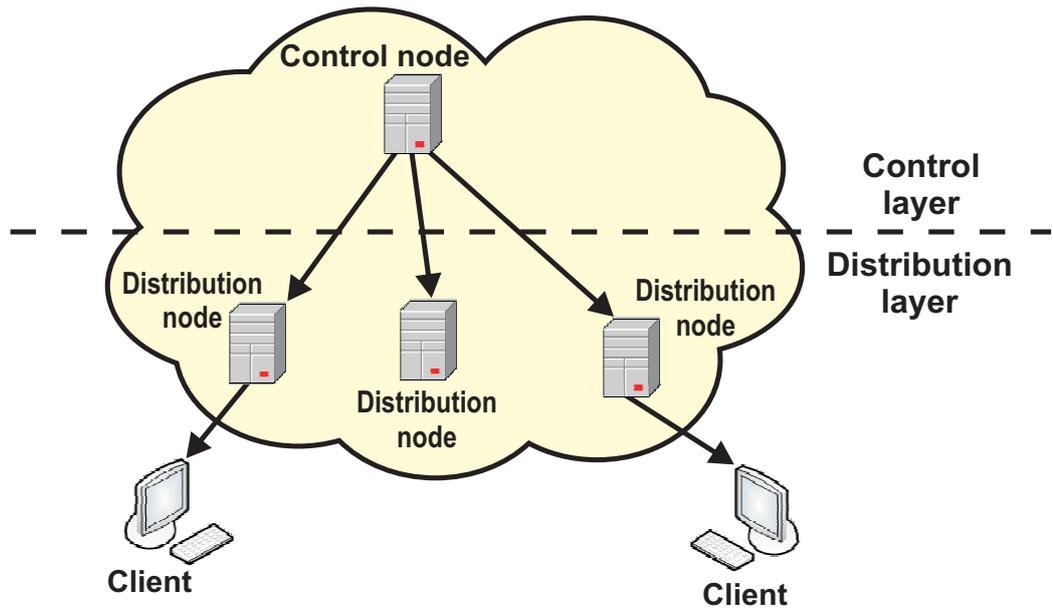


Good example is sport match, when every customer wants to see the match with minimal delay (acceptable value is tens of seconds).

2. off-line – information which generation has started and ended in the past. Off-line distribution can be split in two types of delivery methods: *download*, *streaming*.
-



Example can be well known video stream (YouTube) or distribution of updates.



CDN Architecture

6.2 Nowadays CDNs in World

CDNs are now currently implemented as a separate commercial product. To provide unique functionality and effective operation, CDN operators have implemented their own algorithms and transfer protocols. There also exists commercial software for CDN creation at own premises of Telecommunication operators and other companies interested in own CDN implementation, e.g. EdgeCast [42].

-
- + Open-source projects are based on activities of single projects, which have developed their own CDN networks based on open source software. Every volunteer can join this network with his own server or cluster. This new server will become a new node of open CDN and will be controlled by this open CDN network management.
-

6.3 Content Flow

Nowadays, the Internet content based service are being split in main parts: *Content, Service, Network (Transport), Consumer*. All these entities have clear relations among them, which defines clear content flow from a content originator (author) to a content consumer. These parts are provided and represented by adequate entities. Content is originated from the author. Then the author passes content with right to a content provider. The content provider will then provide content to service operators. Service operators are offering service consisting of content to the end customer via network. Network is provided by a network operator. Not all the entities are necessarily separated from each other. Mostly, the service provider and network operator are the same entity or the content provider also operates the service.

6.4 Control Layer



This layer is responsible for a management of content distribution, content ingestion, reporting, logging, request routing function.

Every CDN have some capabilities for content delivery methods, content adaptation and content security. All these information are very important for content transfer from a source to the consumer.

- Delivery methods – by delivery methods, we understood different methods for content transfer from distribution node to end consumer.
- Content adaptation – is set of methods, which can be used to change original information to form, which is more useful for transfer over real-time channel.
- Content security – is set of rules and mechanisms for protection of content including access authorization, digital rights management, copy protection, watermarking, ...



It is very important to map, which area can be served from which distribution node or network. In IP network this is done by a routing table, where every item – subnet is described by a network address and network mask. In CDN networks the address and size is defined in one parameter called a footprint. The footprint can be potentially any information about coverage like geographical area [43], Internet provider [44], etc. There exists a footprint table which contains list of all footprints with their distribution nodes.

6.5 Distribution Layer



This layer is responsible for content delivery to the consumer. It consists of huge number of delivery nodes, which can be organized also in clusters. We can call these nodes or clusters as distribution points.

Every distribution point contains big storage capacity on local or external disk arrays. This capacity is used for the content storage.

File based content



File based content is sequence of bytes which have started and ended in the past.



File based content can be downloaded without QoS, but almost always requires zero tolerance to change or modification at any level of content.

This type of content is majority of all content delivered over the Internet and also over the CDNs. For this purpose the ideal transfer protocol is HTTP, which is widely used in current Internet.

Stream based content



A stream is sequence of bytes which started and ended in the past.



The stream based content contains useful information per parts, so it can be transferred continuously and during the transfer it can be also consumed. Streaming content is mostly sensitive to QoS of transfer channel, especially to delay and jitter.

Typical content of this type is a media stream. In contrast to data transfers in Internet the video and audio content is becoming more and more popular. CDN like part of Internet is helping to fulfill these requirements by spreading the content and the load over more location and distribution points.

Media consumers are sensitive to audio information, because every country is using different language. To save resources in the network it is effective to transfer only that audio stream which contains required language. To do this a distribution point can contain content with video stream and multiple audio streams. Only one audio stream (elementary stream) is transferred to consumer in a session according consumer set up. Every elementary stream can be transferred over different paths from source to destination.

The same concept can be also used for multiple angle content, where single content is recorded from more video cameras from different angles (views). Again only video is transferred from distribution point to consumer.

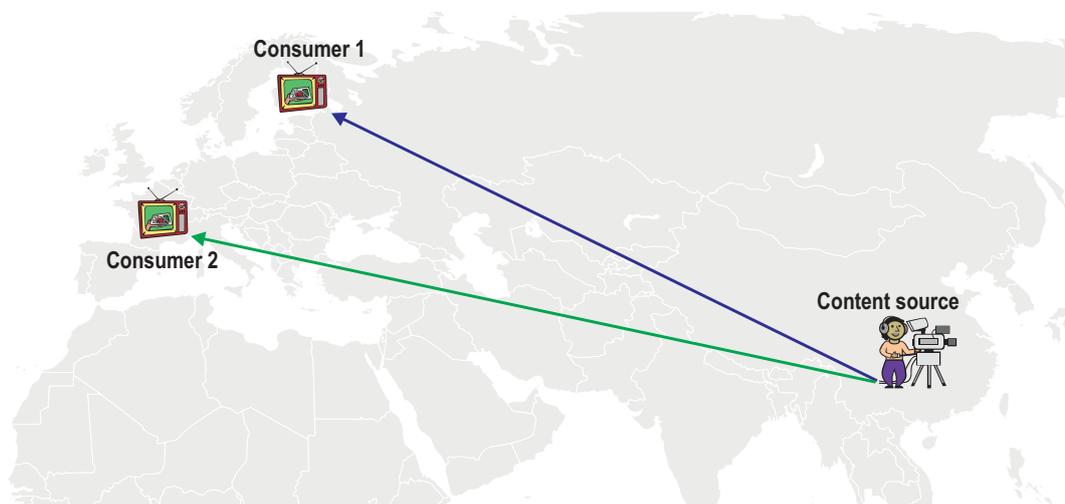
Live stream based content



This stream is a sequence of bytes which have started in the past and is not finished until now and continues.



Because this stream is live, it cannot be cached for further streaming and all the efficiency in content distribution is in single transfer over the same parts of network.



Live stream distribution without CDN and with CDN distribution





More clear explanation would be on simple example: imagine that two consumers in Europe want to see live video show from China Olympic Games. In a normal unicast network, which Internet is, the stream will be duplicated at China at a streaming platform and transferred to consumers in two separate streams (Figures above).

By redirection of consumers to CDN, CDN can join the video stream from China Streaming server, transfer it to its location in Europe (Figures above). The consumers then connect to CDN and stream is duplicated in CDN distribution point and transferred to the consumers in two separate streams, but only inside the European network. This concept can save Internet connectivity from Europe to China. In this case bandwidth is $1/2$ of previous concept. If there are three consumers of same content saving is $1/3$ of previous concept, etc.

6.6 Federated CDNs



It is very hard to deploy global international network, which is split over the whole globe. CDN benefits from huge number of delivered content (bytes). More bytes delivered by certain platform are increasing utilization of network; therefore the price of one transferred byte is lower. Lower price per byte brings more customers, which brings more number of content which is delivered to the customers. More content will consume more bytes for delivery. As we can see the pricing process is rounded in one loop without small external impact. This loop is “responsible” for creation of big CDN players, which are controlling the market.

Telecommunication operators are losing market share day by day and international global service providers (Akamai, Globix, Limelight ...) are overtaking the service market. Telecommunication operators are pushed to role of a network provider “cable owner”. This situation is unwanted for Telecommunication operator and therefore they are investing afford to value add services like CDNs.



Under CDN federation, there is huge number of initiatives driven by CDN owners and mostly by telecommunication operators [45], [46]. Federation is way of CDN interconnection, where content can be copied from one CDN to another. Then, the end user can be served by both CDNs by content originated network or by federated network.



Few federated networks act and look like one robust CDN from user perspective and from content provider perspective.

7 Optical Technologies

7.1 Optical Networks



A development of single-mode optical fibers with nearly unlimited bandwidth opens a massive development of long-haul and metropolitan optical networks of the point-to-point type. Using optical cables allows tremendous decreasing of costs to network facilities and maintenance and dramatically increases a quality of services. Many business companies have now an access to services providing through optical fibers. In spite of their advantages, optical cables were not very used in access networks – in a network segment that starts in the central point and finishes at subscribers. Because this segment was usually based on utilizing of metallic homogeneous lines, high-speed services available for residential subscribers and small businesses are limited by possibilities of xDSL and HFC technologies.

7.2 Optical Access Networks

- A main barrier of service provisioning by means of optical fibers directly to residential and business customers are costs of connecting subscribers to the central office. A large number of point-to-point connections could require many active components and a large number of optical fibers. This leads to enormous increasing of installation and maintenance costs. An attractive solution of these problems is the **FTTx** (*Fiber To The x*) architecture. The *Passive Optical Network* (**PON**) together with the FTTx allows some customers to share the same connection without any active components.
-



Optical access network architectures should be simple and the network should be simple from a viewpoint of the activity and services. It means that passive architectures without any switching and controlling are preferred to active architectures. Moreover, the *optical network unit* (**ONU**) should be very simple for decreasing of costs and for improving of the reliability. Components used in the ONU should be able to operate without any temperature controlling. These rules exclude a using of more sophisticated lasers and other optical components in the ONU. An equipment of the *optical line termination* (**OLT**) can be a little bit more sophisticated, because it is placed in the controlling environment and costs are amortized between many subscribers.



Optical networks suggested for this application are together called passive optical networks PON. They use a particular form of passive components as the remote node. Their main advantages are reliability, simplicity of maintenance and an absence of power supplies. The passive optical networks PON divide a signal from one central transmitter between various outgoing fibers, each forwarding to the separated receiver in the particular customer location. Separated receivers from customer locations send signals to the central distribution point, so users share a total transmission capacity of the system. A key advantage of this approach is using only passive components in the outdoor distribution network.

7.3 Optical Access Networks – PON Concept



The PON is essentially a two-way point-to-multipoint system [1], [47]. The downstream data signal originates at a central point with a single transmitter. Passive optical couplers divide this signal among output fibers that distribute the same signal to all customers. The receiver at the customer end selects only the data directed to that terminal, discarding data directed to other users. Thus the data stream from the transmitter is divided among users. Each customer terminal has its own transmitter, all of which can return upstream signals to the central distribution point. The upstream transmission may go through the same fibers as the downstream transmission, or through.

Because all the signals go back to a single receiver, separate time slots are assigned to the transmitters at each subscriber terminal so they don't interfere with each other. The feeder cable transmits optical signals between a central office and a splitter that allows a number of ONTs to be connected to the same feeder cable. The ONT is required for each subscriber and provides connections for various services. Because the one FTTx infrastructure usually provides a service for up to 32 subscribers, many of such networks are required for the community service provisioning. There exist different architectures for connecting of subscribers to the PON. The simplest one uses only one splitter, but in others also more splitters can be used.

Optical terminals



The central distribution terminal serves as the central controller for the passive optical network and provides an interface with the outside world. Many of these terminals can be assembled in one location, each serving its own group of subscriber terminals. The FSAN standard calls this controller and transmitter jointly as an optical line terminal OLT. Terminal at the subscriber end are called optical network terminals ONTs and they provide interfaces between the network and the subscriber's equipment. Losses inherent in the signal splitting limit one OLT to serving no more than maximum 32 ONTs.

The downstream transmission



The standard PON works at two or three wavelengths. The OLT includes a distributed-feedback laser transmitting downstream at 1550 nm, which couples more than a milliwatt into the output fiber. Each cell or packet in the downstream signal carries the address of its destination terminal. Passive splitters divide the light among all terminals, but each terminal only reads those packets addressed to it. The downstream data transmission also provides timing signals needed to control the upstream transmission. The OLTs can use relatively expensive 1550nm transmitters, because each PON requires only one of them. However, the PON requires many more ONTs, so they must be relatively inexpensive, and operate in the less-controllable environment of the customer site. This led to the

choice of lower-cost 1310 nm transmitters for the upstream channels. The FSAN standard also provides for a third wavelength channel at 1490 nm.

In a central office (a head end) for the PSTN and data networks, interfaces to the optical distribution network ODN through the optical line terminal OLT are located. The downstream 1490 nm and upstream 1310 nm wavelengths are used for the data and audio transmission. An optical transmitter of video signals converts signals of video services to the optical format at the 1550 nm wavelength. The **WDM** (*Wavelength-Division Multiplexing*) coupler and together transmitting by the downstream combine the 1550 nm and 1490 nm wavelengths. Summary, three wavelengths transmit different information signal simultaneously and in various directions over the same optical fiber. The transmission rates depend on the chosen applications, software allocates a transmission capacity to each terminal dynamically, and so it can be changed as necessary.

The upstream transmission



In the upstream transmission, the PON is the multipoint-to-point type. For avoiding of data collisions from different ONT signals incoming to a splitter in the same time, the TDMA approach is used. The TDMA can transmit data bursts from each ONT back to the OLT in a concrete, specified time. Each ONT transmission time slot is legitimate by the OLT so that packets from different ONT don't overlap with others. The upstream transmission goes through a network of fibers that are combined with passive couplers, so all transmitters send their signals to one receiver in the OLT. To keep these signals from interfering with each other, the PON uses a time-division multiple-access protocol TDMA that assigns different time slots to each ONT.

Each subscriber terminal turns on and transmits signals upstream during its assigned time slot, then switches off so the next can begin transmitting. The control software allocates these time slots and the downstream transmission provides the clock signals to synchronize the upstream transmission by all subscriber terminals.

The fiber architecture



All PONs utilize single-mode fibers with signals divided by splitters. The placement and number of splitters depend on a system design – 1x8, 1x4 and 1x8... The splitters are purely passive devices that require no power, so they can be placed in spliced cases or enclosures anywhere between the distribution center and the subscribers. The FSAN standard provides for both single- and dual-fiber systems and each have their attractions.

A single-fiber system reduces fiber costs by transmitting upstream and downstream signals through the same fibers. The trade-off is that it requires wavelength-division multiplexing (WDM) optics on both ends of the system. A dual-fiber (two-fiber) system is avoiding the added cost and complexity of a WDM optics, utilizes the ability to dedicate the first fiber to a downstream

distribution of analog video signals for a cable television and the second fiber to a digital transmission of audio, data and digital video signals.

FTTx Architecture



The PON technology can be involved in all architectures of the FTTx type that offer a mechanism for allowing of sufficient network bandwidth to deliver of new services and applications. The PON network can be common for all these architectures. A question is a placement of active electronic in the outdoor environment. Only in the case of the FTTH/B configuration, all active components are apart from the outdoor environment. The FTTCab and FTTC architectures require active electronics in the outside environment to be placed in a cabinet or in a curb [2].



Equipment at the central office are connected to the PSTN network, equipped by ATM or Ethernet interfaces, and connected to a cable interface or to a satellite receiver. All these signals are then combined for inputting onto the one fiber by using of WDM elements and transmitted to end customers through a passive optical splitter. A splitter ratio can be in a range up to 32 subscribers without using active components in a network. A signal is then delivered to the house over the individual optical fiber. In equipment at the users end, an optical signal is converted into an electrical form by using of OEC converters and at the same time the OEC transforms a signal in service demanding by end users. Ideally, the OEC should have standard user interfaces without requiring special set-top boxes. Main advantages of next FTTx architectures are above all facts that they are passive networks without any active components between a central office and users, they are using only one optical fiber per end user, they have local battery backups and low power consumption, they are reliable, scalable and secure.

Various implementations of the FTTx

Multimedia and Internet services drive the need for higher bit rates (some Mbps) to the home. To ensure that the loop plant of the future will be and remain bandwidth-scalable, fiber-optic cables are replacing metallic cables wherever possible.



Depending on where the fiber is terminated, this technology is known by different names such as:

- *Fiber-to-the-home (FTTH)* – if the fiber reaches the premises of the end user where it is also terminated
- *Fiber-to-the-curb (FTTC)* – the optical fiber is situated from the central office to the optical splitter and then to a small cabinet at the curb that is close approximately 200 m to the subscriber and where a signal is converted back to its electrical form.

- *Fiber-to-the-cabinet (FTTCab)* – the optical fiber is situated from the central office to the optical splitter and then to a cabinet at the neighborhood that is close around 1 km to the subscriber and where a signal is converted back to its electrical form for a signal transmission over metallic homogeneous lines to the subscriber.
 - *Fiber-to-the-desk (FTTD)* – if the fiber is terminated directly at the desk or even at the PC.
-

7.4 Metropolitan Optical Access Networks



Metropolitan network is important part of NGN. Such a network serves for concentration of traffic from several local networks and switches it into core network. Looking on contemporary networks, metropolitan networks are bottleneck. Traditionally, these networks are based on simple topologies, like circle or star. They deploy optical fiber in conjunction with WDM technology to provide sufficient transport speed.

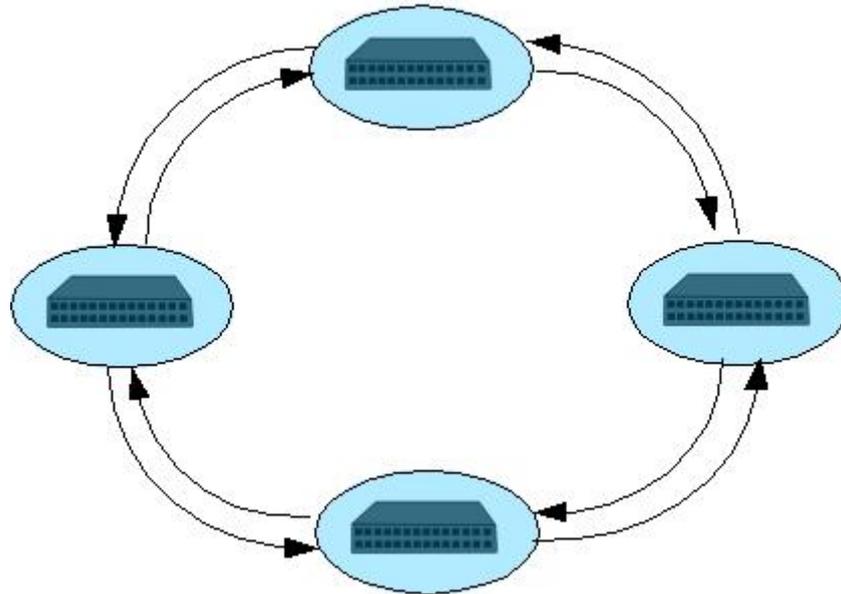
STARNET is one of the well-known examples of star topology. This network is based on broadcast of data from central point. Such data are sent and receiver selects only data addressed to it. Ring based networks are even more interesting, particularly **RPR** (*Resilient Packet Ring*), which has biggest potential of application, are certainly most important.

RPR

RPR has been proposed to respond to the problems of both SONET and Ethernet in MAN environment. Resilient Packet Ring (RPR) is the technology proposed for packet oriented MAN network established over ring architecture using optical fiber. RPR has been standardized by IEEE as 802.17 standard [48], [49].



RPR architecture (Figure below) is composed of two rings oriented in different directions. Advantage is that each node can communicate with both neighbors directly without necessity for whole ring way. Each packet controls traffic passing through the node. In the case, when the packet is aimed at it, the node drops it from the ring. Otherwise, it passes transparently through this node. In the case, when the node wants to add a new packet to the ring, it can do that only in the case, if there is enough long gap on the ring. The gap can be created also by dropped packet.



RPR architecture

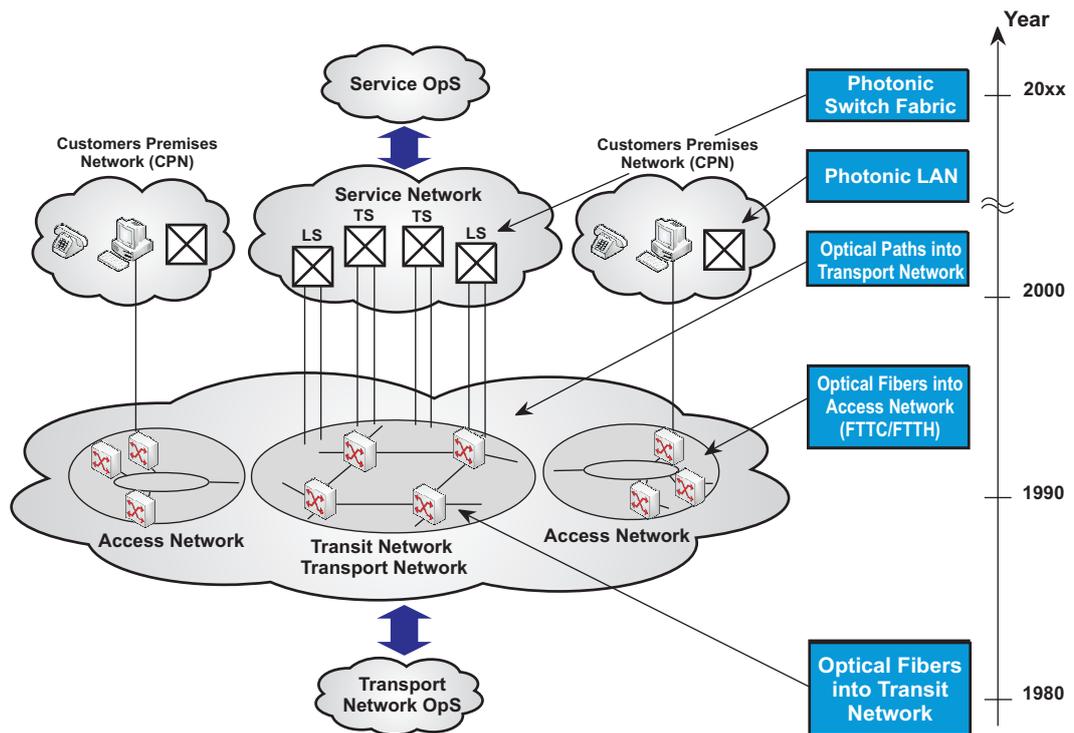


RPR can be characterized by following properties:

- **ADM (Add Drop Multiplexer)** architecture. Each RPR equipment behaves like add/drop member, what means that it can add and also drop packets to the ring.
 - Universality of physical layer. RPR defines only 2nd layer of ISO OSI model. The physical layer is not defined within the standard, what allows possible interoperability with other solutions like Ethernet, SONET, or DWDM.
 - Resiliency – thanks to the deployed technologies, RPR reaches extremely low protection times (lower than 50ms). In this case, the packets are routed in the opposite direction. In such case, the packets reach their destination.
 - Equity in bandwidth distribution. RPR contains algorithms regulating bandwidth use and medium access.
 - Broadcast and multicast. Ring topologies are ideal for multicast and broadcast implementation.
 - Service provisioning without extensive configuration. Comparing to SONET no pre-configured circuits are necessary.
-

7.5 Optical Transport Networks

On present days, optical technologies utilizing optical transmission media dominate in the transport level of telecommunication networks. Except an evolution of low-loss optical fibers, various optical components needed to realize optical-fiber transmission systems have been developed. The penetration of optical technologies into the various network levels is schematically illustrated in Figure below.



Penetration of photonic technologies into network levels



Optical technologies were first applied to transport networks, because these technologies are best suited to this application, which requires long-distance and wide-bandwidth transmission. Moreover, costs of the system are shared by many users (connections). Thus, the transport transmission cost per bit is much lower with optical media than with metallic media. Primary applications of optical technologies realize only point-to-point connection [50], [51]. However, to create large-bandwidth end-to-end connections cost-effectively, it is essential to reduce not only a transmission cost, but also a transport node cost. A key role in reducing both costs is playing by optical technologies that can be developed also in local area networks, in optical networks of the type Ethernet, FDDI or Fiber Channel. From first introduced optical-fiber transmission systems in 1981, a transmission capacity has been increased more than one order of magnitude per decade. This has resulted in a more than 90% reduction on a cost in this decade. The continued enhancement of electronic devices has also driven research into a high-speed transmission, so a rapid progress to multi-Gbps transmission systems is allowed. Another technology that must be emphasized is a technology of optical amplifiers.

Their advantages including a wide bandwidth, a low noise, a high gain/power ratio and an easy connection to single-mode fibers greatly increase an application range of optical transmissions by overcoming the optical loss of signal transmission, devices and optical components. Furthermore, new optical technologies – a soliton transmission, a dense wavelength-division multiplexing DWDM, an optical frequency-division multiplexing OFDM and a high-speed optical signal processing – have the potential to enable further expansion of the transmission capacity and the network flexibility.

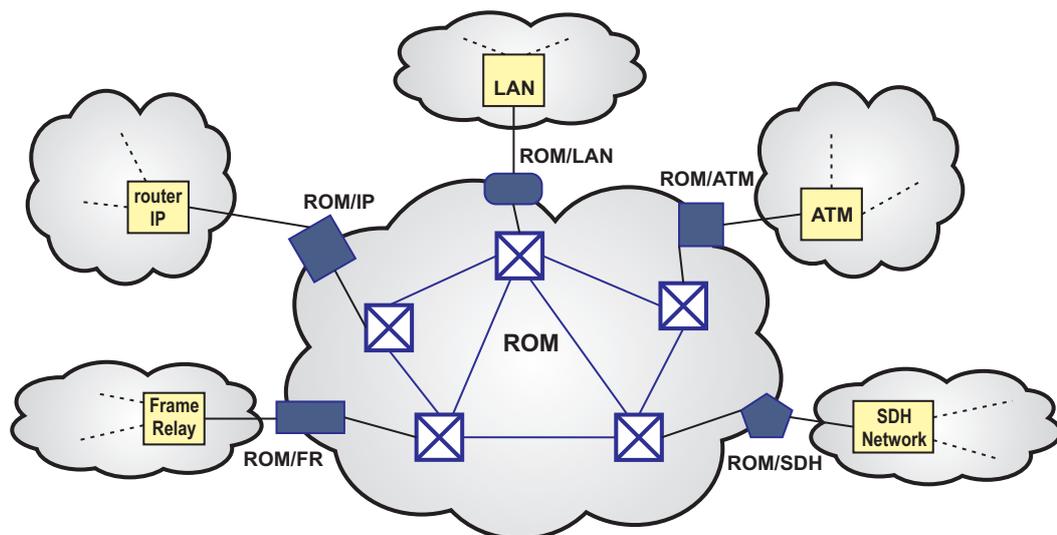
1G and 2G optical networks



In first-generation networks, the electronics at a node must not only handle all the data intended for that node, but also all the data that is being passed through that node on to other nodes in the network. If the latter data could be routed through in the optical domain, the burden on the underlying electronics at the node would be significantly reduced. This is one of key drivers for second-generation optical networks. Examples of these networks are the OTDM and WDM networks. WDM networks are expected to be deployed in the next few years, not only in interexchange networks and undersea networks, but also in local-exchange and access networks. OTDM networks constitute a longer-term approach.

7.6 All-optical Transport Network Architectures

Previously we have discussed that current transport systems are coming to their limits. New perspectives are focusing mainly on optical packet switching networks, which integrate high bandwidth availability and ability of switching in transport layer [52]. While proposed technology isn't mature yet, intermediate step between current networks and all-optical networks has been proposed. This step combines advantages of optical switching and electronic memories and will be discussed later. All-optical network should deploy WDM multiplex on physical layer (Figure below).



Model of optical multiservice network (ROM)

- To reflect described drawbacks of the optical technology, special structure of optical switch has been proposed. The outgoing port of the switching matrix chooses which packets are sent to the optical fiber. Therefore, it receives packets from all delay lines and using filters, it chooses right packets. The number of packets is equal or inferior to the number of wavelengths. This choice is made by control unit, which see global state of the node. Proposed switching matrix is general and can be used in various ways. The next step is to parameterize it, mainly the delay lines.



The most important is the length of one delay line. This parameter can be different in synchronous and in asynchronous networks. A synchronous network requires this value to be set equal to the length of optical packet. On the contrary, an asynchronous network sets this constant to the inferior values so that it is possible to order packets one after another.

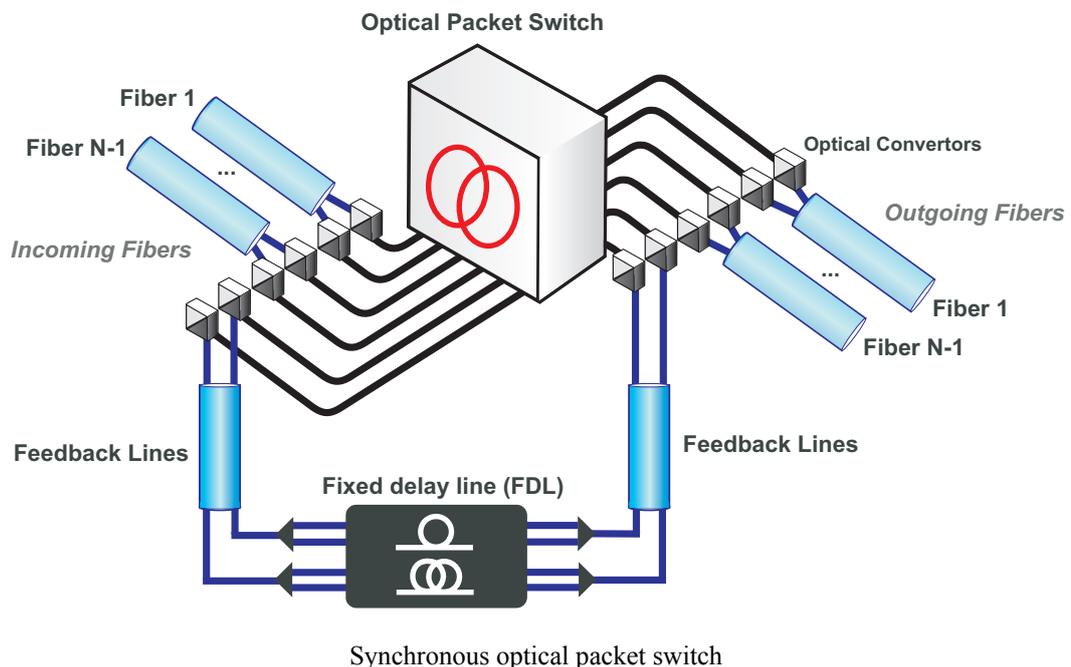


As it was said previously, two approaches exist in optical packet switching: synchronous and asynchronous switching. It is obvious that they are different in their approach to the switching in each node. Synchronous switching takes advantage from switching of constant length packets, which are synchronized in the nodes. An asynchronous network supports packets of different lengths and these are switched without synchronization. In the next subsections, both approaches are examined in details.

7.7 Synchronous Optical Network



As we have stated above, synchronous optical packet switching network is based on switching of constant length optical packets. The architecture of such network should take into account this fact. Ingress node into proposed network transforms incoming flows into optical packets with format depicted. Optical packet is composed of header and of load. Header and payload are separated by guard time fields. These fields allow us secure delimitation of the header from payload and payload from next header. The beginning of both header and load is composed of synchronization bits used for synchronization in nodes.

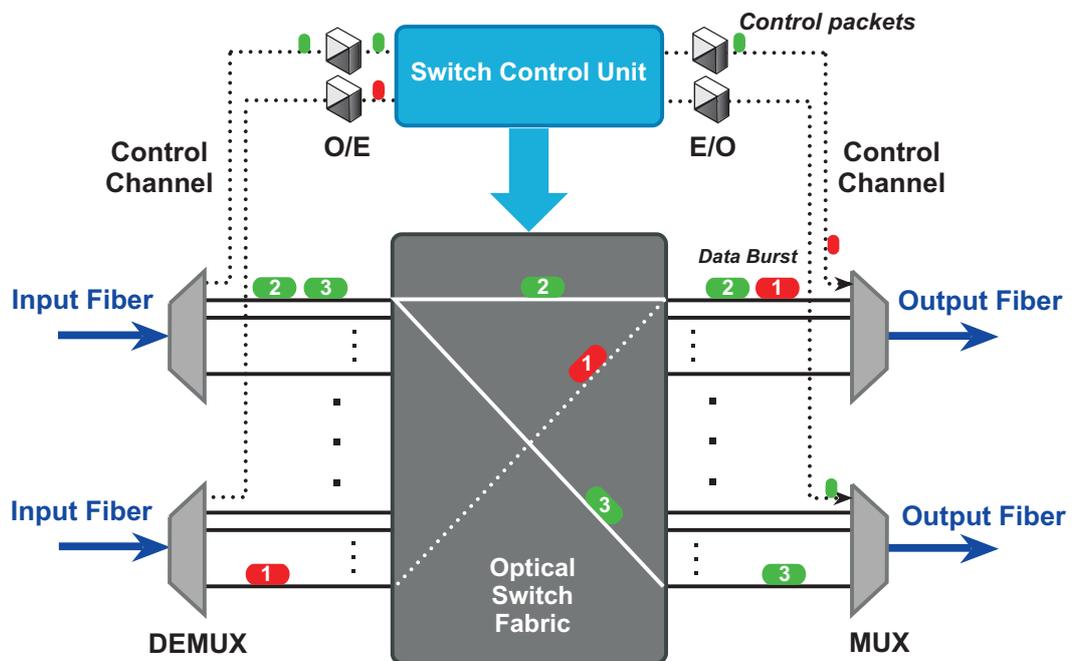


Header contains routing information and is 64 ns long. Guard times are 50 ns long each, yielding in 100 ns per packet. The rest of the packet is composed of data. Switching node is depicted in Figure above. Optical packets enter to the node and are firstly synchronized. Header is afterwards delimited from payload. Payload enters delay lines, where it waits until the header is processed and switching matrix is parameterized. Afterwards, payloads enter the switching matrix and are routed to the outgoing port. Before sending packets, new header is generated and concatenated to the header to the beginning of the optical packet. Optical packets are of the fixed length and therefore they are transported within fixed length time-slots. In the same time, it is impossible to guarantee that packets enter the node synchronized. Delay variation is affected by many variables, and the most important is the temperature. Therefore, there is a need for synchronizations. Synchronization can be divided into two groups: coarse and fine. Another important issue is the performance of optical switching matrix, when there are no optical memories. To improve this parameter, recirculating lines are used. Recirculating lines are intended for packets, which do not succeed to find free

outgoing port. Such packets enter recirculating lines and they appear at the node entry several time-slots later.

7.8 Burst Switched Network

Alternative to the synchronous network is burst switched network. This approach is similar to contemporary Internet, where packets are of various lengths. Therefore, they can enter the node anytime and there is no necessity for synchronization.



Burst switch architecture

Comparing to synchronous switching, burst switching introduces more flexibility but also new problem. This problem is temporal concurrence, which should be resolved.



In synchronous networks, only spatial (which fiber) and spectral (which lambda) problems were to be solved in each time-slot. In burst network, there is no notion of time-slot and consequently packets should be ordered one after another. Even more, packets are of variable length. In the situation when one packet is blocking outgoing port, another one should wait in the delay lines. From this point of view, delay lines should be significantly smaller than mean packet length. Several enhancements have been proposed to improve performance of burst network. Majority of them consists in introducing new protocol for path reservation.

7.9 The WDM Technologies



With progresses in lasers and optical-electrical equipment technologies, it is possible to transmit one or more wavelengths on the same fiber. This is known as the wavelength-division multiplexing WDM. By adding of wavelengths into the same fiber, a transmission capacity and a bandwidth of the optical fiber are increasing and, therefore, a need for installation of additional optical fibers is decreased. In WDM systems, each used wavelength presents an independent channel. From a viewpoint of the WDM deployment in various network types and from a viewpoint of the channel spacing, three basic WDM network type are distinguished – long-haul, metropolitan, access [53].



Early *broadband WDM (BWDM)* systems operated with an overall broad channel spacing, because two wavelengths were separated and were located in two different transmission windows – at the 950 and 1300 nm or at the 1300 and 1550 nm. Latter *wide WDM (WWDM)* systems utilized more wavelengths in the transmission window and optical channels were usually separated a few of nm – the 1275.7, 1300.2, 1324.7 and 1349.2 nm. In present days, these systems can be employed in the PON networks with three wavelengths – the 1310, 1490 and 1550 nm. The most preferable recent *dense WDM (DWDM)* systems have a channel spacing usually not more than couple of nm in a range 1530 – 1625 nm of the erbium-fiber transmission window and use necessarily cooled lasers to prevent the wavelengths from drifting outside this window or from interfering with each other [54]. Latest *coarse WDM (CWDM)* systems utilize optical channels extended in a range 1270 – 1610 nm with a large channel spacing of 20 nm and therefore they use uncooled (non-thermally controlled) lasers.

Both – CWDM and DWDM – systems are the WDM types: the dense WDM is an implementation in long-haul networks distances and the coarse WDM is an implementation in metropolitan and access networks. Different demands for these two implementations require different architectures and determine performance demands for system components. The goal of DWDM systems is to maximize a bypassed distance without an electrical regeneration at amplifier costs spread over a maximum number of wavelengths. The goal of CWDM systems is minimize a components cost in the system, where a distance is shorter and amplifiers are not necessary.

DWDM systems – wavelengths in the 1530 – 1625 nm range; expensive cooled lasers to prevent wavelengths from drifting outside this region and from interfering with each other. Increasing of the channel number requires a narrowing of the channel spacing in filters, an extended spacing and a using of translators for reaching narrower channel spacing and an opening of the new spectral area – except a common C-band also a new L-band.

CWDM systems – wavelengths in the entire 1280 – 1625 nm band; less expensive uncooled lasers (cost savings are a direct reflection of the packaging differences between DWDM and CWDM lasers). Medium distances without amplifiers, an elimination of the EDFA amplifier bandwidth limit allows distributing of

wavelengths over a broad region and locating of them sufficiently far between, a need for cheap multiplexers, demultiplexers, add/drop and switches (not simply retrofitted from DWDM systems) with a low loss, a high isolation and proper channel spacing.

The DWDM and CWDM systems have different sources operating at given wavelengths and have different filter for combining of wavelengths onto the same fiber at the transmitting end and for separating of wavelengths at the receiving end. However, technologies of used filter can be the same. Also, the adding and the dropping of wavelengths in midpoints of the system can be performed by the same technology for both systems. A main difference is in channel spacing – the DWDM channel spacing can be nearly 0.2 nm, the CWDM channel spacing is usually 20 nm. Therefore, it is possible to consider about a practical utilization of the CWDM/DWDM combination.

The CWDM is an alternative to expensive and complex DWDM-based architectures because it provides an occasion to continue in a direction given by the DWDM technology to all optical networks. An advantage of the DWDM in eliminating of expensive regenerators is not used in metropolitan networks, where optical amplifiers are not required or where block modules with cheap uncooled laser pumps easy satisfy demands on distances in a majority of metropolitan circuits and paths.

The CWDM varies from the DWDM in broader optical channel spacing between light sources that are multiplexed into the same fiber. As well, CWDM transmitters/receivers use an optical multiplexing technique for reaching serial equivalent data rates, whereas the DWDM multiplexes many serial data streams for reaching a bandwidth up to hundreds of Gbps. This is reached by using of a thermal control in the DWDM channel spacing. This exact spacing control allows combining a large number of separated channels. The typical CWDM system has spacing in order of some nm (some THz) and doesn't require a thermal control. So, CWDM transmitters/receivers without a thermal control have directly modulated lasers and include lower-speed components for reaching of higher data rates.

7.10 The WDM Architectures



WDM network architectures [53] can be classified into two broad categories: broadband and select architectures and wavelength routing architectures.



A broadband and select (B&S) WDM network has different nodes transmitting at different wavelengths. Their signals are broadcasting by a passive device in the middle of the network to all the nodes. In this case, this device is a passive optical star coupler. The coupler combines signals from all nodes and delivers a fraction of the power from each signal on to each output port. Each node employs a tunable optical filter to select the desired wavelength for reception.

This form of a network is simple and suitable for use in local- or metropolitan-area networks, such as access networks. The number of nodes in these networks is limited because the wavelengths cannot be reused in the network and because the transmitted power from a node must be split among all the receivers in the network.

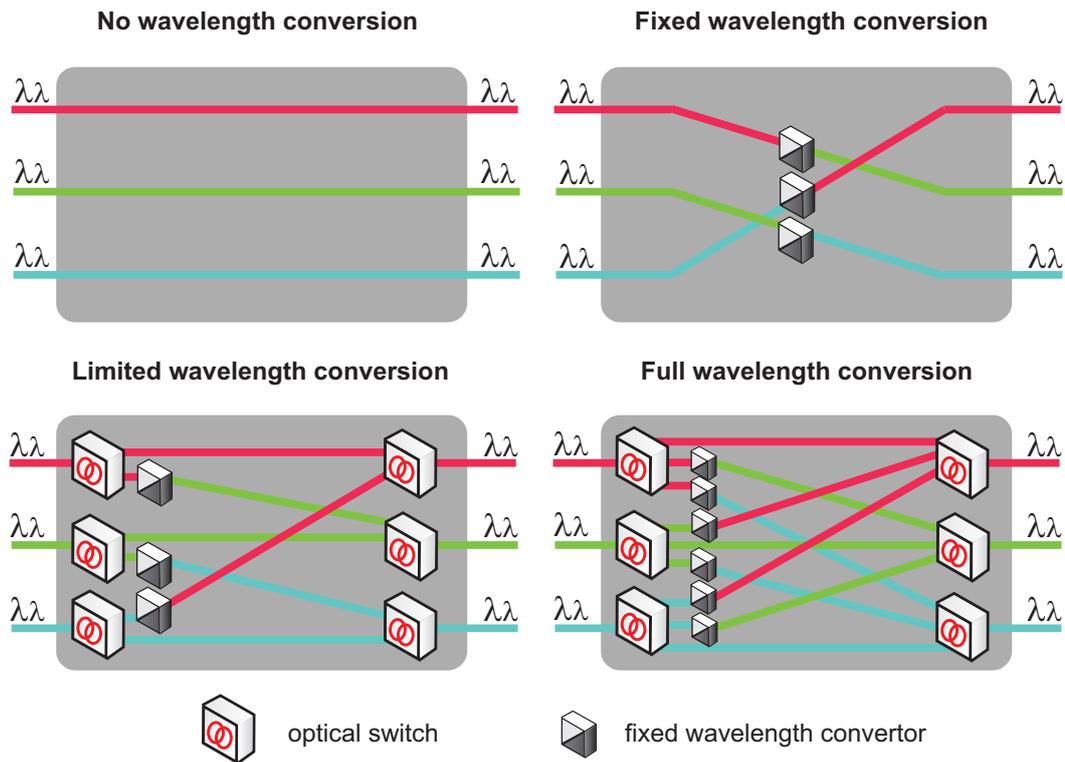


A more sophisticated and practical architecture today is that employs a wavelength routing. The nodes in the *wavelength routing (WR)* WDM network are capable of routing different wavelengths at an input port to different output ports. This enables us to set up many simultaneous light paths using the same wavelength in the network; that is, the capacity can be reused spatially. These light paths all use the same wavelength on every link in their path. This is a constraint that we must deal with if we do not have wavelength conversion capabilities within the network. This architecture also avoids broadcasting the power to unwanted receivers in the network. Thus these networks are suitable for deployment in metropolitan- and wide-area networks, such as local-exchange and interexchange networks.

Ideally, all the functions inside the node would be performed in an optical domain, but in practice, certain functions, such as a processing the header and a controlling the switch, get relegated to an electronic domain. This is because of the very limited processing capabilities in an optical domain. The header itself could be sent at a lower bit rate than the data so that it can be processed electronically.

7.11 Development Trends in Transport WAN

Contemporary technologies are coming to their limits in bandwidth provisioning. Therefore, evolution trends in transport networks use more intensive new technologies deployment. These new technologies concern mainly optical networks latest research: WDM circuit switching networks and optical packet networks.



Various levels of wavelengths conversion

Few decades ago, metallic cables were used for long distance data transport. They were replaced by optical fibers characterized by better quality parameters (error ratio, attenuation and higher bandwidth). Results of the research allowed even more efficient bandwidth use thanks to WDM multiplex.



WDM means that several wavelengths can be transported in parallel via one optical fiber. It is similar to frequency multiplex in classical networks. WDM is based on parallel transport of data on different wavelengths using one fiber. Therefore, wavelength can be used as supplementary information concerning packet routing. Data can be transported using one wavelength between end-points, but it is possible to use wavelength conversion in the nodes. Level of wavelength conversion in the nodes expresses also level of flexibility of WDM networks. It is possible to have WDM network without possibility of wavelength conversion, partial conversion and full conversion (Figure above). Event more, conversion can be implemented only in some nodes. Such approach leads us to serious problem, which is known as **RWA** (*Route Wavelength Assignment*). This problem is formulated how to optimize wavelength use between points in the network, which

topology is known to maximize transported data and minimize PLR. Well-resolved RWA problem can also help with network topology design by minimizing degree of conversion. Let us stress that with increasing level of conversion in the network also management complexity and price increases. WDM is relatively new technology, but researchers are already preparing next step, which is optical packet switching network. In this case, we are talking about all-optical network, where data are transported via proposed network without being converted into electrical form.

Optical packets are created at the ingress node and original electronic packets are formed at egress node of the optical network. As we show later, all-optical packet technology is too revolutionary. Therefore, an intermediate step has been proposed. This step uses optical switching matrix, but includes electronic memories.