VŠB – Technical University of Ostrava
Faculty of Electrical Engineering and Computer Science
Department of Telecommunications

Measurement of Triple Play Services in Hybrid Network

Měření Triple play služeb v hybridní síti

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Bc. Tomáš Putna

VŠB - Technical University of Ostrava
Faculty of Electrical Engineering and Computer Science
Department of Telecommunications

Diploma Thesis Assignment

Student: **Bc. Tomáš Putna**

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Title: Measurement of Triple Play Services in Hybrid Network
Měření Triple play služeb v hybridní síti

The thesis language: English

Description:

Student task, in this work, will be to perform Triple play services measurement in hetero-structure network consisting of GEPON Allied Telesis iMAP 9102 and xDSL technologies (ZyXEL IES-1000 či ZyXEL IES-5005). On the build-up network will be measured typical optical parameters (attenuation, power levels, dispersion, power losses, etc.) From the metallic network part point of view will be then measured (resistance, capacity, etc.). During measurement will be changed length of the optical route as well as metallic route by inserting additional optical link or in case of metallic part by line simulator (Spirent DLS-6900). To the created hybrid network will be connected VLC/IP-DVB streamer, which will be source of video streams. VoIP exchange will be used for testing of voice streams and the last service will be data connectivity. Altogether then all services will be customer oriented services via Triple play. Student will be also performing quality analysis of all services with the help of measuring devices. His task will be to measure network parameters according to ITU-T Y.1564, RFC 2544, etc. on the given hybrid network.

1. Description of access and hybrid xPON networks.
2. Description of xDSL technology.
3. Creation of experimental hybrid xPON/xDSL network workplace for testing of Triple play customer services.
4. Evaluate obtained results from measurements.

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Extent and terms of a thesis are specified in directions for its elaboration that are opened to the public on the web sites of the faculty.

Supervisor: **Ing. Jan Látal, Ph.D.**

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VYSOKÁ ŠKOLA BÁŇSKÁ-TECHNICKÁ UNIVERZITA
Fakulta elektrotechniky a informatiky
1D16
OSTRAVA

doc. Ing. Miroslav Vozňák, Ph.D.
Head of Department

prof. Ing. Pavel Brandštetter, CSc.
Dean

I hereby declare that this master's thesis was written by myself. I have quoted all the references I have drawn upon.

Ostrava, 29. April 2018 ....

I would like to thank Ing. Jan Látal, Ph.D. for technical assistance and consultation in the creation of this master's thesis.

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Abstract

The master's thesis deals with a project regarding the implementation, design and the quality of IPTV, VoIP and Data services within the Triple Play services. In heterostructural networks made up of GEPON and xDSL technologies. Different lengths of the optical and metallic paths were used for the measurements.

The first part of the thesis is theoretically analyzed the development and trend of optical and metallic networks.

The second part deals with the measurement of typical optical and metallic parameters on the constructed experimental network, where its integrity was tested.

Another part of the thesis is the evaluation of Triple play results, regarding the test where the network was variously tasked/burdened with data traffic and evaluated according to defined standards.

The last part is concerned with the Optiwave Software simulation environment.

Key Words: Data, EtherSam, IPTV, OAN, Optiwave, RFC 2455, RFC 6349, Triple Play, VoIP, xDSL, xPON.

Abstrakt

Diplomová práce se zabývá návrhem, realizací a kvalitou služeb IPTV, VoIP a Data v rámci Triple play služeb v heterostrukturní sítí tvořené GEPON a xDSL technologiemi. Pro měření byli využity různé délky optické a metalické trasy.

První části diplomové práce je teoreticky rozebrán vývoj a trend optických a metalických sítí.

Druhá část se zaměřuje na měření typických optických a metalických parametrů na vybudované experimentální síti, kde byla následně testována její integrita.

Dalším bodem práce je vyhodnocení výsledků Triple play, kde síť je různě zatěžována datovým provozem a následně vyhodnocována podle definovaných norem.

Závěr práce je věnovaný simulačnímu prostředí Optiwave.

Klíčová slova: Data, EtherSam, IPTV, OAN, Optiwave, RFC 2455, RFC 6349, Triple Play, VoIP, xDSL, xPON.

Contents

List of symbols and abbreviations

ACR	–	Absolute Category Rating
ADSL	–	Asymmetric digital subscriber line
AON	–	Active Optical Network
APC	–	Angled Physical Contact
APON	–	ATM PON
ATM	–	Asynchronous Transfer Mode
BGP	–	Border Gateway Protocol
BRI	–	Basic Rate ISDN
CB	–	Conduction Band
CIR	–	Committed Information Rate
CQ	–	Conversational Quality
CRC	–	Cyclic redundancy check
CSMA	–	Carrier sense Multiple Access
CSRC	–	Contributing source
DCR	–	Degradation Category Rating
DNS	–	Domain Name System
Downstream	–	Data transfer from the public network/Internet
DSCQS	–	Double Stimulus Continuous Quality Scale
DSL	–	Digital Subscriber Line
DVB	–	Digital Video Broadcasting
DVB-S	–	Digital Video Broadcasting-Satellite
DVB-T	–	Digital Video Broadcasting-Terrestrial
EIGRP	–	Enhanced Interior Gateway Routing Protocol
EPG	–	Electronic Programming Guide
EPON	–	Ethernet Passive Optical Network
EtherSAM	–	Ethernet Service Activation Methodology
ETSI	–	European Telecommunications Standards Institute
FCS	–	Frame check sequence
FSAN	–	Full service access network
FTB	–	Fused Bionic Taper
FTTB	–	Fiber to the Building
FTTC	–	Fiber to the Curb
FTTH	–	Fiber to the Home
FTTN	–	Fiber to the Node
GEM	–	GPON Encapsulation Method
GEPON	–	Gigabite Ethernet Passive Optical Network

GPON	–	Gigabitet Passive Optical Network
HDSL	–	High speed digital subscriber line
HDTV	–	High Definition TeleVision
IDSL	–	ISDN Digital Subscriber Line
IEC	–	International Electrotechnical Commission
IETF	–	Internet Engineering Task Force
IGMP	–	Internet Group Management Protokol multicast
IP	–	Internet protocol
IPTV	–	Internet Protocol Television
ISDN	–	Integrated Services Digital Network
ISO	–	International Organization for Standardization
ITA	–	Industry Telecommunications Association
ITU	–	International Telecommunication Union
ITU-T	–	ITU- Telecommunication Standardization Sector
LLID	–	Logical Line Identifier
LQ	–	Listening Quality
LTE	–	Long Term Evolution
MAC	–	Media Access Control
MHP	–	Multimedia Home Platform
MOS	–	Mean Opinion Score
MPCP	–	Multi-Point Control Protocol
MPEG	–	Moving Picture Experts Group
MPLS	–	Multiprotocol Label Switching
MSE	–	Mean Squared Error
MTU	–	Maximum Transmission Unit
NGA	–	Next Generation Access
OAN	–	Optical Access Network
ODN	–	Optical Distribution Network
OLT	–	Optical Line Terminal
ONT	–	Optical Network Termination
ONU	–	Optical Network Unit
ONU	–	Optical Network Unit
OSPF	–	Open Shortest Path First
P2MP	–	Point to Multi Point
P2P	–	Point To Point
PAM	–	Pulse Amplitude Modulation
PAMS	–	Perceptual analysis measurement system
PCM	–	Pulse Code Modulation
PDU	–	Protovol Data Unit

PES	–	Packetized Elementary Stream
PESQ	–	Perceptual Evaluation of Speech Quality
PIM	–	Protocol Independent Multicast
PIM-DM	–	PIM Dense Mode
PIM-SM	–	PIM Sparse Mode
PIN	–	Positive IntrinsicNegative
PLC	–	Planar Lightwave Circuit
PMD	–	Physical Medium-Dependent
PON	–	Passive Optical Network
PPV	–	Pay Per View
PSNR	–	Peak Signal to Noise Ratio
PSQM	–	Personal Speech Quality Measure
PSTN	–	Public Switche Telephone Network
QoE	–	Quality of Experience
QOS	–	Quality of service
QRV	–	Queriers Robustness Variable
RADSL	–	Rate Adaptive DSL
RFC	–	Request for Comments
RTCP	–	Real Time Control Protocol
RTP	–	Real-time Transport Protocol
RTPC	–	Real-time Transport Control Protocol
RTPS	–	Real Time Streaming Protoco
RTSP	–	Real Time Streaming Protocol
RTT	–	Round Trip Time
SDSL	–	Symmetric digital subscriber line
SFD	–	Start of Frame Delimiter
SMPTE	–	Society of Motion Picture and Television Engineers
SOA	–	Semiconductor Optical Amplifier
SSRC	–	Synchronization source identifier
TC	–	Trellis Coded
TCP	–	Transmission Control Protocol
TDM	–	Time Division Multiplex
TS	–	Time Stamp
UDP	–	User Datagram Protocol
Upstream	–	Data transfer from the user to the public
VCR	–	Video Cassette Recorder
VDSL	–	Very High Bit Rate Digital Subcriber Line
VLAN	–	Virtual Local Area Network
VLC	–	Variable Lenght Coding

VoD	–	Video on Demand
VoIP	–	Voice over IP
WDM	–	Wavelength Division Multiplexing
xDSL	–	Generic term for any type of DSL

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1 Introduction

The thesis is divided into a theoretical and a practical part. The theoretical part provides an introduction to the basic principles of the used technologies, and subsequently describes the used equipment. The practical part implements a complete Triple Play service within the infrastructure created by the hybrid xPON/xDSL network. The thesis focuses on the integrity verification and the deployment of these multimedia services on the hybrid access network. Furthermore, the issue of the optical network is introduced into the Optiwave Software simulation environment.

Much of the work is devoted to the measuring of Triple Play services in the hybrid network. Measurement takes place in both, optical and metallic parts. All services are measured on a complete constructed route and are configured to represent a real traffic in the production version. The RFC 2544, RFC 6349 and EtherSAM (ITU-T Y.1564) tests are used.

The Triple play services are analyzed by objective methods. Audiovisual service is evaluated using MSU Video Measurement Quality Tool based on objective MSE, PSNR, and SSIM methods for multicast traffic. For the video analysis, the AXS-200/625 measuring device of bit rate and the loss of video packets was used? The voice service is analyzed using IxChariot to monitor MOS and R-factor values. The data service is analyzed using the BWMeter software, which allows downstream and upstream data traffic to be monitored. All services are analyzed for distinct combinations of lengths of optical and metallic paths.

The last chapter of the practical part is focused on the creating of an optical network in the simulation environment of Optiwave OptiSystem based on measured real-time data while monitoring the following parameters: bit error rate, Q-factor, eye diagram and spectral analysis.

The aim of the thesis is to build correctly the communication of Triple Play services and to describe the individual sections both practically and theoretically, and to measure Triple Play services and everything related to these technologies on the hybrid network made up of GEPON Allied Telesis iMAP 9102 and xDSL.

The practical measurements were carried out at the new FEI building in internal laboratories. Concrete EB315, EB316, EB211 and EB215.

2 Technologies in access networks

In this chapter, describe some of the technologies and basic functional units that are located on access networks.

2.1 Introduction xPON technology

It is a member of the Ethernet family. Work on EPON started in March 2001 by the study group IEEE 802.3ah and was completed in June 2004. Ethernet covers the physical and data layer of the OSI Reference Model.

EPON belongs to the PON group of access networks. This is a ODN containing passive transmission parts (optical fibers, connectors, welds, connectors, passive optical hubs and filters) OLT, ONT and ONU. From a topology point of view, the tree structure is the most common way.

EPON layers are very similar to traditional Ethernet point-point. The Ethernet standard further divides the OSI physical layer and data layer into multiple sub layers. The optional Ethernet point-to-point access control access sub layer is replaced with a point-to-multi-point layer.

The MPMC layer uses a multi-point control log to control access to a shared PON medium among all ONUs. Although the OLT and ONU layers look almost the same, the MPMC entity is referred to as master in OLT, and on the other hand, it acts as a slave in ONU.

The basic comparison of the APON/BPON, GPON, and EPON optical networks can be seen from the table 1.

Table 1: Comparison of individual PON variants [1]

PON	**APON/BPON**	**GPON**	**EPON (type 2)**
Standard	ITU-T G.983	ITU-T G.984	IEEE 802.3ah
Transmission speed- Download	155,52 or 622,08 Mbit/s	1,244 or 2,488 Gbit/s	1,25 Gbit/s
Transmission speed- Upload	155,52 or 622,08 Mbit/s	1,244 or 2,488 Gbit/s	1,25 Gbit/s
Wavelength - Downstream	1480-1500 nm	1480-1500 nm	1490+-10 nm
Wavelength – Upstream	1260-1360 nm	1260-1360 nm	1310+-50 nm
2 Layer protocol	ATM	ATM, GEM	Ethernet
max. connected user	32	64 (prospectively 128)	32
logical/physical range network [km]	20/20	60/20	20/20

From a comparison, it can be seen that the EPON variant offers 1.25 Gbit/s symmetric bit rates in both directions (speed on the physical layer). By comparing GPON, it is possible to connect a smaller number of end-users thanks to a lower bridging attenuation [1] [2].

2.1.1 EPON Architecture

Ethernet includes the physical and line layer of RM-OSI. Fig. 1 shows a comparison of the architecture of the traditional point-to-point (P2P) and point-to-multipoint (P2MP) EPON connections.

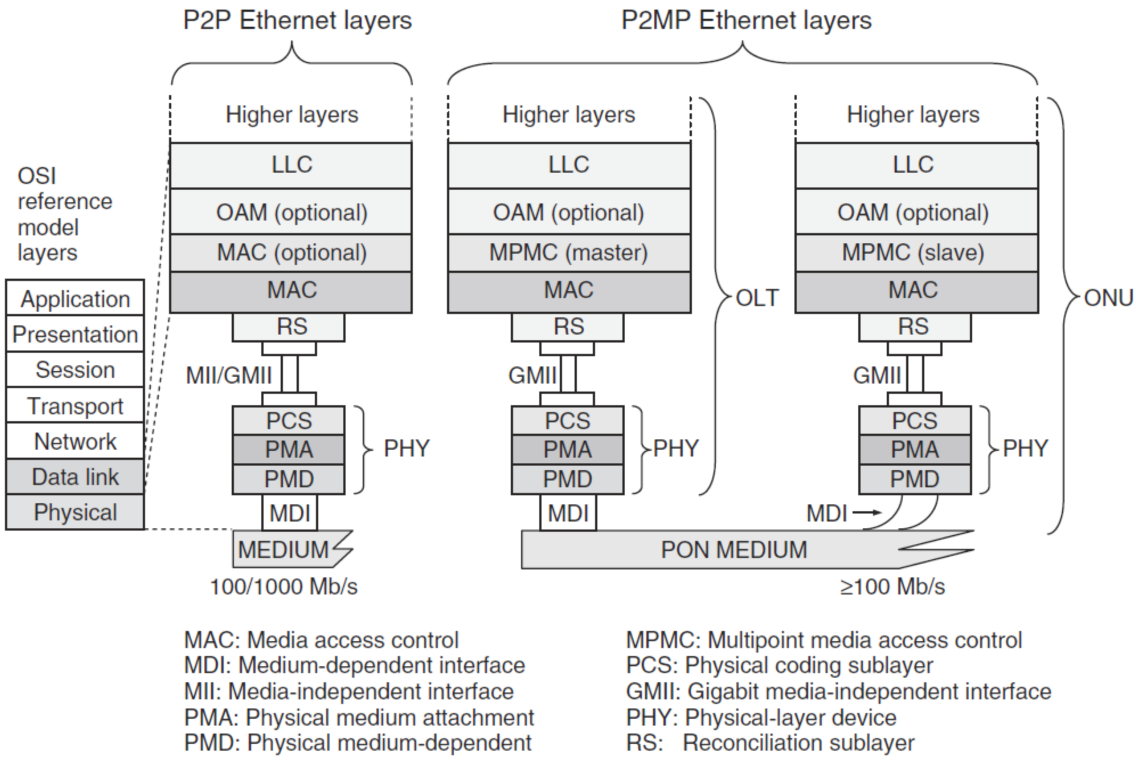

Figure 1: P2P Ethernet and P2MP EPON layering architecture. [81]

As can be seen in fig. 1, the traditional P2P Ethernet and EPON architecturally very similar. The optional MAC sub layers in the P2P Ethernet are replaced by a mandatory multi point MAC MPMC (Multi point Media Access Control) sublayer. This Multi point Control Protocol MPCP co-ordinates access to the PON shared media between ONU. Although the OLT and ONU Ethernet architecture is almost identical, MPCP on OLT works as a master and ONU/ONT drive as slave [71].

2.1.2 Ethernet Frame

Ethernet PON is a passive optical network that transmits data in the form of Ethernet frame defined according to IEEE802.3. The construction of the Ethernet frame is described on figure 2.

The frame begins with a preamble and one SFD octet to highlight the beginning of the frame. The EPON standard modifies the preamble field to transmit the LLID.

Each frame carries MAC addresses of resources and targets that have a length of 6 octets. The length/type field is used to represent the size of the transmitted data if its value is from 0 to 1500, which is the maximum amount of data being transferred. If this field is between 1536 and 63 535, this field is used to designate the Ethernet frame type. Ethernet data is variable in size, ranging from 46 to 1500 octets. Additionally, the Checksum field is used to detect damaged frames.

On table 2 it is also possible to see that the Ethernet frames carry a minimum of overhead bytes to transmit control information. This information and OAM (Ethernet OAM is a protocol for installing, monitoring, and troubleshooting metro Ethernet networks and Ethernet WANs.) information is transmitted using protocol data units (PDUs) and OAM frames, which are standard Ethernet frames labeled with a special value in the length/type field [3] [4].

Table 2: 802.3 Ethernet packet and frame structure

<table>
<tr><th>Layer</th><th>Preamble</th><th>Start of frame delimiter</th><th>MAC destination</th><th>MAC source</th><th>802.1Q tag</th><th>Length or Type</th><th>Payload</th><th>Frame check sequence</th><th>Interpacket gap</th></tr>
<tr><td>octets</td><td>7</td><td>1</td><td>6</td><td>6</td><td>4</td><td>2</td><td>46-1500</td><td>4</td><td>12</td></tr>
<tr><td>Layer 2 Ethernet frame</td><td></td><td colspan="7">← 64 - 1522 octets →</td><td></td></tr>
<tr><td>Layer 1 Ethernet packet</td><td colspan="7">← 72 - 1530 octecs →</td><td colspan="2">← 12 octecs →</td></tr>
</table>

- **Preamble and start frame delimiter** is used to sync hours of recipients and indicates the beginning of the frame.
- **MAC Source and Destination Address** - Media Access Control (MAC) of the source and destination network interface with a length of 48 bits.
- **802.1Q** standard allows you to split one physical Ethernet network into multiple logical networks by extending headers Ethernet frame for additional items. The optional part defines the VLAN virtual network.
- **Length or Type** for Ethernet II, it determines the type of the higher protocol, for IEEE 802.3 it specifies length by data.
- **Payload** use to transfer data in frame.
- **FCS** is used to check the sum of the frame.
- **CRC** is a special hash function used to detect errors during data transfer or storage.
- **Interpacket gap** is the time interval between packets that can not be transmitted by any station. After a packet has been sent, transmitters are required to transmit a minimum of 96 bits (12 octets) of idle line state before transmitting the next packet.

2.1.3 Way of communication in EPON

The IEEE802.3 standard defines two basic types of communications over Ethernet. One of these is to transfer information over shared media using the CSMA/CD protocol. The second way of communication is to connect terminal stations using switches and to use point-to-point full-duplex transmission. The EPON features combine both of these communication ways.

The basic transmission scheme in both directions is similar to that of other PON networks, duplex operation being solved using two different wavelengths for both transmission directions. In the download direction, the OLT continuously transmits time multi frames in which contributions to individual end units are allocated by using the Time Division Multiplex time division multiplex. These multi frames pass through the passive hubs to all ONU/ONT endpoints, where only the portion assigned to that end user is selected. The start of the multi frame is marked with a defined sequence to make it easier to detect its origin and derive bit sync. Data units in the multi frame are stored in an Ethernet frame format with an altered header and securing array.

In the download direction (Fig. 2) passive optical hubs are used, so this behavior resembles shared access to the media. OLT packet is broadcast, the destination ONU will take it to MAC address.

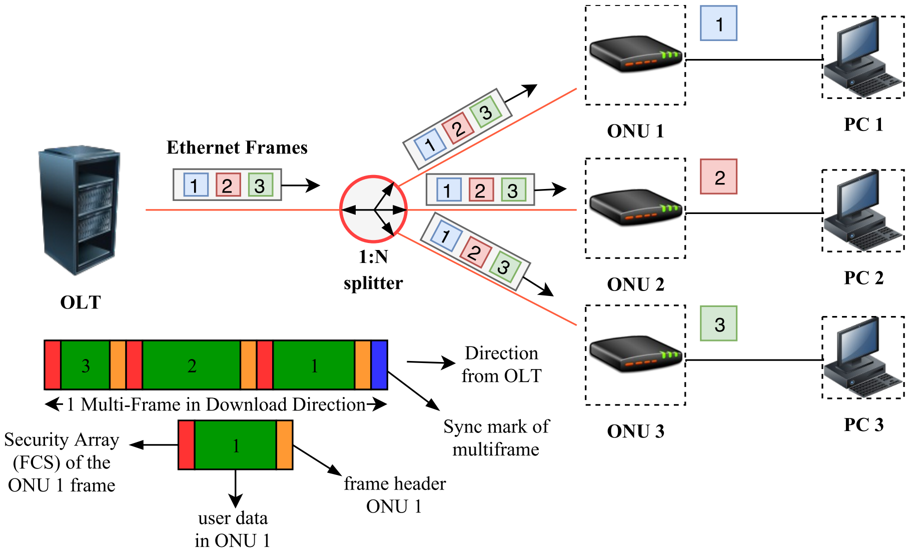

Figure 2: EPON TDM, downlink transmission

In the upload direction, a dedicated broadcasting system is implemented to ensure collision-free operation. In addition, the data units in the resulting multi-frame are separated by a guard interval. The CSMA/CD mechanism is hard to implement in this case because the ONU can not detect collision in OLT due to the properties of the optical hub. OLT can detect a collision and inform the ONU of the jam signal, but the delay in transmission in the PON network, the length of which may exceed 20 km, will greatly affect the effectiveness of such a measure.

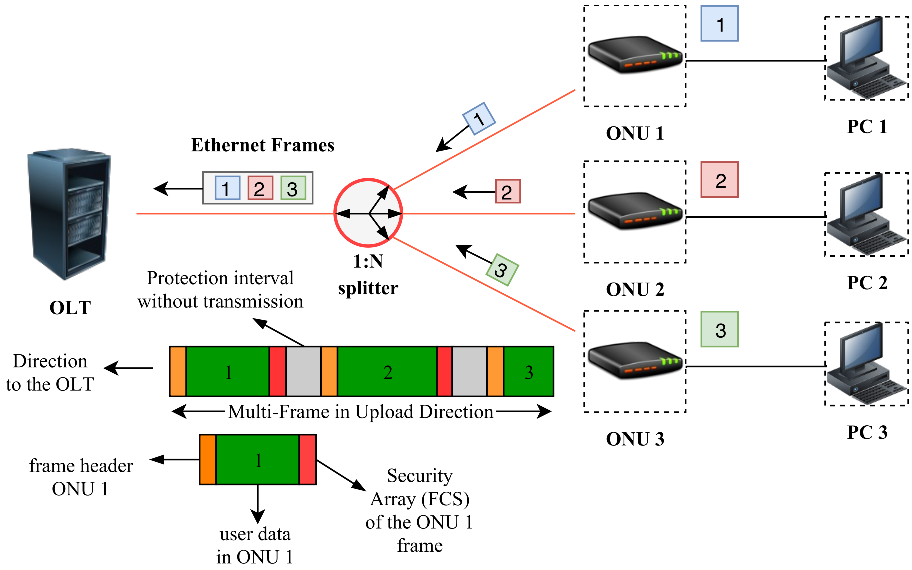

Figure 3: EPON TDMA, uplink transmission

Figure 3 illustrates upload of the data flow direction based on the splitting of the access time. ONUs are synchronized to a common reference time, and each unit has a dedicated time slot. Each such slot is capable of transferring several Ethernet frames. ONU collects the frames received from the end device and waits for its time slot. When its time slot occurs, the unit sends all frames collected at full speed that corresponds to one of the standard Ethernet speeds of 10, 100, 1000 or 10,000 Mbps. In case that ONU does not collect any frames to send, it sends 10 bits that indicate the inactivity [1].

2.1.4 Multiple Point Control Protocol (MPCP)

To support a time slot allocation by the OLT, the multi-point control protocol (MPCP) is being developed by the IEEE 802.3ah task force. This protocol relies on two Ethernet messages: A GATE and REPORT. A GATE message is sent from OLT to an ONU and used to assign a transmission time slot. A REPORT message is used by an ONU to convey its local conditions (such as buffer occupancy, etc.) to the OLT to help it make intelligent allocation decisions. Both GATE and REPORT messages are MAC control frames (type 88-08) and are processed by the MAC control sublayer.

There are two modes of operation of MPCP: auto-discovery (initialization) and normal operation. Auto-discovery mode is used to detect newly connected ONUs and learn the round-trip delay and MAC address of that ONU, and, in addition, perhaps some additional parameters yet to be defined. Normal mode is used to assign transmission opportunities to all initialized ONUs [12] [13].

The automatic update and initialization process takes place in the following steps:

1. OLT allocates an initialization slot, an interval of time when no previously initialized ONUs are allowed to transmit.

2. OLT sends an initialization GATE message advertising the start time of the initialization slot and its length.

3. Only uninitialized ONUs will respond to the initialization GATE message. Once the GATE message is received, the ONU sets its time according to the MPCP protocol time stamp.

4. When the ONU timer arrives at the initialization slot time (this time is also delivered in the GATE message), the ONU will send the initialization REPORT message. This message contains the ONU's source address and a time stamp representing a local ONU's time when the REPORT message was sent.

5. When the OLT receives the REPORT from an uninitialized ONU, it learns its MAC address and round-trip time. The round-trip time of an ONU is exactly the time difference between the time the REPORT is received at the OLT and the time stamp contained in the REPORT.

It is important to notice that MPCP is not concerned with particular bandwidth-allocation schemes. Rather, it is a supporting protocol necessary to deliver these decisions from the OLT to the ONUs [12] [13].

Run the normal MPCP protocol mode without detecting the presence of new ONUs is as follows:

1. From its higher layer (MAC control client), MPCP gets a request to transmit a GATE message to a particular ONU with the following information: time when that ONU should start transmission and length of the transmission.

2. MPCP layer in OLT and each ONU maintains a clock. Upon passing a GATE message from its higher layer to MAC, MPCP time-stamps it with its local time.

3. Upon receiving a GATE message matching that ONU's MAC address (GATE messages are unicast), the ONU will program its local registers with transmission start and transmission length times. The ONU will also verify the time when the GATE message arrived is close to the time-stamp value contained within the message. If the difference in values exceeds some predefined threshold, the ONU will assume that it has lost its synchronization and will switch itself into an uninitialized mode. In that mode, the ONU is not allowed to transmit. It will monitor its incoming traffic, waiting for the next initialization GATE message to perform initialization.

4. If the time the GATE message is received is close to the time-stamp value in the GATE message, the ONU will update its local clock to that of the time stamp. When the

local time reaches the start transmission value, the ONU will start transmitting. That transmission may include multiple Ethernet frames. The ONU will ensure that no frames are fragmented. If the next frame does not fit in the remainder of the time slot, it will be deferred until the next time slot.

2.1.4.1 Example of message communication

OLT sends a message to the GATE terminal all the ONUs/ONTs after another in the download direction and the expected response with the user data in defined time intervals.

This method is shown in fig. 4 and allows for better use of the multi-frame capacity in the up-link, but it has to be implemented a limitation for the maximum length of the sent user data so that each end unit ONU/ONT has a designated minimum capacity. Terminal units that have a larger amount of user data to send would fill a larger portion of the multi-frame in the download direction, and the remaining units would not have the required capacity.

A third method is also illustrated in FIG. 4, but adds another non-broadcast protection interval to the end of all the responses from the first cycle. New cycle queries and responses separates from the first cycle with an additional time interval without communication [12].

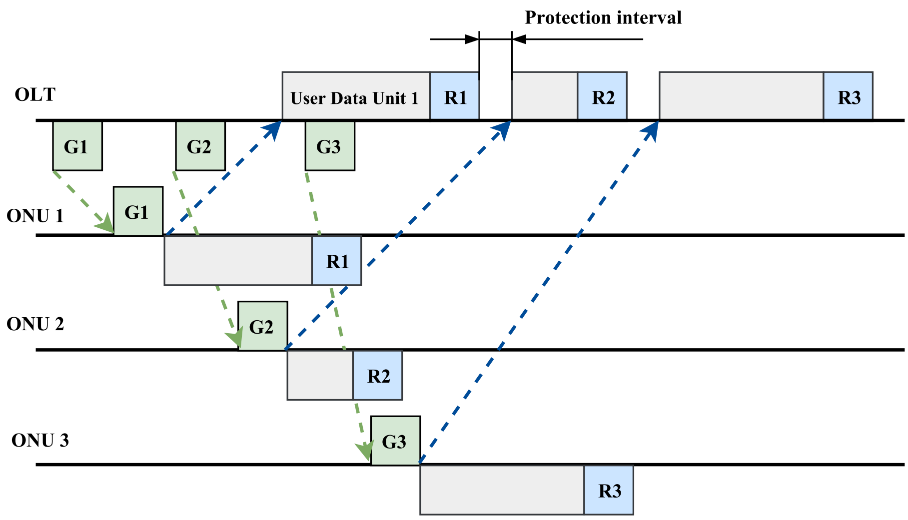

Figure 4: Query and response cycle of end units

2.1.5 Physical Medium-Dependent Sublayer (PMD)

The PMD dependent layer specifies the characteristics of the optical transmitters. Ethernet to acquire low-cost designs and use them for mass production. This philosophy is the key to the huge commercial success of Ethernet.

Two different distances between OLT and ONU are defined in EPON standards, a distance of 10 km and 20 km. Transmitter features for the 10 km long path define 1000BASE-PX 10-D

PMD and 1000Base-PX10-U PMD. For a distance of 20 km, the characteristics of the transmitter are described by the 1000BASE-PX 20-D PMD and the 1000BASE-PX20 PMD.

Most changes are made to the OLT unit. This increases the likelihood that end-users can use the same UN unit even if the path is extended [5].

2.1.6 Ranging Process

All ONUs are synchronized to the OLT clock based on a loop-timing mechanism. thus, an ranging process, which measures the Round Trip Time (RTT) between an OLT and an ONU, can be carried out using normal GATE and REPORT control messages.

The figure 5 describes the process between OLT and ONU. The OLT sends a normal GATE message with the time stamp T_0. The ONU would respond to the GATE message with a REPORT message after some delays, T_R. The time stamp on the REPORT message T_1 represents exactly $T_0 + T_R$ because the ONU has very precise loop timing received from the OLT. Upon received from the OLT REPORT message at T_2, OLT could then use the information T_2 - T_1 to determine RTT [69].

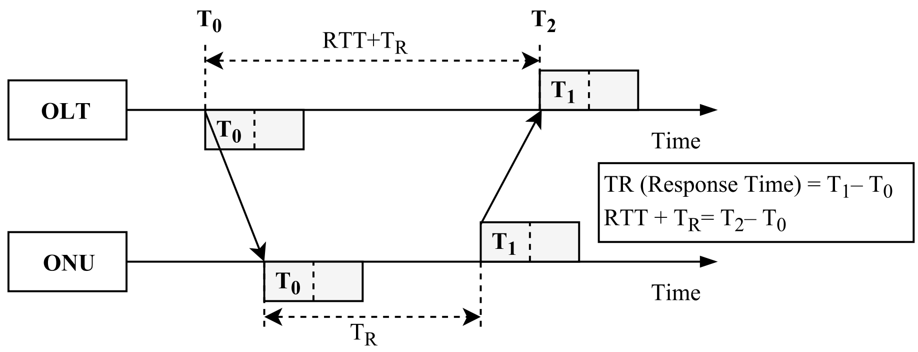

Figure 5: EPON ranging process [69]

2.1.7 Gate and Report Operations

Gate operation provides an OLT mechanism for specifying time windows in which end-to-end ONUs can be transmitted.

Differently from GPON, where OLT specifies start and end times in 1 byte increments, Gate operation specifies start time and time window length in increments of 16 ns (equivalent to 2 bytes in Gbit EPON).

2.1.8 Encryption and protection

The IEEE 802.3ah standard does not specify encryption and protection mechanisms for EPON. Encryption is important to ensure privacy when ONUs are directly linked to users, as is the case with FTTH systems. Protection is important in FTTB and FTTC systems where ONU is shared by a group of users.

Many implementations of EPON chips involve specific encryption mechanisms for each manufacturer, which may, if necessary, be authorized by the service provider.

2.1.9 GPON Architecture

Gradually increasing the transfer rate also increases the requirement for the PON optical layer. In order to cover the entire transmission distance, it was necessary to use more sensitive but also more expensive avalanche photo diodes instead of cheap PIN receivers. Without sufficient circuit protection, these photo diodes can easily be damaged if input optical power will get high values.

To support higher transmission speeds, GPON needs to use more powerful transmitters. The receivers must be able to handle a higher reception power overload and therefore a wider dynamic range. To reduce the requirements and easy implementation of the up-link OLT burst receiver, GPON sets the performance level mechanism.

In this mechanism OLT tries to balance the amount of power it receives from different ONUs, by sending instructions to each ONU regarding the increase or decrease of the transmitted power. As a result of ONUs that are closer to OLT, they transmit lower performance than the ONUs that are more distant. This concept of performance levels or performance control already existed in cellular radio networks.

2.1.10 10G-EPON

The IEEE 802.3av standard defines a passive optical access network with a transmission rate of up to 10 Gbit/s and backward compatibility with EPON technology.

This technology is also compatible with WDM-PON technologies. The transfer rates achieved by 10G-EPON are symmetrical 10 Gbit/s in both directions and unbalanced 10 Gbit/s down-link and 1 Gbit/s up-link.

2.1.11 Comparison of EPON and GPON

From a technical point of view, the main difference between GPON and EPON is support for TDM circuits.

GPON divides the upload and download signals into 125 μ with frames. Data frames are encapsulated with GEM (GPON encapsulation method), with the ability to segment. This allows the creation of TDM circuits with a guaranteed bandwidth of 64 kb/s between OLT and ONU.

EPON systems use variable length Ethernet frames in the transport layer. Therefore, you need to use Circuit Emulation to implement TDM circuits. While EPON systems have been optimized for Ethernet frame transmission, GEM makes it easier to adapt to other signal formats.

We consider splitting 1:32 for both systems, EPON offers an average bandwidth of 31.25 Mbit/s for each ONU drive direction.

GPON with symmetric bandwidth 2488 Mbit/s offers 77.75 Mbit/s for each ONU, that is 38.87 Mbit/s for each direction.

2.2 Introduction xDSL technology

Digital Subscriber line, it is a high-speed digital communication line. The biggest advantage of this technology is its price because DSL uses already existing copper cabling, so there is no need to install any cabling for existing buildings. In addition, most DSL systems allow simultaneous voice and data transmission. The data traffic is routed to a packet data network, while the voice information is sent to the public telephone network [65].

Table 3: Types of xDSL services available and the associated ITU-T recommendations [64]

DSL type	Description	ITU-T Rec	Data rate	Distance limit	Application
ADSL	Asymmetric DSL	G.992.1 (ex G.dmt)	1.544-8.448 Mb/s downstream 32-768 kb/s upstream	1.544Mb/s at 5.5km 2.048Mb/s at 4.8km 6.312Mb/s at 3.6km 8.448Mb/s at 2.7km	Internet and Web access, video on demand, motion video, remote LAN access
ADSL Lite	Splitter less ADSL	G.992.2 (ex G.lite)	1.544-6 Mb/s downstream (depends on subscribed service)	5.5km on 0.5mm wire	Same as ADSL, without splitter at the user's home or business, at lower speed
HDSL	High Bit Rate DSL	G.991.1 (ex G.hdsl)	1.544 Mb/s duplex on two twisted-pair lines; 2.048 Mb/s duplex on three twisted-pair lines	3.6km on 0.5mm wire	Replacement for E1/T1 service, WAN, LAN, server access
IDSL	ISDN DSL	-	128kb/s	5.5km on 0.5mm wire	Similar to the ISDN service but data only (no voice on the same line)
SDSL	Symmetric DSL	G.shdsl (G.992.2)	1.544 Mb/s or 2.048Mb/s on a single duplex line downstream / upstream	3.6km on 0.5mm wire	Same as for HDSL but requiring only one line of a twisted pair cable
VDSL	Very High Bit Rate DSL	G.vdsl (G.993 series)	12.9-51.8Mb/s downstream, 1.5Mb/s to 2.3 Mb/s upstream	1.4km at 12.96Mb/s, 0.9km at 25.82 Mb/s, 0.3km at 51.84 Mb/s	ATM networks; fiber to the neighborhood (FTTx)

2.2.1 Basic Rate ISDN

BRI is the first of the DSL family. The Integrated Services Digital Network (ISDN) was first developed in 1976 and was largely defined by recommendations developed by the CCITT (now known as the ITU). ISDN vision was a unified global network for data communication and telephony. ISDN focused on telephone services and data transfer by switching packets at lower speeds. This focus eventually became the biggest weakness, because ISDN networks were not suitable for higher speeds and long-standing sessions, which are characteristic of the Internet.

BRI transmits a total 160 kbit/s of symmetrical digital information. It consists of two B channels with a speed of 64 kbit/s, one D channel with a speed of 16 kbit/s and a control channel with a speed of also 16 kbit/s. B channels can be changed packets or circuits. The D channel transmits signaling packets. BRI uses 2B1Q transmission. Data is sent in both directions at the same time using a hybrid transmission that cancels the echo. The BRI range is 5.5 km from DSLAM.

2.2.2 IDSL

Application using BRI transmitters. BRI symmetric channels (128 kbit/s or 144 kbit/s) are concatenated into a single channel to transmit packets between the router and the computer. The transmitted signal is subjected to a 2B1Q encoding. The bit stream is divided into two bits, which are then expressed in one of four states. Most versions of IDSL work with a traditional ISDN network termination on the client side, so with IDSL, ISDN is replaced with a local switch by the router.

2.2.3 HDSL

The first concept of HDSL (high-bit-rate Digital subscriber line) appeared in the year 1986. The design of the HDSL transmitter was essentially an improved ISDN design. For the first time HDSL was used in 1992 and already in 1997 was 450 thousand HDSL lines were used around the world.

HDSL provides both directions via a 1.544 or 2.048 Mbit/s phone line up to 3.7 km across a twisted pair without a repeater. HDSL provides reliable transmission with a BER value ranging from 10-9 to 10-10. HDSL systems with a speed of 1.544 Mbit/s use two pairs of wires where each pair transmits 768 kbit/s data in both directions. For the description of HDSL transmission, the term uses double duplex. HDSL system with higher speed can use two or three pairs, where each pair uses full duplex. When using the repeater, the achievable distance increases up to 7.3km, using two repeaters even up to 11 km [64] [65].

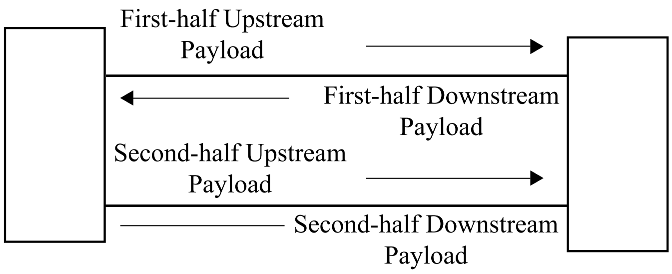

Figure 6: Dual-duplex HDSL

2.2.4 HDSL2

The development of standards for the second generation of HDSL technology began in the year 1995, to provide the same transmission speed and range for the use of only one pair of wires. HDSL2 uses Trellis-Coded Pulse Amplitude Modulation (TC-PAM) modulation and more sophisticated coding techniques. A suitably selected down-link and up-link offset helps to prevent crosstalk.

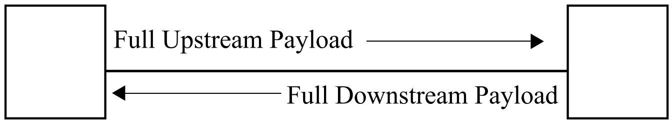

Figure 7: Single-duplex HDSL2

The down-link rate is much larger than the up-link, so the asymmetric term is used. The analog voice is transmitted in the frequency of the base band and is combined with the data transmission over a low pass, which is usually referred to as a splitter. ADSL is further composed of the ADSL transmitting unit on the side of the panel (ATU-C), the local loop and the ADSL transmitting unit on the client side (ATU-R). ADSL is specified in ITU-T recommendation G.922.1.

2.2.5 SHDSL

SDSL (Symmetric Digital Subscriber Line) and SHDSL (Single-Pair High-Speed Digital Subscriber Line) are two different terms for the same technology. The two names exist because two separate organisations introduced their own standards for this technology. (The European Telecommunications Standards Institute approved the SDSL standard and the International Telecommunication Union (ITU) approved the SHDSL standard.).

A single-pair high-speed digital subscriber line (SHDSL) is a type of symmetric digital subscriber line (SDSL) that has symmetric upload and download speeds over conventional copper telephone lines, which is faster than what a conventional voiceband modem can provide. SHDSL uses trellis-coded pulse-amplitude modulation (TC-PAM) that operates at frequencies that encompass those used by analog voice POTS (plain old telephone service), meaning that a DSL splitter or frequency splitter cannot separate analog voice and data [66] [67].

2.2.6 ADSLx

Asymmetric digital subscriber line is a type of digital subscriber line (DSL) technology. ADSL differs from the less common symmetric digital subscriber line (SDSL). In ADSL, Bandwidth and bit rate are said to be asymmetric, meaning greater toward the customer premises (downstream) than the reverse (upstream).

The ADSL concept was developed in the early 1990s. First, ADSL had fixed 1.5 Mbit/s for down-link and 16 kbit/s in up-link for MPEG-1 applications. This is sometimes referred to as ADSL1.

It was later shown that some applications require higher speeds, which have been achieved through more advanced transfer techniques. 3 Mbit/s in the direction of the client and 16 kbit/s in the direction from the client was sufficient for two simultaneous running MPEG-1 transfers (ADSL2). ADSL3 offered speeds of 6 Mbit/s and 64 kbit/s to support MPEG-2 video [14].

The ADSL concept contains two basic components:

- Crosstalk are reduced by a much lower bit rate and a lower bandwidth for up-link compared to down-link.
- Simultaneous transmission of analog voice with data is enabled by transmission of data in the frequency band higher than voice telephony.

Many ADSL systems use a frequency multiplex, which places the broadcast away from the customer in a different frequency band than the broadcast towards the customer, thus preventing cross-talks.

The protection band must make the noise from POTS interfere to digital broadcasting. Other ADSL systems use a transmission technique for echo cancellation, where the frequency for up-link is located within the bandwidth for down-link. By overlapping the bands, the total bandwidth is reduced. However, there may be crosstalk and implementation involves more complex signal processing.

Table 4: ADSL standards [15]

Version	**Standard name**	**Common name**	**Downstream rate**	**Upstream rate**	**Approved in**
ADSL	ITU G.992.1 Annex B	ADSL over ISDN	12.0 Mbit/s	1.8 Mbit/s	2005
ADSL	ITU G.992.1 Annex A	ADSL over POTS	12.0 Mbit/s	1.3 Mbit/s	2001
ADSL	ITU G.992.2	ADSL Lite (G.lite)	1.5 Mbit/s	0.5 Mbit/s	1999-07
ADSL	ITU G.992.1	ADSL (G.dmt)	8.0 Mbit/s	1.3 Mbit/s	1999-07
ADSL	ANSI T1.413-1998 Issue 2	ADSL	8.0 Mbit/s	1.0 Mbit/s	1998
ADSL2	ITU G.992.3 Annex L	RE-ADSL2	5.0 Mbit/s	0.8 Mbit/s	2002-07
ADSL2	ITU G.992.3	ADSL2	12.0 Mbit/s	1.3 Mbit/s	2002-07
ADSL2	ITU G.992.3 Annex J	ADSL2	12.0 Mbit/s	3.5 Mbit/s	2002-07
ADSL2	ITU G.992.4	Splitter-less ADSL2	1.5 Mbit/s	0.5 Mbit/s	2002-07
ADSL2+	ITU G.992.5 Annex M	ADSL2+M	24.0 Mbit/s	3.3 Mbit/s	2008
ADSL2+	ITU G.992.5	ADSL2+	24.0 Mbit/s	1.4 Mbit/s	2003-05

2.2.7 RADSL

RADSL (Rate-Adaptive DSL) is a term used to describe ADSL systems capable of automatically determining the transport capacity of individual local loops and then working the highest allowed speed on that line.

2.2.8 VDSL/VDSL2

VDSL, or Very-High-Bit-Rate Digital Subscriber line, allows Internet Service Providers to provide fast connection speeds via legacy copper lines. VDSL and VDSL2 can provide faster broadband performance when compared to ADSL/2+ up to approximately 1.5km distances. After 1.5 km distances, VDSL2 exhibits performance rates comparable to ADSL2+.

Optical fiber is used for transmission rates in addition to the last few hundred meters. Most DSL systems are primarily used to connect directly to the switchboard and the customer. On the other hand, VDSL is primarily used for the subscriber line from ONU that which is usually less than a mile from the customer. Few VDSL lines are serviced directly at the control Panel.

The optical fiber connects the ONU to the central office.VDSL transmission via twisted pair is used only for a few hundred meters from the ONU to the subscriber.

VDSL2 represents the best of ADSL, ADSL2 + and VDSL. xDSL maintains its dominant position in the market for broadband connections as it offers universal fast access for a variety of services and applications.

Standards VDSL transmits at frequencies up to 12MHz. Downstream frequencies are from 138kHz to 3.75MHz and from 5.2MHz to 8.5MHz. Upstream transmission are reserved frequencies from 3.75MHz to 5.2MHz and from 8.5MHz to 12MHz. Standards VDSL2 uses frequencies up to 30 MHz.

VDSL2 supports 8 distinct "profiles" with varying maximum downstream and upstream throughput as well as different bandwidth frequencies and transceiver power. For example, profiles 8a-8b and 12a-12b are ideal for Fiber to the Node (FTTN) deployments. Profiles 17a can be used for Fiber to the Cabinet (FTTCAB), and Fiber to the Building (FTTB) can utilize profile 30a [68].

Table 5: VDSL2 profiles [68]

Profile	8a	8b	8c	8d	12a	12b	17a	30a
Bandwidth [kHz]	8.832	8.832	8.832	8.832	12.000	12.000	17.664	30.000
Sub channel	2048	2048	2048	2048	2783	2783	4096	3479
Maximum Downstream bit rate [Mbps]	48	48	48	48	48	48	60	120
Maximum Upstream bit rate [Mbps]	10	10	10	10	30	30	30	100

2.2.9 Hybrid Access Networks

Access networks are communications networks that connect private LAN, such as networks in individual homes, with public urban and backbone networks, such as those built by service providers, to join subscribers to the Internet. Private LANs often use high-speed wired or wireless communication technologies (such as IEEE 802.3 Ethernet or IEEE 802.11 WiFi). These communication technologies are cost-effective in small networks, due to short distances and consequently low installation costs.

Urban and backbone networks use, for example, DWDM technologies. These technologies are cost-effective because they share a large number of subscribers. Access networks require significantly higher installation fees compared to the local networks, because they need to cover much larger areas.

At the same time, for smaller access networks, there is no cost sharing as in the urban network, so the price of paying subscribers increases. As a result, it means that the technology for access networks must keep the installation costs low. This will be achieved by using already existing, limited bandwidth, copper cables, or optical fiber sharing, which holds the installation keeps costs low.

Hybrid networks use both metallic cables and shared fiber optics. The thread extends from the service provider's headquarters to the location, called the Drop-point. The last couple of hundred meters to the subscriber is reached by an existing twisted pair.

Fig. 8, shows a hybrid network architecture that combines the advantages of existing cabling with the benefits of DSL technology that can make transmission up to 1 Gbit/s at short distances.

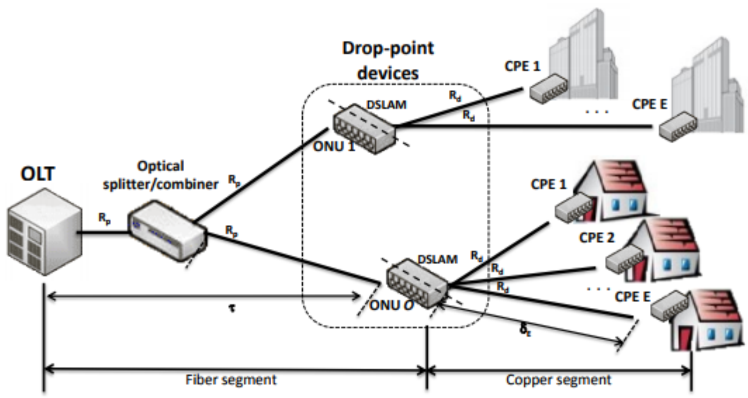

Figure 8: Architecture hybrid xPON/xDSL [16]

A hybrid PON/xDSL access network architecture consists of a passive optical network (PON) connected to multiple copper digital subscriber lines (DSLs). The PON OLT connects to several drop-point devices. Each drop-point device is a combined PON Optical Network Unit (ONU) and DSL Access Multiplexer (DSLAM). Through the DSLAM, each drop-point device connects to multiple subscriber DSL customer premise equipment (CPE) nodes [16].

2.3 Vectoring VDSL2

It is a state-of-the-art technology to achieve high transfer rates by removing crosstalks between parallel lines in a metallic cable.

One of the ways to increase the speed of connection is VDSL2 vectoring technology. Combined with investments in the development of the fiber optic network through the construction of the Fiber To The Cabinet (FTTC), when the local loop is shortened, the Internet will be significantly accelerated even with the use of existing metallic lines. Deploying vectoring on VDSL2 technology will allow you to reach speeds of over 100 Mbit/s.

For the efficient operation of VDSL2 vectoring, it is crucial that the end-user device on the Internet Service Provider side supports VDSL2 vectoring.

2.3.1 Principle of vectoring

The principle is known by the technicians of the company for decades. On airplanes, background noise is normally read by the pilot to make his message heard. It's the principle that Dolby uses in the headphones to suppress noise. The same is true for VDSL2 on lines but in the 17 MHz band. We know from each pair what crosstalks are generated in the other pairs, and we read them. When we have 500 pairs, they all mutually interrupt, and relatively complex mathematics and computational power are needed. Suppliers today have DSLAM with up to 576 lines.

It is not an extreme revolution, but rather a natural evolution. It's not an optics yet, but we're getting closer to it.

2.3.2 Closer and faster

Vectoring gives the best results on the shortest lead. On connections up to 300 meters it is able to deliver theoretically up to twice the speed, over half a kilometer but no difference is so dramatic and over a mile is already very small.

VDSL2 with vectoring can do over 150 Mbit at short interconnections in a laboratory environment, but that does not end. In the future, VDSL3 or Vplus is awaiting us. It should roughly double the speed. There is also G.fast technology, which makes sense on very short lines. So far we can not use its potential because we are not so close to our customers [96] [97].

2.4 Active Optical Network (AON)

The main advantage of AON with PON comparison is to provide greater reach. The disadvantage is the provision of power to the active network elements used in the distribution network. Low Passive Optical Access Network appears to minimize operating costs. The active optical network contains active elements in the form of digital transmission equipment and is mostly implemented by SDH technique. SDH standardized protocols that transfer multiple digital bit streams synchronously over optical fiber using lasers or highly coherent light from light-emitting diodes (LEDs).

2.5 Pasive Optical Network (PON)

Passive optical access networks do not use the active element, but a passive optical hub, called splitter. It does not reinforce or repeat the optical signal, it only splits it in the same proportion as it is built. Output signals of the passive optical hub are weaker than when they exit the active element.

2.5.1 TDM-PON

It uses the TDM methods that have been used since the digital transmission, and the synchronous optical multiplex. The principle of TDM-PON is the use of one wavelength in duplex mode, or using two filaments.

OLT transmits TDM down-link traffic and manages up-link traffic. In the TDM down-link direction, OLT traffic reaches a 1:N optical divisor. This means that all N outputs of the distributor carry the same operation. However, the time window is associated with a particular ONU, that finds its window and pulls out the data for itself. Operation from OLT usually takes place at a wavelength of 1490 nm.

In the up-link direction, the ONU sends the data in the corresponding time window using a combiner. To the combiner arrive time windows from all ONUs and merged into a single TDM stream in a single-mode fiber. It is necessary for the time windows that come to the combiner to be properly synchronized, otherwise they would collide. Operation towards OLT usually takes place at a wavelength of 1310 nm.

3 Optical Access Network

Optical access networks are not a new concept, but have been considered as a possible solution for access networks since the early 1980s. This idea was rejected because the technology was not yet sufficiently advanced. This was a costly solution and there was no demand for bandwidth. In last years there has been significant progress in the field of fiber and optical components. Their increased production has, of course, reduced their purchase price. It also increased demand for faster and better transmission. That is why the worldwide introduction of optical communications is becoming a common matter. Fiber Access Networks are capable of transmitting long-range bandwidths and anticipated future voice, data and video services.

Nowadays the optic access network becomes the basis for connecting the customer and the provider. In the past, metallic lines were the most commonly used for connections. Today, the metallic leadership that has been used for many years now prevails. Because of the higher demands on the speed and quality of the services provided, optical access networks are being introduced. These networks have speeds in tens of Gbit/s and are more suited to multimedia services that are now highly desirable. These services include, for example, IPTV or VoIP. All communication in optical access networks takes place over fiber optics [73].

3.1 Arrangement optical access networks

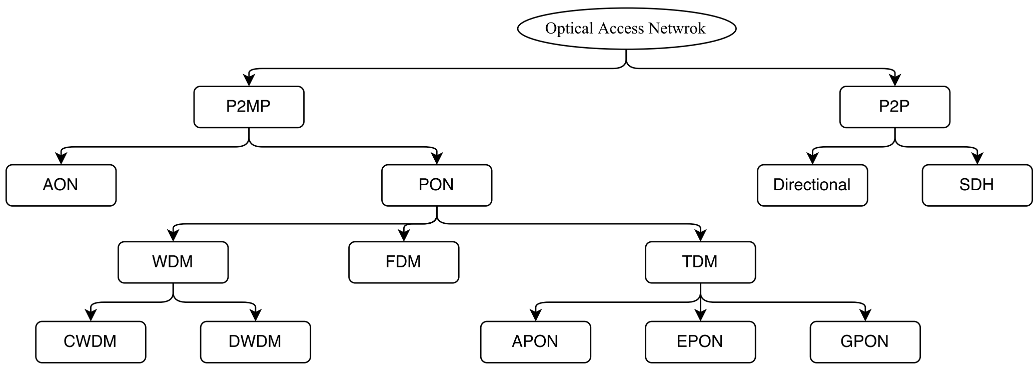

Figure 9: OAN Architecture

On fig. 9 it can be seen that the optical access networks are divided into two branches. Distribution is caused by according to signal to the end user. It is divided into multi-point networks and single-point networks. These groups are further subdivided into other subgroups.

For Point-to-Multi-point (P2MP), it is divided into passive optical networks (PONs) and active optical networks (AON). In the multi-point network, the most widely used passive optical access network is primarily the use of active elements that use electricity, but in the end and main optical units the active elements are necessary. The second group in the multi-point network is an active optical access network, and it uses optical amplifiers and other active elements to make them move over longer distances.

In this connection mode, each user is individually connected with one thread or two. If the connection was made by two threads, then each thread would be used for a different transfer method. One thread would be used for upstream and the other for downstream [49] [1].

3.1.1 Evolution Passive Optical Networks of Next Generation

New generation access networks are successors existing passive optical networks, which, in the case FTTNs complement the existing metallic access network or using FTTH technology are composed only of optical elements. They are based on the parallel transmission of a plurality of optical signals of different wavelengths in a common optical fiber.

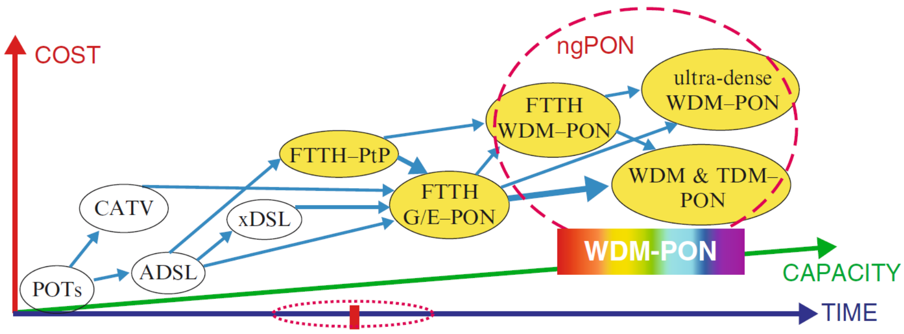

Figure 10: Evolution of access technologies [80].

3.1.2 P2P and P2MP

We can divide optical access networks by access means. There are two basic types of access networks that use optical data transmission:

- Point-to-point networks.
- Point-to-multi-point networks.

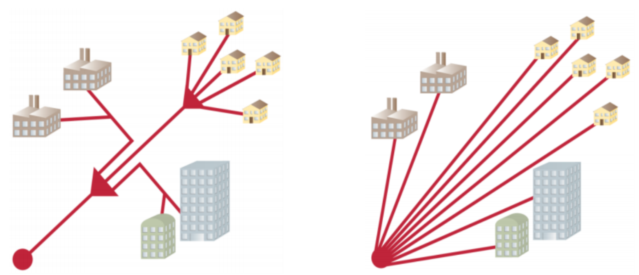

Figure 11: Point to Multi-Point (P2MP), Point to Point (P2P) [49]

P2P networks are mainly used in backbone networks, but are now more used including optical access networks. The usual use is that each ONT is connected to optical fiber directly to the OLT optical port. The advantage is large transmission speed and bandwidth, but these networks are more expensive to build, because there is one thread to each end point and much more fiber is needed and resources for implementation.

P2MP is a characteristic use in access networks. Their main advantage is that multiple end users can be connected to one optical fiber. This reduces the financial difficulty resources and the amount of consumed fiber. Between the central office, where is OLT and the end customer where it is ONU, there is an active or passive member that divides the optical signal according to the ratio offered by the member. The latest technology can divide the signal up to 1: 128. P2MP networks can be divided into Active Optical Networks (AON) and Passive Optical Network (PON).

3.2 Basic units of the optical access network

There are three elements in optical access networks. These include a transmitter, a transfer medium, and a receiver. Elements are understood to mean, that the transmitter serves as the radiation source, the transmission medium is an optical fiber through which the information is transmitted in photon form.

The optical access network transmits the signal between the end point and the connection network. It is composed of several active elements. Distributor side is used OLT and end-user ONT and terminating ONUs.

3.2.1 Optical Line Termination (OLT)

It is located on the Internet provider side. This device converts the electrical signal to optical and conversely.

OLT device is an optical line termination, or an endpoint from a PON service provider. The main function of OLT is to control information flow through ODN in both directions. The maximum distance supported for ODN transmission is 20 km. OLT has two directions of operation:

1. Up-link - traffic to OLT. It transmits different types of data and voice from users.
2. Down-link - This is traffic from OLT. The received data, voice, and video traffic from the network is sent to individual end users.

3.2.2 Optical Distribution Network (ODN)

ODN provides optical transmission medium for the physical connection between OLT and ONU. ODN includes fiber optics, optical connectors, passive optical dividers, and accessory components. Data transfer and its quality directly affect the performance, reliability and scalability of the PON system.

3.2.3 Optical Network Unit (ONU)

ONU is an optical network element to end the PON on the subscriber's side. It serves as an interface between the optical and metallic access networks. An optical network drive may establish a Network Termination.

3.2.4 Optical Network Terminal (ONT)

ONT is an element of the optical network, which is actually a special type of ONU, which mediates services specific to one customer.

3.2.5 Network Termination (NT)

The NT element serves as a general network termination that is used on the subscriber side.

3.3 Fiber to the x (FTTx)

Their name describes the length of conduction of the optical fiber and the location where the optical fiber is terminated and converted to metallic.

With its high bandwidth potential, FTTx has been closely coupled with triple play of voice, video and data services. And the world has now evolved beyond triple play to a converged multi-play services environment with a high bandwidth requirement. Applications like IPTV, VOIP, RF video, interactive online gaming, security, Internet web hosting, traditional Internet and even smart grid or smart home are widely used in FTTx network [47].

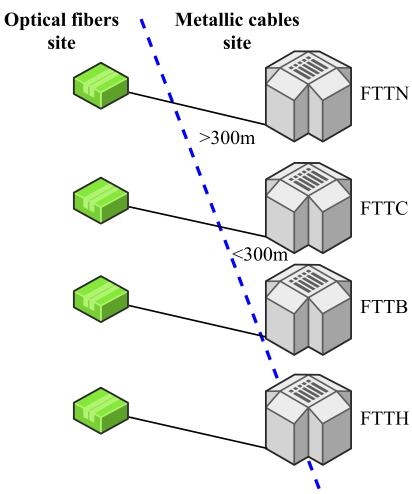

Figure 12: Different types of FTTx networks.

A schematic illustrating how FTTx architectures vary with regard to the distance between the optical fiber and the end user. The building on the left is the central office; the building on the right is one of the buildings served by the central office. Dotted rectangles represent separate living or office spaces within the same building [48] [49].

3.3.1 Fiber To The Node (FTTN)

The optical fiber is brought to the distribution cabinet, where several hundred participants can be connected from nearby surroundings using metallic wiring or coaxial cables. This type of connection is connected via an optical line to the node, which is located 1km from the customer. From this node leads to a customer line that is metallic.

3.3.2 Fiber To The Curb (FTTC)

The thread is brought to the central office, located in close proximity to the subscribers connection. The endpoints of the network are connected to metallic twisted wires or coaxial cables. This architecture involves connecting fewer users and a maximum distance of 300m from the cabinet.

3.3.3 Fiber To The Building (FTTB)

Optical fiber is brought into reserved areas on the ground floor or basement of the building. A switchboard is located in these spaces, from which the connectivity is distributed through metallic wiring, CAT-5 category cabling within the internal LAN network in the building. The method of this connection is mainly used in larger buildings. The divorces of the building to the user are solved using a classic UTP cable.

3.3.4 Fiber To The Home (FTTH)

The fiber is conected directly to the house or apartment and is terminated by the customer. The advantage of this solution is that the customer can have connectivity up to 1Gbit/s. Customer distance from the central office can be up to 20 km. However, this method is the most expensive of all variants.

3.3.5 Fiber to the Premise (FTTP)

FTTP is a North American term used to include both FTTH and FTTB deployments. Optical fiber is used for an optical distribution network from the central office all the way to the premises occupied by the subscriber. Since the optical fiber cable can provide a higher bandwidth than copper cable over the last kilometer, operators usually use FTTP to provide voice, video and data services.

3.4 Optical Signal Transmission Specifications in the OAN

Data transmission in the form of an optical signal in an optical access network must provide duplexing transmission environment (RFOG technology only requires simplex transmission at a wavelength of 1550 nm). Optical signals for both directions may be in the optical access network transmitted in the following ways:

- **Space Division Multiplexing** (SDM) - simplex transmission with SDM splitting: for everyone the direction of transmission of the optical signal is one optical fiber.

- **Wavelength Division Multiplexig** (WDM) - duplex transmission with WDM division: optical the signals are transmitted duplex by one optical fiber, for the optical character OLT-ONU P2P (Point-to-Point) access networks one direction at a wavelength of 1310 nm and a second direction at a wavelength of 1550 nm, for the character of the OLT-ONU optical access network P2MP (Point-to-Multipoint), referred to as PON (Passive Optical Network), one direction at wavelength 1310 nm and second direction at wavelength 1490 nm),

- **Frequency Division Multiplexing** (FDM) - Duplex transmission with FDM division: Optical the signals are transmitted in both directions by one optical fiber on one wave length, the directions of transmission are mutually separated by frequency.

3.5 Fiber dispersion

Dispersion is the spreading out of a light pulse in time as it propagates down the fiber. Dispersion in optical fiber includes model dispersion, material dispersion and waveguide dispersion [88].

3.5.1 Chromatic Dispersion (CD)

Material dispersion also called Chromatic Dispersion created on single thread SM (Single Mode) is caused by the velocity of light (or its refractive index) being a function of wavelength. Each wavelength takes different amounts of time to propagate the same path.

A D_{ch} unit that is independent of the length of the fiber is used to compare the optical SM fibers.

$$D_{ch} = D_m + D_{vl} \tag{1}$$

The chromatic dispersion is composed of two components, D_m - material dispersion and D_{vl} - waveguide dispersion.

The material dispersion is due to the refractive index dependence on the wavelength of the radiation, the waveguide is caused by being closed in the waveguide.

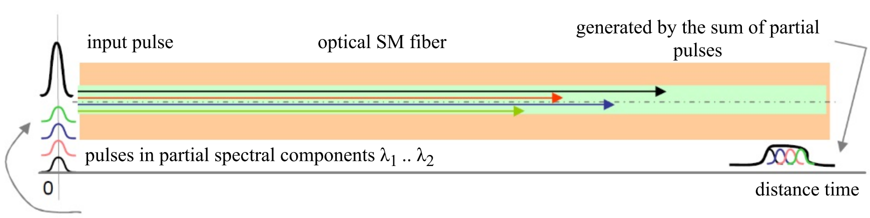

Figure 13: Chromatic Dispersion.

3.5.1.1 Compensation of chromatic dispersion

Using Fiber Dispersion Compensation (DCF) are the fibers that are used to compensate for the dispersion of classical SMF fibers according to G.652 at wavelengths greater than 1300 nm (C, L band), where the fiber exhibits a large value of the D_{ch} coefficient (eg. $D_{ch} = 17ps/nm \cdot km$) [87].

3.5.2 Polarization Mode Dispersion (PMD)

PMD is only important in single mode fibers. In single mode fiber, only one mode can propagate. This mode is actually composed of two distinct polarization modes. The electric fields of the two modes are perpendicular to each other.

Caused by the anisotropic properties of the optical fiber material, the refractive index is not constant in all directions. PMD is a random variable dependent on time, wavelength, fiber deformation, bends along the path, etc.

3.5.2.1 Methods for reducing the impact of PMD

lower transmission rates per unit, use of dispersed-controlled volitions, use of forward self-repair code (FEC) [87].

3.6 Next Generation Access Optical Networks (NGA)

NGN is a packet-oriented network provide telecommunication services and provide access to broadband transmission enabling technologies QoS, where services are not tied to used transport technologies. NGN provides users with unlimited access to service providers, supports mobility of users and allows them permanently and versatile services available.

By merging the properties of the telephone and data networks that are based on the IP and the benefits of both of them were the Next Generation Network (NGN). The main features of NGN is the packet transfer of information and the provision of telecommunication services (audio, video and data).

The new generation network is a high-speed packet network. The high-pass network uses packet transfer based on the IP protocol. NGNs guarantee service providers quality (QoS). The services provided are independent of the technology used and can be used by different providers. They also guarantee the free movement of users in the network, making them constantly using the services.

New generation networks are separated from each other individual planes. The lowest transport plane that carries user data is separate from the management plane that is in charge of networking and maintenance. Similarly, the management layer is separated from the highest application level that represents the service provided to individual providers. Split planes allow each other to provide services to users from different operators within one network

Technology is the successor to existing passive optical networks, which aims to exponentially increase data traffic and speed up. The main goal of this technology is primarily to reduce total costs. This reduction is due to multiplexing with limited complexity. New generation networks are based on parallel transmission bigger the amount of optical signals of different wavelengths in the common optical fiber [50].

3.6.1 NGA Architecture

The NGA architecture is composed of three domains, when the first domain is made up of application servers. These servers provide for services that are referred to as IMS (IP Multimedia Subsystem) and are connected to the backbone network. The second domain of NGA is called management and uses reliable servers that evenly spread across the network. The last domain that makes up NGA is the transmission part of the network that is implemented with IP technology. Except of these domains, the NGA architecture is still composed of gates. These gateways provide backward compatibility with existing networks that work with TDM and circuit connections (ISDN, PLMN). The gates are divided into several types:

- Media Gateway (MGW) – is used to convert user information and convert encoding.
- Signaling Gateway (SGW) – signaling of packet networks (SIP, H. 323) on signaling, telephone networks (SS7, ISDN).
- Media Gateway Controller (MGC) – coordinate the operation of the system.

The main advantage of the new generation network is the emergence of a unified communications network for all types of services. Costs are reduced operation and maintenance of the network. The advantage for the end-user is the possibility of using different types of services with one terminal [74].

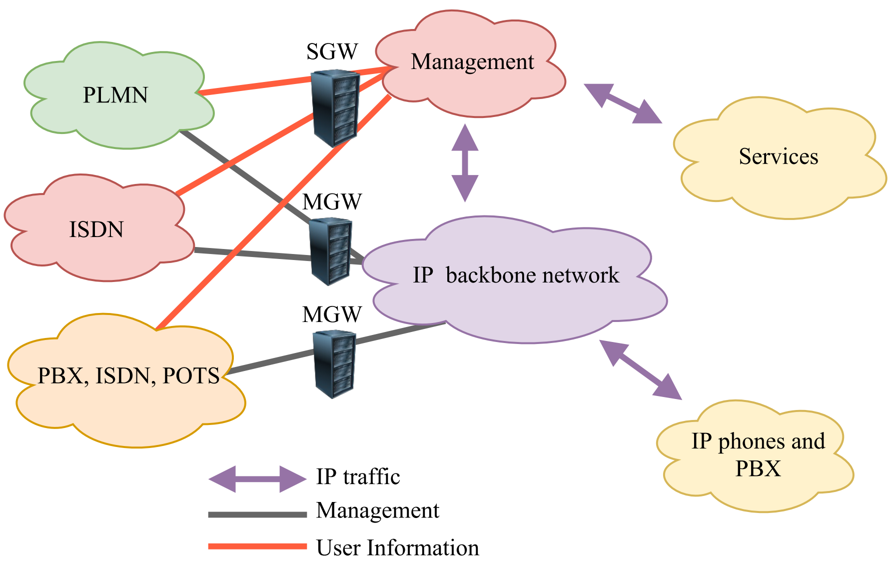

Figure 14: Architecture of converged NGN solutions [74]

3.6.2 NGA1

It is the first generation of new generation NGA optical networks introduced by ITU-T and FSAN (Full Service Access Network). This NGA1 architecture is an upgrade of existing optical networks while NGA2 is a long-term solution in passive optical networks.

The cost of building an access network is the most costly optical distribution network (ODN). They represent up to 70% total investment. The main requirement for NGA1 is coexistence with deployed GEPON systems.

3.6.3 NGA2

The FSAN (Full Service Access Network), which is made up of 85 leading global operators, at the end of 2012, accepted ITU-T NG-PON2 recommendations. This new standard is to be ratified by 2013. It will be based primarily on the WDM wavelength multiplex.

The theoretically designed version of the NGA2 concept will, for example, achieve shared transmission rates of up to 40 Gbit/s (4 Wavelengths of 10 Gbit/s or 40 Wavelengths per 1 Gbit/s). Of course at the cost of a brand new design and concept without the possibility of backward compatibility with previous generations of networks [62].

3.6.4 X Gigabit PON1 (XG-PON1)

The name XG-PON1 is associated with the name 10G-PON. This is a variant that is backwards compatible with the older GPON, where it is not necessary to change the ODN optical access network. For functionality backward compatibility it was necessary to appropriately select wavelengths in the band for each direction. Suitable wavelengths that are used for upload and download.

For XG-PON1, G.987 is recommended the 1: 256 splitter used, as opposed to the older GPON variant means a 4-fold increase of connected users (GPON is a ratio of 1:64). Another improvement is the maximum distance between the OLT and the ONU, which is 20km. A significant improvement is also the bit rate that now moves for 10 Gbit/s downlink and 2.5 Gbit/s uplink.

3.6.5 X Gigabit PON2 (XG-PON2)

This new generation puts much more demands on the lasers that are used on ONU and this fact is reflected in the higher costs. The speed for this variation is for the 10 Gbit/s downlink and for the 10 Gbit/s uplink.

3.6.6 10GEPON (10 Gigabit Ethernet PON)

Passive Optical Access Network 10GEPON is based on the previous EPON variant. It is based on the transmission of Ethernet frames, but it brings changes in transmission parameters (shared bit rate, wavelength, and security). The main requirement was retrospective compatibility with the older version of EPON so that both variants could operate within one optical distribution network at the same time. This has resulted in a cost saving because it was enough to replace modules on OLT optical line terminals.

The directional separation of the transmission is solved by wavelength division. In the downstream direction, a band with wavelengths of 1575 nm to 1580 nm is reserved for both variants. For the upstream direction, it is a symmetrical mode of 1260 to 1280 nm and a non-symmetrical mode of 1260 to 1360 nm. The only OLT unit is connected via a passive hub up to 32 ONT terminals. The range is 20 km. For the 10GEPON network, three attenuation classes were determined. Symmetric mode are marked PR10, PR20 and PR30, asymmetrical as PRX10, PRX20 and PRX30. There is an expansion of attenuation classes [72].

3.6.7 WDM-PON

This technology created that passive optical networks that are based on time division TDMA are close to performance limits. This is another new generation of optical access networks, which it uses for transmission wave multiplex, it is able to work with a plurality of separate wavelengths in one optical fiber. The new generation of WDM-PON gives us superior performance, flexibility and overall performance of optical access networks.

For older versions of optical access networks, only one optical fiber could be used for one transmission of information. This meant that all transmission channels were transmitted at the same time. With the advent of the WDM-PON technology, individual transmission channels can be transmitted individually in one thread. This transfer can take place both upstream and downstream, which means it is fiber bidirectional [76] [80].

3.6.7.1 Wave Division Multiplex (WDM) Previously, it was possible to use optical fiber for only one transmission. Today, using WDM technology, each color can be transmitted individually by one thread. We understand the color as a single transmission channel that can transmit separate data even in the opposite direction. Optical fiber becomes bidirectional. The basic principle of wave division is shown in the fig. 15 [76].

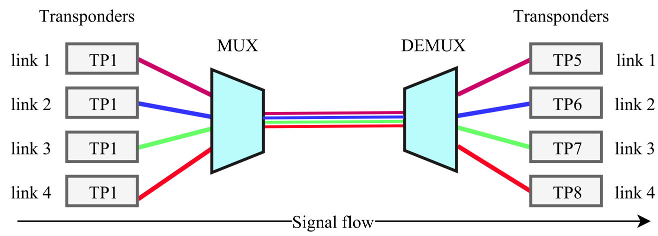

Figure 15: wavelength-division multiplexing (WDM)[75]

3.6.7.2 Wide Wavelength Division Multiplex (WWDM) technology belongs to older techniques. Wave multiplex is most commonly used for the transmission of 1Gbit/s and 10Gbit/s Ethernet. The technology uses only wavelengths around 1310 nm with a single wavelength spacing of 20 nm.

3.6.7.3 Coarse Wavelength Division Multiplex (CWDM) is like a robust and economical solution against DWMD. It focuses mainly on metropolitan networks. Wavelengths specify ITU-T G.694.2 with the first carrier 1270nm. Channel spacing is 20 nm with a tolerance of 6.5 nm. Larger channel spacing is required due to the considerable dependence of the transmitted wavelength on the temperature (less quality designs and cheap sources of radiation for the access network), which may fluctuate according to the ambient temperature.

CWDM technology broadcasts up to 16 channels divided into the following bands:

- O band (Original): wavelengths 1260-1360 nm, carrier number 1-5,
- E band (Extended): wavelengths 1360-1460 nm, carrier number 6-10,
- S band (Short): wavelengths 1460-1530 nm, carrier number 11-14,
- C band (Conventional): wavelengths 1530-1565, carrier number 15,
- L band (Long): wavelengths 1565-1625 nm, carrier number 16-18.

3.6.7.4 Dense Wavelength Division Multiplex (DWDM) is one of the most sophisticated systems used in optoelectronic. The basis is the wave division in the band C and L band, which can work at the minimum channel spacing, such as 0.8 or 0.4 nm. For remote and backbone optical links of band C, L and S with a spacing of 0.4 nm. DWDM requires a highly stable performance wavelength of the source and precise optical filters. Is important thermal stabilization, to prevent coverage of wavelengths. This total cost the traffic is growing and are much larger than CWDM [80].

4 Optical Amplifiers

Optical Fiber Amplifiers invented E. Snitzer in 1964 when he presented a neodymium-doped optical amplifier working at 1.06 μm. The fiber had a length of about one meter and was wrapped around lamps, which stimulated neodymium ions.

The most important type of these amplifiers is the Erbium Doped Fiber Amplifier (EDFA), which was invented in 1985. His invention is a revolution in optical communications because it allowed the replacement of 3R regenerators on lines limited by fiber attenuation. Enabled the creation of optically transparent networks. EDFA has become an optical amplifier for remote, multichannel digital and analog applications at 1550 nm. The neodymium doped fiber amplifier is used for the 1330 nm region. Fiber amplifiers are especially attractive because they offer high gain, low noise and no non-linear effects, but require an external pumping laser [53].

4.1 Properties of optical fibers

The transmission properties of the optical fiber are limited by two factors. Attenuation and dispersion. Attenuation leads to a decrease in signal power, which is a limiting factor for the transmission time. The dispersion causes an expansion of the optical pulse and limits the fiber bandwidth.

Because attenuation and dispersion increases with the length of the optical fiber is at a certain point on the optical path, the transmitted signal needs to be regenerated.

The optical fiber consists of a core and a shell. The fiber core has a higher refractive index than its shell.

Light is on fiber through the full reflection. Beam, which impinges on the kernel and shell interface under greater than the critical angle, is completely reflected back into the core and continues to the receiver. The fiber can only spread beams, which entered it at the right angle. The degree of fiber abilities to establish from your surroundings into the core of the optical beam, expresses a numerical aperture.

It is defined by the equation:

$$NA = n_0 \cdot \sin \varPhi_a, \tag{2}$$

where n_0 is the refractive index of the material from which the beam is bonded and $\varPhi_a$ is the maximum input angle.

4.2 Radiation detector

Radiation detector is one of the elements influencing the performance of the system. To achieve the required performance, so that the detector meets the essential requirements. It should exhibit high sensitivity over a wide band of wavelengths used for transmissions and as low a noise characteristic as possible. Furthermore, it should have very little sensitivity to temperature changes, low cost and long service life. The last important feature of the radiation detector is its dynamic range. The dynamic range specifies the minimum and maximum signal levels it can detect.

At present, two types of radiation detectors are most used, PIN and avalanche photodiode APD. PIN photodiodes have the advantage of low cost and higher reliability while the APD diodes exhibit higher sensitivity and accuracy. Apart from higher prices, APDs have a disadvantage in higher current demands and thermal sensitivity [51].

4.3 Types of optical amplifiers

There are several types of optical amplifiers. The first type is Fiber Optic Fiber Amplifier (OFA), other types are Semiconductor Optical Amplifiers (SOA). The most widespread type are amplifiers from the first group, but there is also a growing interest in using SOAs in optical networks and optical signal processing devices [52].

4.4 Amplifiers Properties

Reinforcement in optical amplifiers is through stimulated emission, with the same principle used in lasers. It is therefore possible to consider similar characteristics such as profit, spectrum, bandwidth, etc. An important parameter of the amplifiers is the noise generated by the amplifier.

4.4.1 Noise Amplifier

The signal-to-noise ratio of SNR in optical amplifiers is degraded by spontaneous emission, which adds noise to the signal during its amplification. This noise is defined by:

$$F_n = \frac{SNR_{in}}{SNR_{out}}, \tag{3}$$

SNR refers to the electrical power generated when the signal is converted into an electric current using a photodetector. We model Fn with respect to the ideal detector limited only Shot Noise:

$$SNR_{in} = \frac{I^2}{\sigma_2^s}, \tag{4}$$

where $I = RP_{in}$, the average photo detector is $R = \frac{q}{hv}$ is the sensitivity of the ideal detector with the unit quantum efficiency, $\sigma_2^s = 2q(RP_{in})\Delta f$ is the deviation from the shot noise and Δf is the bandwidth detector. Output is generated by the spontaneous emission of the receiver:

$$S_{sp}(v) = (G-1)n_{sp}hv. \tag{5}$$

Here, S_{sp} expresses the spectral density of the spontaneous emission noise, v is the optical frequency and n_{sp} is a spontaneous emission factor or a population inversion factor. The n_{sp}

value for amplifiers with a complete population inversion is equal to $n_{sp} = 1$ and $n_{sp} > 1$ for incomplete population inversion. For two-level systems, the following equations apply:

$$n_{sp} = \frac{N_2}{N_2 - N_1}, \tag{6}$$

where N_1 and N_2 are the atomic populations in the lowest and highest energy. Total deviation shot noise and spontaneous emission noise is then given by the relationship:

$$\sigma^2 = 2q(RPG_{in})\Delta f + 4(RPG_{in}) \cdot (RS_{sp})\Delta f. \tag{7}$$

Any other noise noise of the receiver is negligible. At the output the SNR is amplified the signal given by the relationship:

$$(SNR)_{out} = \frac{I^2}{\sigma^2} = \frac{(RPG_{in})^2}{\sigma^2} \approx \frac{GP_{in}}{4S_{sp}\Delta f}, \tag{8}$$

assuming that $G >> 1$ and neglect of the first member in formula 2.5 by using the definition of Fn, gain

assuming $G >> 1$ can be used by the definition of F_n, it is possible gain:

$$F_n = 2n_{sp}\frac{G-1}{G} \approx 2n_{sp}. \tag{9}$$

This shows that even for an ideal amplifier ($n_{sp} = 1$) the amplified signal is degraded by a factor 2 (3 dB). In practice, the F_n value ranges from 6 to 8 dB.

4.5 Semiconductor Optical Amplifiers (SOA)

First study on the subject semiconductor optical amplifiers were conducted at the time of the semiconductor laser invention in the 1960s. The first device was based on GaAs homophones operating at low temperatures. The arrival of devices using double heterostructures has prompted the development of SOA in the area of optical communication systems. In the 1970s and 1980s, advances in SOA design and production were made. Initial studies focused on AlGaAs SOA operating in the band 830 nm. In the later years, studies have appeared on InP/InGaAsP SOA designed for operation in the 1300 nm and 1550 nm regions. In 1989, SOA with a more symmetrical waveguide began to be produced. This has resulted in a much lower sensitivity to polarization. SOA development and design continues with advances in the manufacture of semiconductor materials, device designs, anti-reflective surface technologies, and photonic integrated circuits. Now there are reliable and cost-competitive, capable of deploying in commercial optical communication systems.

4.5.1 The principle of optical amplification

In SOA they are electrons (more commonly referred to as carriers) from an external current source are injected into an active environment. These carriers take up energy states in the conductive material web active environment and leave holes in the belt. There are three radiating mechanisms in the semiconductors.

With stimulated absorption, the incident photon may have sufficient energy stimulate the carrier from the valence to the conductive belt. This is considered a loss process as the incident photon is extinguished.

If a photon of sufficient energy reaches the semiconductor, may cause stimulated recombination of the carrier from the conducting band and holes of valence. The recombinant carrier releases energy in the form of a photon. This new photon is in all aspects (phase, frequency, direction) identical to the original photon. Both photons can continue to stimulate further transitions. If injected current sufficiently large, there is a population inversion where the number of carriers in the conduction band is greater than in the valence. In this case, the probability of stimulated emission is greater than stimulated absorption, and the semiconductor will produce optical gain.

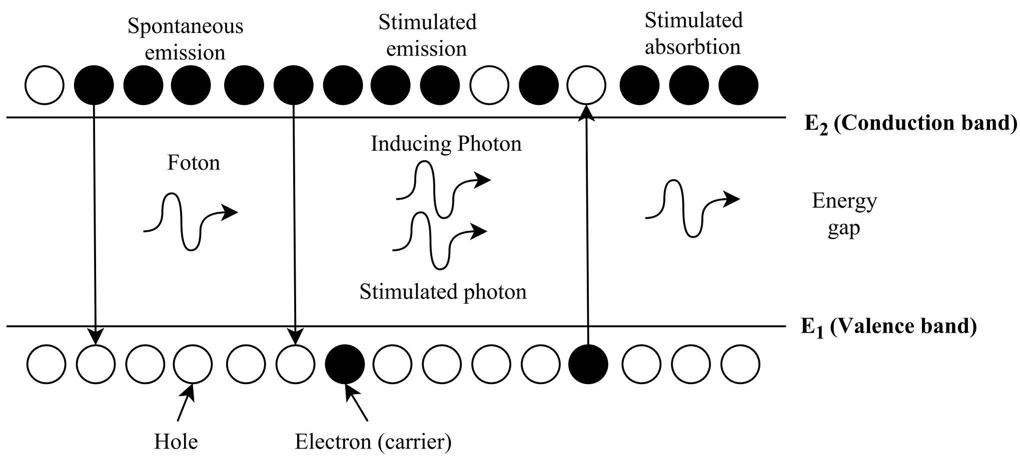

Figure 16: Two-level system of semiconductor material process

In spontaneous emission process the carriers are recombined spontaneously and random-phase and propagation photons are released. These photons are noise and reduce the number of carriers available for optical gain. Spontaneous emission is a direct consequence of the amplification process and can not be avoided, so it is not possible to create without noisy SOA. Stimulated processes are proportional to the intensity of the exciting radiation, and the spontaneous processes are independent of it.

The higher energy Conduction Band (CB) is represented as energy level like index E1 (on the figure 16) and the lower energy Conduction Band (CB) is represented as energy level like index E2 (on the figure 16). They are separated by an energy gap. Energy is released in the form of a photon when there is recombination of pairs of electrons with a higher energy band (CB) and energy Valance Band (VB) [54] [70].

4.5.1.1 SOA pump area

The area with semiconductor amplifiers has been one of the fastest growing in recent years. The typical use of SOA amplifiers is shown in Figure 17. The signal from the transfer fiber is fed to the SOA input of the amplifier where it is amplified. After amplification is redirected back to the fiber. The connection between the fiber and the SOA amplifier is via appropriate optical bond members [99].

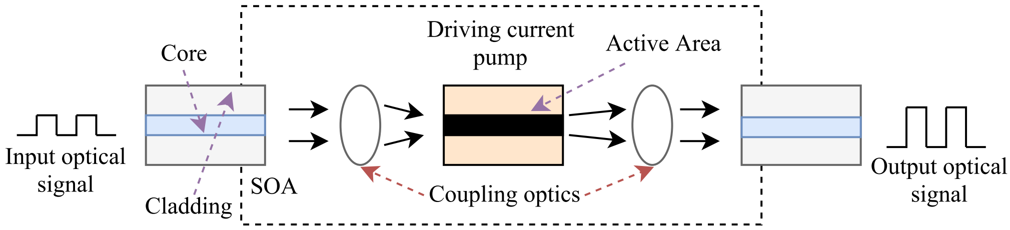

Figure 17: Configuration of SOA Amplifier [99]

4.5.2 Cross Phase Modulation

The cross phase modulation (XPM) uses interferometer configuration to convert the modulation phase to amplitude modulation. This configuration is used for conversion wavelength, demultiplexing, switching and for clock recovery.

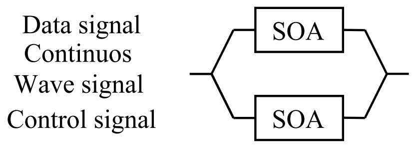

Figure 18: Configuration for single pump FWM [90]

The schematic of the Mach-Zehnder Interferometer using SOA is shown in Figure 18. First upper signal is injected the input signal beam and the control signal is injected into the last. The continuous signal is interconnect with each interferometer. A phase shift in the signal data and control signal is caused by the continuous signal. The first signal arm is a phase shift according to the strength of the signal. These signals are then aggregated and interfered at an output junction [89] [90].

4.5.3 SOA properties

The properties are semiconductor optical amplifiers are suitable for a wide range of applications such as signal processing, pulse shaping and recovery, optical routing, dispersion equalization, or wavelength conversion in WDM systems.

The SOA features five basic parameters:

- gain,
- gain bandwidth,
- saturation power,
- noise number,
- polarization dependency.

4.5.3.1 Gain SOA should have as much profit as possible for its area of use. The optical gain depends not only on the frequency (or wavelength) of the incoming signal but also on the intensity of the local beam at each point of the amplifier.

4.5.3.2 Noise number Semiconductor amplifiers have the same theoretical value of added noise as EDFA, but actually produce more noise. This is mainly due to internal losses and greater losses in connection at the input.

4.5.3.3 Dependence on polarization The undesirable feature of semiconductor amplifiers is their sensitivity to the polarization of the input signal. In optical communication systems, polarization can not be determined in advance because it changes during propagation. This makes the gain amount dependent on the polarization state of the incoming beam. Special procedures for the design and manufacture of semiconductor amplifiers reduce polarity sensitivity to less than 1 dB.

4.5.4 Non-linear phenomena in SOA

Nonlinear phenomena in semiconductor amplifiers are caused by changes in carrier density caused by them input signal. The three main nonlinear phenomena are: Cross 40 Phase Modulation (XPM), Cross-Modulation Profit (XGM), and Four-Wave Mixing (FWM). On the other hand, these nonlinear phenomena can be used to extend the functionality of semiconductor amplifiers. These amplifiers can in the optical network perform wavelength converter, multiplexer, or optical switch.

In Table 6 are the basic parameters of all three groups of optical amplifiers.

Table 6: Comparison of the properties of optical amplifiers

	SOA	**EDFA**	**Raman**
Gain [dB]	>30	>40	>25
Wavelength [nm]	1280-1650	1530-1560	1280-1650
Bandwidth (3dB)	60	30-60	1280-1650
Max. saturation [dBm]	18	22	0.75 x pump
Sensitivity to polarization	yes	no	yes
Noise number [dB]	8	5	5
Pumping power	<400 mA	<25 dBm	>30 dBm
Cost	low	medium	high

4.5.5 Types of SOA Amplifiers

The SOA amplifier is very similar to semiconductor lasers. There are two category of SOA amplifiers FP-SOA and TW-SOA.

FP amplifiers achieve high gain, but non-uniform amplification in the wavelength spectrum, while the TW semiconductor amplifiers have broadband gain with lower gain value. FP amplifiers have both ends the cavity has a high degree of reflectivity, resulting in resonance gain and consequently to a large gain at wavelength corresponding to Fabry-Perot's longitudinal cavities.

TW-SOA amplifiers have very low reflectivity due to the anti-reflection layer, resulting in amplification over a wide spectrum of wavelengths with low gain variability, which is caused by residual surface reflectance. It is much more suitable for deployment in systems, but the gain must be polarized independent [99].

5 Triple Play

Triple Play is presented as a package that includes three services - IPTV, VoIP and Data.

These three services are distributed to the end user through a network via a single socket/connection. Triple Play service means an end-user advantage especially in its simplicity of engagement. This means that it does not have to deal with services from multiple providers but has only one that distributes IPTV, VoIP and data services.

The services in the Triple Play package (Data, IPTV, VoIP) are spread over the network using the Internet Protocol (IP), located at 3rd layer ISO/OSI model. Transmission Data Operators use the Transmission Control Protocol (TCP), located on the 4th layer (transport layer) of the ISO/OSI model. This protocol is used because of its high reliability of data transfer. It works in such a way that if data is lost, the data will be re-query-ed and resubmitted. For Multimedia Services IPTV and VoIP, User Datagram Protocol (UDP) is also used, which is also found on the 4th OSI/ISO model. This is a non-conspicuous and unreliable protocol. It works in such a way if data is lost, it will not be reclaimed and the lost data will be sent. This means that it does not happen to a lengthy wait for re-sending lost data [60].

Depending on the provider we divide the services into:

- Interactive services: source information is sent at customer request (VoD).
- Distribution networks: same source information at the same time spread to a larger number of users where the individual participant is unable to change the time or content of the television and radio broadcast.

By type of access network we can share services by:

- metallic lines: ADSL, ADSL2,
- fiber optics: FTTH,
- optical-metallic lines: FTTB + ADSL.

5.1 QoS

Quality of service is the ability of a network to provide services with predictable performance. Packet-switching technologies are very flexible and can use the network infrastructure very efficiently, unlike circuit switching networks. The disadvantage is the complex predicting of their performance in the area of delay resulting from waiting in the queue, or time spent waiting for processing.

Compromise between performance and efficiency limits for network technologies arises when one converged network must support interactive and non-interactive operation.

This is exactly what happens with Triple Play applications. An important goal of the provider is to establish services on a single network that can transmit voice, video and data [55].

Important in QoS provision is to find parameters that can quantify and compare network performance.

These parameters are:

- Packet Loss - this parameter specifies the average number of lost packets over a given time expressed in % in proportion to the total number of transferred packets.
- Bandwidth - This parameter specifies the transmission channel capacity (Mbit/s, Kbit/s) and is the default parameter for the service offer.
- Jitter - This parameter represents the variability in packet delivery to the target device (i.e., the transmission delay). For IP set-top-box causes overflow or overflow buffer.
- Delay - This is a variable-length delay and occurs when packets are loaded into a queue on the outbound interface.
- Latency - This parameter represents the time that elapses since the message was sent to the source device after it was received on the received device.

Table 7: Basic service performance by service class (Metro Ethernet)

Service class	**Characteristics of the service**	**CoS ID**	**Basic operational properties**
Premium	IP telephony in real time or IP video apps	6, 7	Delay <5ms Jitter <1ms Loss <0.001 %
Silver	Transferring important data files with the fluctuating load required low losses and little delay	4, 5	Delay <5ms Jitter <N/S Loss <0.01 %
Bronze	Data transfer with fluctuating profileload and the need to guarantee a certain widthband	3, 4	Delay <15ms Jitter <N/S Loss <0.1 %
Standart	Best effort service	0, 1, 2	Delay <30ms Jitter <N/S Loss <0.5 %

5.2 IPTV

Internet Protocol Television (IPTV) or high-speed television is one of the Triple Play services provided. In addition to television broadcasting, IPTV can also be presented as video on Demand (VoD) or audio content (radio).

One of today's biggest benefits of IPTV solutions is that the recipient of IPTV is no longer just a passive recipient of TV broadcasts, as is the case in many of today's ways of distributing TV broadcasts. IPTV brings much more interactivity than it has been so far, and together with more interactivity, other services are coming through which IPTV has a great future.

IPTV providers have a clear overview of how many people are looking at each channel. IPTV provides the ability to individually personalize the broadcast content. This option is not available for classic TV channels, and all recipients are always broadcast the same content [18] [17] [23].

5.2.1 Protocols, codecs and services are used in IPTV

Codecs and protocols are based on the IPTV model Tab. 8, which is created on the ISO/OSI Reference Model. Like the OSI model, the IPTV model consists of 7 layers and each layer of the model has its own task. The principle is gradual, so if one layer accomplishes its task, it sends the data to the next layer.

Table 8: Layer IPTV model

OSI Model	**IPTV Model**
Application	Video/audio (services)
Presentation	PES (interface)
Session	MPEG-TS
Transport	RTP, UDP
Network	IP
Data Link	MAC
Physical	Physical

Before the video stream is sent to the network, it is necessary to modify the video signals for transmission in the data networks and then transfer them to the user. The video input signal can be analog or digital. Analog signal is further digitized by the encoder and then compressed using suitable (MPEG2, MPEG4/H.264, Windows media). This creates a data stream (video/audio) that is divided into small blocks of PES (Packetized elementary stream).

Data blocks are provided with a slide header and a data block header. Each part of the basic packet flow is 188 bytes in size. This is associated with the MPEG-TS transmission stream. Up to 7 blocks can be inserted into one Ethernet frame.

This processed video signal enters four OSI model layers (transport, network, link, and physical). These four layers are used to further encapsulate the transmitted video signal and transfer it between the source (video server) and the end users.

We can divide the service into three main categories:

1. Direct Broadcast – shows the program the TV station is currently broadcasting,
2. Broadcast with shifting – the broadcast program does not synchronize with the TV broadcast. The user can watch programs from the archive, or they can play the currently broadcast program from the beginning,
3. Video on Demand (VoD) – is a streaming service that gives viewers the choice of what they will be watching on TV or on a computer or mobile. It is not necessary to look at the prescribed television program.

Three protocols: UDP, RTP, and RTPS are usually used to transfer images over IPTV. The used broadcast codecs are typically MPEG-2 and MPEG-4, the new codecs are H.264 and VC-1. IPTV broadcasting is limited to packet loss, even a small percentage of lost packets will cause image quality to deteriorate. This also connects the bandwidth requirements in the network. Codecs have different bandwidth requirements and different data streams. The overview is in the following Tab. 9 [19] [17].

Table 9: IPTV transmission

Broadcast type	SDTV		HDTV	
Compression type	MPEG-2	H.264/MPEG-4	MPEG-2	H.264/MPEG-4
Transmission speed	4-7 Mb/s	2-3 Mb/s	18-20 Mb/s	720p - 5-7 Mb/s 1080i - 8-14 Mb/s 1080p - 22 Mb/s

5.2.1.1 Compression with Audio and Video Codecs

There are a number of codec specifications for audio-visual signals that define encoding, compression and transmission of audiovisual content. The most popular family is the MPEG (Moving Pictures Expert Group), defined by ISO/IEC and ITU-T. Windows Media, which is developed by Microsoft, is also an interesting codec that is being considered by many new IPTV operators.

IPTV signal compression enables better bandwidth utilization by reducing video files using mathematical algorithms. Compression methods use primarily human imperfections (the human eye can not reveal all graphic patterns).

Compression methods use human observer and audio deficiencies. It uses this fact using mathematical algorithms. For example, the human eye can not recognize all of the image patterns. For this reason, compression reduces the size of the original signal by removing these parts of the image.

The compression level applied to video content is measured as a numeric expression. For example, a compression ratio of 100:1 means that the size of the original content has been reduced by a factor of 100. The rule is that the quality of the resulting video content is often reduced when increasing the compression ratio. Compression methods can be divided into two categories, without loss and loss.

The most commonly used data compression methods for IPTV are [61]:

- **MPEG-1** (ITU-T H.261), published in 1993, was the first codec for the standard for digital video. It enabled development from analog to digital, regardless of analogue standards. Sound and motion are encoded at a rate of about 1.5 Mbit/s, ensuring video resolution matching VHS cassettes.

- **MPEG-2** (ITU-T H.262), published in 1995, is an extension of MPEG-1, providing a broader bandwidth of 2 to 20 Mbit/s, several levels of quality and screen resolution. MPEG-2 applications are used in satellites, DVDs and the first IPTV implementations.
- **MPEG-4** (ITU-T AVC/H.264), published in 1999, is a very flexible codec that provides a transmission rate ranging from 5 kbit/s to 10 Mbit/s, making it suitable for mobile video, standard definition and high-definition TV. MPEG-4 can save up to 50% of bandwidth, so new IPTV providers choose it. Existing MPEG-2 applications are slowly being moved to MPEG-4.
- **SMPTE VC-1** (WM9V) is a video codec specification that has been standardized by SMPTE (Society of Motion Picture and Television Engineers) and implemented by Windows Media 9. It has similar features to MPEG-4 but provides seamless integration with PC or hybrid PC-TV devices.

The standards are used to compress data and compress the data stream. Newer standards are suitable for encoding progressive and interlaced video. The latest MPEG-4 standard for high compression is interactive, with high compression, supports both real and synthetic objects. The line speed is around 4.8 - 64 kbit/s and for film applications up to 4 Mbit/s. MPEG 4 contains many coding algorithms that are based on the division of the scene into so-called image objects. Separate encoding of image objects allows easy manipulation of selected image information. The image and sound are encoded separately, which provides the advantage of decoding [61].

5.2.1.2 Internet Group Management Protocol (IGMP) is a communications network protocol that allows users to join or leave a multi-cast group, so they must be implemented by all users wishing to receive multi-cast information.

IGMP communication takes place between guests and the local (first) multicast router. If a guest wants to become a member of a multicast group, the IGMP sends a message to a local multicast router that monitors IGMP messages and maintains the current routing table.

The multicast router regularly sends a general query to hosts on the network, to see if there is at least one station in the LAN that wants to get information from the group. If no one wants to be a member of a multicast group, it will delete it from the routing table. There are currently 3 versions of the IGMP protocol.

IGMP snooping is a mechanism to optimize load on L2 switches. Without this mechanism, multicast is spread on the switch as well as the broadcast where the signal is forwarded to all ports except the port from which it came. This solution raises the network load because behind each port there may not be a user who needs to receive the information (multicast group). IGMP snooping monitors multicast traffic, detects "join" and "leave" messages to determine which ports the client is, which requires a given traffic. Specifies the ports behind which the router is located and sends client responses to routers only and does not send them to other clients. Based on the above findings then compiles a table, there is a dynamic configuration of the multicast forwarding port ports.

IGMP messages are divided into 3 versions. Version 1 IGMPv1 defined in RFC 1112. Version 2 IGMPv2 defined by RFC 2236. The latest version of IGMP is version 3 (RFC 3376 later replaced by RFC 4604), the format of which is shown in fig. 19. This version has added the "resource filtering" option, which allows you to specify the INCLUDE and EXCLUDE flag. This principle is shown in fig. 21 [24].

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  Type = 0x11  | Max Resp Code |           Checksum            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                         Group Address                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| Resv  |S| QRV |     QQIC      |     Number of Sources (N)     |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                       Source Address [1]                      |
+-                                                             -+
|                       Source Address [2]                      |
+-                              .                              -+
.                               .                               .
.                               .                               .
+-                                                             -+
|                       Source Address [N]                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 19: IGMPv3 format [6]

On fig. 19 is The **Max Response Time** field is used only in Membership Query messages. It specifies the maximum allowed time before sending a responding report in units of 1/10 second. In all other messages, it is set to zero by the sender and ignored by receivers. **IGMP Checksum**. When the data packet is transmitted, the checksum is computed and inserted into this field. When the data packet is received, the checksum is again computed and verified against the checksum field. If the two checksums do not match then an error has occurred. **Group Address** in a Membership Query message, this field is set to zero when sending a General Query, and set to the group address being queried when sending a Group-Specific Query. In a Membership Report or Leave Group message, this field holds the IP multicast group address of the group being reported or left. **Resv** is reserved. It should be zeroed when sent and ignored when received. **S** (Suppress Router-side Processing) Flag when this flag is set, it indicates to receiving routers that they are to suppress the normal timer updates. **QRV** (Querier's Robustness Variable) if this is non-zero, it contains the Robustness Variable value used by the sender of the Query. Routers should update their Robustness Variable to match the most recently received Query unless the value is zero. **QQIC** (Querier's Query Interval Code) this code is used to specify the Query Interval value (in seconds) used by the querier. If the number is below 128, the value is used directly. If the value is 128 or more, it is interpreted as an exponent and mantissa. The **Number of Sources** (N) field specifies how many source addresses are present in the Query. This number is zero in a General Query or a Group-Specific Query, and non-zero in a Group-and-Source-Specific Query. The **Source Address** [i] fields are a vector of n IP unicast addresses, where n is the value in the Number of Sources (N) field [25] [26] [31].

5.2.1.3 Real-time Transport Protocol (RTP) is a protocol on the fourth layer, providing a packet format for sending video and audio content over a packet network. This protocol has been developed by the IETF (Internet Engineering Task Force) as standard RFC 1889, which was later replaced by the RFC 3550 standard. It most often uses the UDP protocol on ports 5004, 5005 and 6970.

```
  0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|X|  CC   |M|     PT      |       sequence number         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           timestamp                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           synchronization source (SSRC) identifier            |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|            contributing source (CSRC) identifiers             |
|                             ....                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 20: RTP packet header [7]

Fig. 20 shows the header of the RTP packet. The first 12 bytes are required. The first field in the header indicates the version of the protocol, the second field labeled P (Padding) is used to indicate whether additional additional bytes are used after the header, which are used, for example, for fixed block size encryption algorithms. The X (Extension) field is set if another Extension header is added. The CC (CSRC Count) field shows the number of CSRC identifiers behind the fixed header. M (Marker) is used to indicate that current data is of special importance for the application layer. PT (Payload Type) indicates the payload format and determines its application interpretation (for example, RTP profile for audio and video conferencing with minimal control). SN (Sequence Number) is a number that incrementally increases with each other RTP packet and serves to re-sort the packets as they were originally sent.

Since some packets can not be delivered (RTP does not send undelivered packets again), each application is retained for empty space (for example, the video application may display the last frame received instead of the missing frame). The TS field in the RTP header (Time Stamp) expresses the moment of removing the sample of the first byte of useful content.

SSRC (Synchronization source identifier) identifies the sync source. If the CSRC count is zero, the source of useful content is the source of synchronization. CSRC (Contributing source) identifies resources contributing to useful content. The number of contributing resources is determined by the number of CSRC (CC). In total, there may be 16 contributing sources. If there is more contributing resources, the resulting useful content is by merging these resources.

The RTP protocol has no mechanism how to find out if the packet has been delivered, if it was delivered in time, it is mostly used together with the protocol RTPC (Real-Time Transport Control Protocol), which is a real-time control protocol [28] [27].

5.2.1.4 Real Time Streaming Protocol (RTSP) is defined by RFC 2326 and used to deliver content in the form of a unicast data stream. This is an application layer protocol, which has been developed specifically to manage real-time data delivery, such as audio content or video content. It is implemented through a bug fix transport protocol [29].

5.2.1.5 Protocol Independent Multicast (PIM) is IP routing protocol-independent and can leverage whichever unicast routing protocols are used to populate the unicast routing table, including Enhanced Interior Gateway Routing Protocol (EIGRP), Open Shortest Path First (OSPF), Border Gateway Protocol (BGP), and static routes. PIM uses this unicast routing information to perform the multicast forwarding function. Although PIM is called a multicast routing protocol, it actually uses the unicast routing table to perform the RPF check function instead of building up a completely independent multicast routing table. Unlike other routing protocols, PIM does not send and receive routing updates between routers [30].

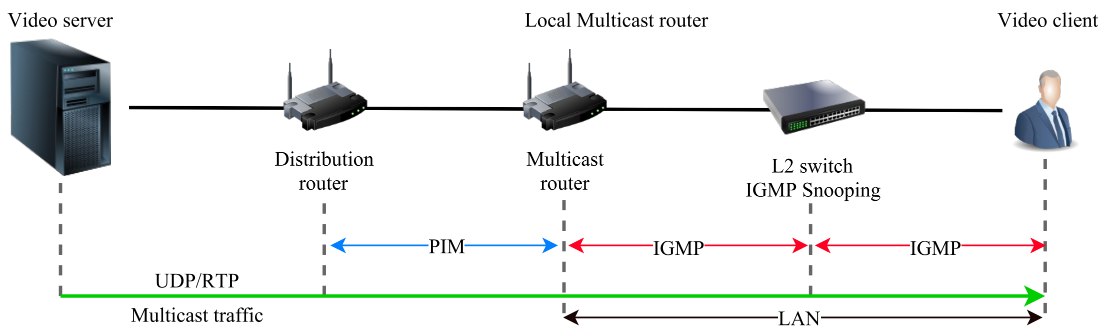

Figure 21: Example of a multicast network architecture

On fig. 21 is PIM works between multicast routers inside the network. PIM has two main modes of operation, the first being PIM DM (PIM - Dense Mode), which is defined by RFC 3973, the second mode is PIM SM (Sparse Mode) defined in RFC 2362.

PIM forwarding modes are described in the following sections:

- PIM Dense Mode (PIM-DM) so that this method is effective in the network, must be deployed there, where there is intense and constant multicast flow with a small number of resources and a large number of recipients. It is most commonly used on backbone networks, but in some cases it can be effective even in the distribution network. PIM-DM supports only source multicast distribution trees, shared trees in this mode can not be used. PIM-DM initially floats the multicast all over the network and then routers that do not have any recipients, they say they do not want to receive the multicast group, this process is repeated every 3 minutes.

- PIM Sparse Mode (PIM-SM) assuming, that the population of active users is significantly lower than the total population, is PIM-SM a better method. Unlike the previous PIM-DM method, when using the PIM-SM method, multicast data is only sent to subnets, who requested this data using the IGMP protocol. Since this method uses shared multicast trees, multicast data is sent from sources to RP (Rendezvous point) where the recipients of these data are registered.

5.2.1.6 Transmission Control Protocol (TCP) is the main protocol of the transport layer. This is a protocol that is link-oriented. When transferring data two-way connections are used between communicating stations. At the moment the application sends a stream of bytes, the TCP protocol divides this stream into reasonably large segments. The segment size is determined by the Maximum Transmission Unit (MTU). Segments created in this way continue sent to the bottom layers to be transmitted over the Internet. Once the data has been migrated, authentication and confirmation are verified. If the data sent does not match the received data, the data is re-sent. TCP protocol guarantees reliable delivery of segments in the right order. TCP uses many application protocols and applications, including websites, emails, and data transfer applications.

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|          Source Port          |       Destination Port        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                        Sequence Number                        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                    Acknowledgment Number                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  Data |           |U|A|P|R|S|F|                               |
| Offset| Reserved  |R|C|S|S|Y|I|            Window             |
|       |           |G|K|H|T|N|N|                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           Checksum            |         Urgent Pointer        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                    Options                    |    Padding    |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                             data                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 22: TCP Header Format [8]

On Figure 22 The **sequence number** of the first data byte in this segment. If the SYN bit is set, the sequence number is the initial sequence number and the first data byte is initial sequence number + 1. If the **Acknowledgment Number** bit is set, this field contains the value of the next sequence number the sender of the segment is expecting to receive. Once a connection is established this is always sent. **Data Offset** is number of 32-bit words in the TCP header. This indicates where the data begins. The length of the TCP header is always a multiple of 32 bits. **Flags** aka Control bits added RFC 3540, RFC 3168, RFC 3540 for example this 1 bit use FIN- Last packet from sender, RST- Reset the connection etc. **Window Size** are number of data bytes beginning with the one indicated in the acknowledgment field which the sender of this segment is willing to accept. **Checksum** is computed as the 16-bit one's complement of the one's complement sum of a pseudo header of information from the IP header, the TCP header, and the data, padded as needed with zero bytes at the end to make a multiple of two bytes. **Urgent Pointer** If the URG bit is set, this field points to the sequence number of the last byte in a sequence of urgent data. **Options** occupy space at the end of the TCP header. All options are included in the checksum. An option may begin on any byte boundary. The TCP header must be padded with zeros to make the header length a multiple of 32 bits [32].

5.2.1.7 User Datagram Protocol (UDP) is a simple ISO / OSI transport layer protocol for client/server application based on IP. UDP is used in real-time applications that do not require reliable transmission. When transmitted, some data may be damaged, lost, or arriving in the wrong order. These are apps, which do not require automatic re-sending of damaged or lost data, only in case they are about their delivery request themselves. UDP, unlike TCP, does not establish a direct connection between the communicating stations and does not use the transferred data control. This means faster transmission at the expense of reliability. UDP is used for VoIP, DNS, or various Internet video conferencing. Data units that are created by UDP are called diagrams.

```
 0              15 16             31
+--------+--------+--------+--------+
|     Source      |   Destination   |
|      Port       |      Port       |
+--------+--------+--------+--------+
|    Checksum     |                 |
|    Coverage     |    Checksum     |
+--------+--------+--------+--------+
|                                   |
:              Payload              :
|                                   |
+-----------------------------------+
```

Figure 23: UDP-Lite Header Format [9]

On fig. 23 you can see **Source Port** the port number of the sender. Cleared to zero if not used. **Destination Port** the port this packet is addressed to. The **length** in bytes of the UDP header and the encapsulated data. The minimum value for this field is 8. **Checksum** computed as the 16-bit one's complement of the one's complement sum of a pseudo header of information from the IP header, the UDP header, and the data, padded as needed with zero bytes at the end to make a multiple of two bytes. If the checksum is cleared to zero, then checksuming is disabled. If the computed checksum is zero, then this field must be set to 0xFFFF [33].

5.2.1.8 Real Time Streaming Protocol (RTSP) is used to control video in VoD. The data transmission itself is performed using a separate Real-time Transport Protocol (RTP) protocol. RTSP is standardized in RFC 2326. This is a client/server control protocol, where both clients and server can send requests, but their behavior is different. RTSP does not create a connection but creates a session. The difference between the connection and the session is that transmission at the connection is continuous, while the session may be inactive for a long time [35].

5.2.1.9 Real Time Control Protocol (RTCP) complements RTP by providing feedback on transmission quality. This is the primary function of RTCP and it transfers permanent identifiers of RTP resources. RTCP packets are multiplexed together with RTP packets. RTPs usually use even ports and RTCPs of neighboring, higher, odd ports [34].

5.2.2 IPTV architecture

The IPTV architecture can be divided into the following parts [21] [22] [82]:

- Head-end,
- distribution network,
- access network,
- digital home network.

5.2.2.1 Head-end receives content such as TV programs or radio stations from various sources such as terrestrial, satellite or cable broadcasts. There are coding, compression and multiplexing of programs. They are then sent in IP packets using the most appropriate physical interfaces. Video and audio content is compressed by an MPEG codec that defines parameters such as pixels per line, frame refresh. Programs are broadcast over the IP network in multicast or unicast address mode. These are mainly DVB-S, DVB-S2, DVB-T and DVB-C broadcasts.

5.2.2.2 Distribution network must send audiovisual signals over regional and metropolitan networks until does not arrive to the target customer. A high-capacity kernel using RTP or RTSP protocols must ensure that each packet, whether unicast or multicast, does not degrade parameter quality such as delay and loss. IP multicast is the method by which information can be sent to multiple users at the same time.

Distribution networks are IP-based and Ethernet-based. In distribution networks most often uses MPLS (Multi-protocol Label Switching) technology. MPLS is a packet switching method.

5.2.2.3 Access network covers the first mile and its role is to transfer data to end users. Available there are several technologies including ADSL2, ADSL2 +, VDSL2, FTTN, FTTH and WiMAX. Selection depends on parameters such as bandwidth, distance, and financial options. The switching feature is a network request, when the user communicates using the IGMP (Internet Group Membership Protocol) which program wants to track. From a technical point of view, this means that the user becomes a part of the multicast group that contains the selected program.

The access network has the task of distributing individual data flows to customers. QoS quality parameters play an important role here where we can, for example prioritize video transmission and prevent delay or fragmentation (division).

5.2.2.4 Digital home network IP packets they will get to the receiver as a video stream. For IPTV playback, you need a computer, a set-top box (STB) connected to a TV or another application for receiving on smart phones. Interaction between user and network is provided by IGMP protocol for channel selection for IPTV and RTSP protocol for VoD applications.

5.2.3 Unicast and Multicast communication

Communication in IPTV is mostly multicast. Broadcasting can also be done with Unicast, but this method is not used at all, as it would cause server overloading. Unicast is only used for some types of services offered by IPTV, such as VoD. That is, for a given end user that he has chosen from a movie or program, he is broadcast only on his IP address.

Multicast is a way to send packets on a network that greatly saves the backbone network bandwidth by sending only one copy of the media stream to the network. Media streams are copied and sent to individual subscribers who are logged into the multicast group.

Multicast address ranges [77]:

- 224.0.0.0 – 224.0.0.255

  These are addresses for network protocols that are used in the local area network (LAN), the TTL value is 1, so they do not pass over the router. For example, address 224.0.0.2 is reserved address, which identifies all routers connected to the local network.

- 224.0.1.0 – 238.255.255.255

  These addresses are global multicast addresses that are used in communication between organizations. They can also be used to multicast IP video content over the public Internet.

- 239.0.0.0 – 239.255.255.255

  his multicast range is used for IP group broadcasts that are limited to local groups or organizations.

Digital TV requires a large bandwidth (approximately 2-20 Mbit/s per channel). Larger television stations like CNN, HBO have several hundreds of thousands of viewers. If we wanted to distribute TV channels using the point-to-point mechanism to all viewers, we would need bandwidth in hundreds of Gbit/s. A much more effective way is to distribute video content via multicast, as can be seen in fig. 24, where it's left shown distribution of the same content via unicast to multiple recipients, and in the right part the same transmission is shown using multicast. The content of each channel is sent from the main station (head-end see 5.2.2 only once, regardless of the number of viewers [24].

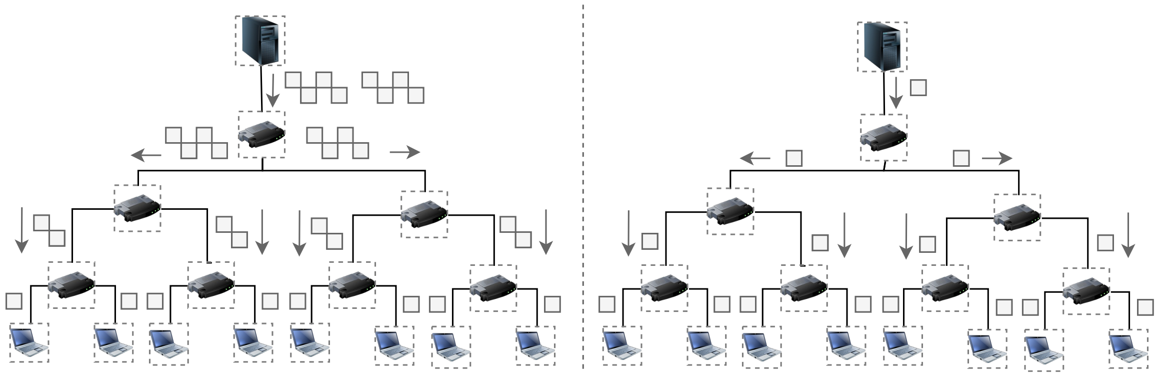

Figure 24: Unicast (left), Multicast (right) [24]

5.2.4 IPTV services

The basic IPTV service is the classic provision of channels. The advantage is that in IPTV you can customize the channel offer to individual users (eg charged channels). With an IPTV set-top box, the end user can communicate with the provider, who provides him with the services he wants. Telecommunications companies using conventional television broadcast only provide linear subscribers to their subscribers. IPTV provides access to a wide range of nonlinear services, ie programs or videos that are not broadcast according to a pre-set timetable.

5.2.4.1 Electronic Programming Guide (EPG) This is a program guide that is implemented as IPTV services and also allows different types of content to search for a specific genre, title, or keyword. The EPG is based on the client/server principle where the client part is located in the IPTV set-top box. First, the recipient on the driver. The STB enters the start of the program guide, then the STB establishes a connection with the server, which subsequently sends the requested data (using HTTP), which includes, for example, a station program [36].

5.2.4.2 Video on Demand (VoD) is a streaming service that gives users a choice of what they will watch on TV. It is not necessary to look at the prescribed television program, but they can choose to choose programs or different movies when they want to. Nowadays, there are many virtual movie rental companies that offer individual films for a fee. You can pause movies at any time, or increase playback speed, for example. These services also provide IPTV VoD. The advantage over traditional virtual video rental is the option to choose a movie directly on your TV and you do not have to use a computer. IPTV VoD offers a selection of movies with a clear menu. These movies are usually stored on VoD servers. The IPTV provider sends the selected content to the user via unicast transmission [37].

5.2.4.3 Pay Per View (PPV) offers us watching certain programs for which we have to pay. These are programs that are not freely available to all users. The PPV service is mainly used to watch live programs, so it is not possible to determine the broadcast time [38].

5.2.4.4 Video Cassette Recorder (VCR) is a service similar to VoD where the recipient lets you record the selected TV program, the provider's server, or your STB (if that feature is included) and later play it [39].

5.2.4.5 Super teletext the service provides a superstructure of the classic teletext that was used in analogue broadcasting. Designed on a similar principle where a user can view a pre-prepared information database that is constantly being broadcast. The environment is in full graphic mode, and offers a host of other features that work with this graphics mode. The appearance of the environment is adapted to the parameters of the receiving TV [40].

5.2.4.6 Multimedia Home Platform (MHP) is an open middle-ware standard designed within the DVB for interactive digital television. This platform allows the reception and execution of interactive IPTV-based Java applications. Applications can be, for example, information services, games, interactive voting, email, SMS or online shopping [41].

5.2.4.7 My Own TV It offers the customer the ability to create recordings that include videos or photos. These works of art can be shared with other users.

5.2.4.8 Check account offers an account status report for a customer who uses paid IPTV services (e.g., VoD, PPV).

5.2.5 Encapsulate and package video content

Encapsulate and package video content includes and organizing video data into individual packets. The term encapsulation is used to describe the process of formatting video content into datagrams. There are many different approaches to encapsulating video content, such as MPEG.

Before video content is sent to the network, the video signal needs to be modified for transmission in the data networks and then transferred to the end user. The video input signal can be analog or digital. The analog signal is converted to digital form by the encoder and compressed by appropriate compression (MPEG-2, MPEG-4). This results in a continuous data stream, which is divided into small blocks of PES. The resulting blocks of data are provided with a block header and a data frame.

The size of each part of the basic packet stream is 188 bytes. Then merging into the MPEG-TS transmission stream. One Ethernet frame can hold up to 7 blocks. The processed video signal enters the lower four layers of the OSI model, which encapsulates the transmitted content and transfers it between the source (video server) and the end user.

5.2.6 Subjective methods of assessment

These methods are used by a user group to evaluate image quality, telephone connections, and more. They are based on the impression that the test video or audio leaves behind on the watch group. The advantage of this method is that it can describe how a service is perceived by a person, this can limit the service to imperceptible information that does not catch the same human senses. Subjective video quality assessment methods include MOS, DSQS, DSIS, ACR, and SSCQE. Tests typically take place in successive steps.

Subjective evaluations are often considered very useful in evaluating voice services. What is impractical is that this rating for each service is specific. Performance evaluation for audio is not useful for video or data applications [55].

Typical steps for subjective measurement of the quality of IPTV services are:

- Select a sample video to test,
- select several configuration parameters,
- set a test environment that meets the required configuration parameters,
- Assemble a group of people to perform the test,
- Perform the test and analyze the resulting values.

There are two ways for a subjective method: **conversational** and **listening** [55].

- Conversational Quality (CQ) - This Quality Score model is designed for laboratories where this simulation is performed. The principle is simple to connect two users to each other and makes a call to each other who independently evaluates the transmission quality. The third person in the lab evaluates the test conditions. The entire rating is governed by the MOS scale.

- Listening Quality (LQ) - This Quality Score model is much simpler than CQ. The principle is that the voice call being played, which is rated using methods such as the Absolute Category Rating (ACR) based on the MOS scale. Other ways are, for example, Quantile-Response Detectability Tests, DCR, CCR.

5.2.6.1 MOS MOS is defined in ITU-T Recommendation P.800. They rank in the Absolute Category Rating (ACR) because quality is rated without reference. This category is the opposite of the Degradation Category Rating (DCR), which is based on a comparison between the user-perceived quality and the reference signal.

It is possible to calculate a parameter similar to MOS, which instead of an absolute evaluation counts with a worsening rating. This parameter is called degradation MOS (DMOS).

First, a series of short videos are set up for the test. Then select a set of parameters that will be the subject of a sample selection by a select group of people (at least 18). After selecting the right group of people and the appropriate test environment, testing is run. This group of people then evaluates the number of impressions from the video according to the rating scale in Table 10. The resulting MOS value is composed of the average of the measured results [10].

Table 10: Mean Opinion Score (MOS)

Rating	**Quality**
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

The MOS is calculated as the arithmetic mean over single ratings performed by human subjects for a given stimulus in a subjective quality evaluation test. Thus:

$$MOS = \frac{\sum_{n=0}^{N} R_n}{N} \tag{10}$$

Where **R** are the individual ratings for a given stimulus by **N** subjects.

Variants for MOS rating:

- MOS-V – Video quality evaluation.
- MOS-A – Sound quality evaluation.
- MOS-AV – Video and audio quality evaluation.
- MOS-C – Impression of IPTV services interaction.

5.2.6.2 Double Stimulus Continuous Quality Scale (DSCQS)

This subjective method for evaluating video quality uses a pair of video sequences, referenced and tested. The observer has no idea which video sequence is reference and which is tested. This method uses a scale ranging from 0 to 100, which should be fed with words indicating quality. At least 15 evaluators of different age and professional groups are used for testing. The method is very time consuming.

5.2.6.3 Absolute Category Rating (ACR)

the method evaluates images or sequence of images. The viewer evaluates the image quality using a five-point scale within ten seconds.

5.2.7 Objective evaluation methods

These methods use mathematical calculations to determine image/video quality. Video quality is tested based on image compression with compressed version and degraded signal quality. Objective methods are simpler, faster and cheaper than subjective methods. The objective methods include MSE, PSNR, MDI MPQM, SSIM and others.

Objective methods are divided into two subgroups, which are intrusive and non-intrusive.

Intrusive methods are based on a comparison of the original signal with the signal transmitted through the telecommunication chain. These include PSQM, PAMS, or PESQ.

Non-intrusive methods are methods in which only the signal that has passed through the telecommunications chain is used to calculate the MOS value. You do not need to know the original signal. These include, for example, 3SQM, INMD, CCI.

Evaluation of MOS is complex, time and costly. This is the reason why in practice the direct evaluation of MOS is replaced by device and algorithmic estimates. Algorithmic assessment methods are said to be objective because they do not rely on opinions, but many try to predict the MOS, which is a parameter obtained by averaging individual views.

Depending on the input data, the algorithms for objective evaluation can be categorized into groups:

- Spectacular models - the input is different network performance parameters, resulting in a conversation MOS estimate. The most common model of this type is the E-model.
- Speech Layer Models - the input is speech signals and the result is an estimate of the listening MOS.
- Models on the packet layer - the inputs are IP packets and the output is the estimated listening MOS. They are used to monitor speech quality at mid points where speech signals are not available.

5.2.7.1 PSNR and MSE

The PSNR parameter represents the ratio between the highest signal value and the MSE parameter. It is given in dB. The objective MSE method expresses the mean quadratic deviation of the original signal from the captured signal.

The PSNR in decibels is defined as:

$$PSNR = 10 \log \frac{m^2}{MSE} [\text{dB}] \tag{11}$$

where m is the maximum value that pixel can take (e.g. 255 for 8-bit image) and MSE (Mean Squared Error) is the mean of the squared differences between the gray-level values of pixels in two pictures or sequences I and $\tilde{I}$:

$$MSE = \frac{1}{TXY} \sum_t \sum_x \sum_y \left[I(t,x,y) - \tilde{I}(t,x,y) \right]^2 \tag{12}$$

for pictures of size X x Y and T frames. I is original image, $\tilde{I}$ received image, x number of pixels in height of image, y number of pixels in width, T number of images.

Technically, MSE measures image difference, whereas PSNR measures image fidelity. The biggest advantage of the PSNR metric is easy and fast computing [63].

5.2.7.2 SSIM SSIM is a newer method that takes into account the human visual system. And by finding similarities between two images. This has helped to improve traditional metrics (MSE and PSNR) that are sometimes used in contradiction with human perception. The SSIM has reference values in the range from 0 to 1, where 0 represents a zero relation to the original image. Values 1 are obtained when matching the shapes.

SSIM is given by:

$$SSIM(x,y) = [l(x,y)]^{\alpha} [c(x,y)]^{\beta} [s(x,y)]^{\gamma} \tag{13}$$

Where: $l(x,y)$ compares signal brightness, $c(x,y)$ compares signal contrast and $s(x,y)$ measures structural correlation.

$$l(x,y) = \frac{2\mu_x \mu_y + C_1}{\mu_x^2 \mu_y^2 + C_1} \tag{14}$$

$$c(x,y) = \frac{2\sigma_x \sigma_y + C_2}{\sigma_x^2 \sigma_y^2 + C_2} \tag{15}$$

$$s(x,y) = \frac{2\sigma_{xy} + C_3}{\sigma_x \sigma_y + C_3} \tag{16}$$

Where: $\mu_x a$ μ_y means the average of the samples x and y, σ_x and σ_y means dispersion from the samples x and y [63] [55].

5.3 VoIP

Voice over IP is a service that delivers voice over Internet networks using RTP, UDP and TCP/IP protocols. VoIP also delivers other services in addition to voice transmission such as faxing, sending short SMS text messages, or voice messages.

IP telephony gives a new competitive advantage to new service providers compared to existing operators, because it provides more options for installing their services. Users benefit from growing competition from service providers. IP telephony has the ability to integrate voice and data into the same network infrastructure. This makes it easy and inexpensive to install and manage this service. It is acceptable for service providers, that earnings are reduced, but it opposes telephony lots of new opportunities through new services that enhance classic video calling, mobile and fixed line integration.

From the technical side, the main advantage of IP technology is the convergence of data and voice. Convergence can not be achieved without IP networks with QoS. Over the past few years, much effort has been spent on designing such networks. However, most of the current IP networks remain not guaranteed, including the Internet. The operator of this service is forced to be connected to a telephone exchange with a public switched telephone network (PSTN). Connection to this public network can be telephoned to other networks with much lower charges than standard telephone services.

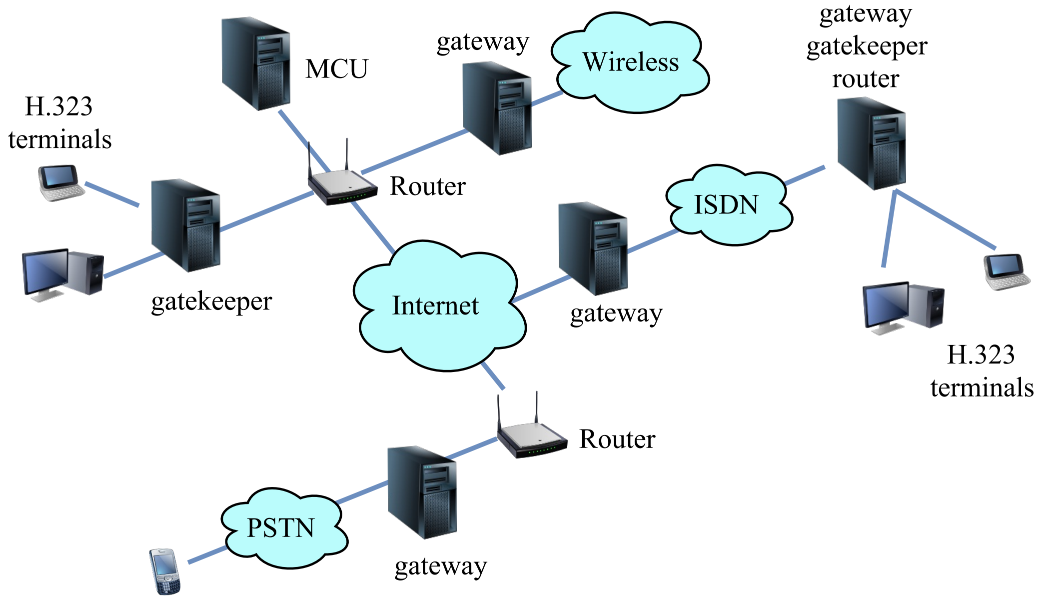

Figure 25: VoIP application scenarios [46]

On fig. 25 are elements of a VoIP network. **Terminals** LAN-based communication end-points. **Gateway** (media and/or signaling) Interface between packet- and circuit-switched networks, voice trans-coding, protocol conversion, call handling, call state and signaling mediation. **Gatekeeper** admission control, SNMP services, address translation and last **MCU** Multi-point Control Unit- Handling of broadcasts / conference calls [46] [83].

5.3.1 Used codecs

Codecs are generally understood to be the various mathematical models used by digital encoding and compression of analog audio information. Many of these models count on the ability of the human brain to create the right impression from incomplete information. Audio compression algorithms rely on the fact that, human perception has the tendency to interpret what we believe than what we really hear. The purpose of the various coding algorithms is to create a balance between efficiency and quality.

Voice calls are determined by VoIP codecs end devices. For voice transmission, we need to convert audio from analog microphone to digital format, with different codecs with different quality and difficulty on data stream. For a quick overview of the most used codecs, see Tab. 11 [42] [43].

Table 11: Overview of codecs used in VoIP

Codec	**Data speed [kbit/s]**	**License**
G.711	64	No
G.726	16, 24, 32, or 40	No
G.729A	8	Yes
G.723.1	5.3 or 6.3	Yes
GSM	13	No
iLBC	13.3 or 15.2	No
Speex	2.15 - 22.4	No

5.3.1.1 G.711 It is a basic PSTN (Public Switched Telephone Network) codec. If someone talks about PCM (Pulse Code Modulation) along with a telephone network, it's G.711. It uses two encoding methods. μlaw in North America and alaw in the rest of the world. It transmits 8 bits 8000 times per second, at a transmission rate of 64,000 bit/s. Compared to CD quality that encodes 16 bits at a sample rate of 44,100 Hz, this is a great concession to quality requirements.

5.3.1.2 G.726 This codec has been in the world for some time (now we can not really see it) and it is one of the original compression codecs. It is known under the name ADPCM. It operates at several sampling frequencies from 16 Kbit/s up to 32 Kbit/s.

G.726 provides almost identical quality to G.711, but only half bandwidth. This is because it only sends information about the description of the difference between the previous and the current sample.

5.3.1.3 G.729A Considering how much bandwidth it uses, the G.729A delivers surprisingly good sound. This is due to the use of the Conjugate-Structure Algebraic Code Excited Linear Predictions (CS-ACELP).

5.3.1.4 G.723.1

It is a codec with one of the best compressions when compressing a voice signal of length 30 ms with sampling frequency of 8 kHz. The G.723.1 codec is under patent protection and must be licensed for use. It uses two kinds of codes. MP-MLQ (MultiPulse Maximum Likelihood Quantization) - 6.3 kbit/s with MOS 3.9 and Algebraic Code Excited Linear Prediction (ACELP) - 5.3 kbit/s with MOS 3.65.

5.3.1.5 GSM

It provides similar features to the G.729A with the difference that it is free and usable. It operates at 13 Kbit/s.

5.3.1.6 iLBC

The Internet Low Bit-rate Codec provides an attractive mix of bandwidth quality and bandwidth usage. It was very good to use on lossy network lines. The iLBC uses a complex of algorithms to achieve a high degree of compression. This is a Global IP Sound patent, you just need to register for use.

5.3.1.7 Speex

Speex is a codec that can change the sampling rate according to network conditions. This is a free product distributed under the GNU license. Speex operates somewhere at a transfer rate of 2.15 to 22.4 Kbit/s.

5.3.2 Protocols used

Two protocols are mainly used for VoIP transmission H.323 and SIP. Another example RTP, RTCP, and more. The main protocols hierarchy is shown in tab. 12 [45] [44].

Table 12: The main protocol hierarchy in VoIP

Signaling	**Samples**	**Signaling**	**Samples**
H.323	RTP	SIP	RTP
TCP	UDP	SIP	RTP
IP		IP	
Protocol H.323		Protocol SIP	

5.3.2.1 SIP

The Session Initiation Protocol (SIP) has been developed since 1996 and was presented as a design standard RFC 2543 in 1999 and was immediately taken up by its simplicity. In 2002, standard RFC 3261 was issued, which contains the core of the SIP protocol, which is still used today. Specifies 6 methods - INVITE, ACK, BYE, CANCEL, REGISTER, and OPTIONS. This protocol is not as simple as at the time of its creation. There are many more extensions covered by more than 80 RFC standards.

SIP is a signaling protocol that allows you to set up, modify, and end a session with one or more subscribers. It is very similar to the protocol used in Internet communications like HTTP or SMTP, so there is also a way of communicating based on the request/answer architecture.

Session Description Protocol (SDP) is used to describe the session properties and the voice itself is transmitted using the Real-Time Transport Protocol (RTP) [86].

To create and manage a multimedia session, SIP must provide 5 activities:

- location of the subscriber ,
- status finding subscriber ,
- Finding Subscriber Options - Encoded Codecs, etc.,
- custom link establishment,
- control the ongoing connection.

SIP responses are the codes used by Session Initiation Protocol for communication. In total, there are 6 applications:

- 1xx = Informational SIP Responses
- 2xx = Success Responses
- 3xx = Redirection Responses
- 4xx = Request Failures
- 5xx = Server Errors
- 6xx = Global Failures

5.3.2.2 H.323 is a multimedia conference protocol that includes voice, image, and data transmissions on packet switching networks. He was appointed ITU-T. Today, H.323 is the most widely used voice and video conferencing protocol in IP networks. The latest version of the H.323 protocol is H.323v6.

The basic elements of the network where the H.323 protocol is used are:

- terminals,
- Multi-point Control Unit (MCU),
- gateways,
- gatekeeper,
- border elements.

The terminal may be a hardware device or a software client. This is the end device that the user uses.

MCUs are responsible for managing multi-point connections such as conference calls. It includes a multi-point controller that manages call signals and can include multi-point processors. They are in charge of mixing individual streams, switching, or other media transmission properties.

The Gateway (GW) is a Media Gateway Controller (MGC) and Media Gateway (MG) device. These two devices can work both individually and simultaneously. MGC serving call signals and other features that are not related to the transmission of the content itself. MG have in charge of the operator and managing the content being transmitted. GW serving as an interface between H.323 and other networks, such as the public telephone network, H.320 and other H.323 proxy networks.

Gatekeeper is an optional component in H.323 networks and serves for address translation and access control. The gatekeeper allows calls to be connected directly between endpoints. He can direct the call through himself, and allows calls to be diverted when a subscriber is busy.

Peer elements are usually located along with Gatekeeper. They have the role of exchanging address information and participate in the authorization of the call inside and between the administrative domains. Peer elements can aggregate address information to reduce the amount of traffic information that travels through the network. Border elements are a special kind of peer elements. They act between two administrative domains. They help in authenticating and authorizing the call directly between two administrative domains.

H.323 is a specification of how each element should work together and work. H.323 protocol communication involves the use of several other protocols:

- H.225.0 defines call signaling between endpoints and Gatekeeper,
- RTP and RTCP,
- H.225.0 Annex G and H.501 specify procedures and protocols for communication inside and between peer elements,
- H.245 is the protocol used to control the establishment and termination of media channels within the context of the call and performs a conference check,
- H.450.x is a series of support service protocols,
- H.460.x describes the H.323 base protocol extensions; are independent of its version,
- T.120 specifies how to exchange data,
- T.38 defines the transmission of the fax service signal,
- V.150.1 defines the modem signal transmission,
- H.235 describes security in systems using H.323

5.3.3 Factors/Parameters influencing VoIP

Parameters like error rate, delay, and signal-to-noise ratio provide us with network performance information. It is difficult to judge on the basis of these parameters how the network will be perceived by users. Two parameters are important for the voice: loss and unidirectional delay. Jitter is a less important parameter because it can be filtered on the receiver at the cost of an increased one-way delay.

Factors affecting the VoIP service are as follows:

5.3.3.1 Loss of packets causes distortion in voice applications, which is growing with rising losses. When the overall voice service quality is represented by the Mean Opinion Score (MOS) then with increasing loss, MOS decreases. In general, packet loss has a minimal impact on voice quality if it is less than 1%. Higher loss rate causes a deterioration in perceived quality. Another factor a codec is used. PCM codecs show a higher tolerance to packet loss than more complex algorithms. Loss of packets is indicated by the percentage or absolute number of packets that are not delivered to the end device.

5.3.3.2 Variable delay is a parameter that is important and greatly affecting the VoIP service, or jitter. This parameter specifies the deviations that have been detected between the passages of the individual packets. This delay can be partially eliminated if a buffer is used, which is used to buffer memory in the end device. The maximum jitter value should be max. 30 ms.

5.3.3.3 Delay is the time that will elapse from sending the packet to its reception. With increase one-way delay in telephony applications, switching between the roles of the speaker and the listener becomes more complex. This undesirable effect reduces the quality of all types of interactive communication, and must be controlled. Delay is one of the biggest calls for IP telephony. Delays in IP telephony are typically greater than traditional telephony calls. The reason for this is the delay on the new network elements. The Delay recommendation defines ITU-T G.114.

The sources of delays in VoIP include:

- Coding/Decoding Delay - Sophisticated codecs achieve high compression of voice data, which increases delays. Two different components of the coding delay account for this delay. The first is the delay caused by processing. This is the time, which is needed to process the voice frame. The second component is a delay created by waiting the decoder for the next data, to use it their correlation with the frame.It is currently being processed to achieve better compression. The coding delay you need to add a decoding delay, which usually takes half of the time that the coding delay. The total encoding/decoding delay may take a few milliseconds. For codec G.729, the coding delay takes 15 ms and the decoding is 7.5 ms.

- Delay by packet compilation - Lots of voice encoders generate very short frames. The G.711 encoder creates 8 bit frames every 125 μs, and G.729 creates 80 bit frames every 10 μs. In order to increase transmission efficiency, transmitters can populate the IP packet with different voice frames. The result is an increased delay before the transmission itself. For example, packet delay for 5 G.729 frames is 50 ms.

- Serializing Delay - It is time to place the packet on the transmission line. Here depends on the size of the packet itself and transmission speeds. For a 100 byte frame and a 64 kbit/s line, the serialization delay is 12.5 ms.

- Delay by waiting in a packet queue - This delay was created in case of network overload. It is variable and difficult to predict, but can be minimized by implementing QoS mechanisms in the network.

5.3.4 VoIP terminal devices

Terminal equipment can be divided into software VoIP phones, IP phones and mobile VoIP phones.

The best-known VoIP software tool is Skype. This service offers voice over VoIP and is a very widespread service across the globe. It offers free-to-call telephony and telephony to other fixed and mobile carrier networks across the globe at significantly lower prices than traditional phone operators. The new VoIP service becomes Google Talk, a major player in the Internet technology field, and can get a lot of affiliate services like e-mail.

In general, software resources for VoIP are a complete replacement of hardware solutions. In addition to voice transmission they offer services such as sending and receiving SMS, fax, or voice-mail. Their graphical interface often resembles a real phone. For use, the end device must be equipped with speakers and microphone. Very popular have become apps installed directly on mobile phones and smart-phones that do not support VoIP.

Hardware VoIP phones are devices that can communicate with IP. They are at first glance indistinguishable from classic phones that do not support IP protocol. In fact, they are small computers that are equipped with FLASH memory instead of hard drive, sound card, simple processor, display, buttons, or touch-pad, and RAM for more powerful models. They also contain network interfaces, typically RJ-45, to connect to the network or provide connectivity for other devices. The cheaper models of VoIP phones typically contain one such interface, with more expensive models being two. Between the most famous VoIP phones manufacturers include Cisco, Panasonic, Siemens, HP, or D-Link. The cheapest models of VoIP phones are moving at price levels within a few hundred crowns, with the most expensive models going up to tens of thousands of crowns.

Mobile VoIP devices are also coming to the forefront thanks to third generation 3G mobile networks and now also LTE. They offer relatively high transmission speeds, for voice transmission within VoIP. There is a An interesting alternative to classic mobile networks, when the user is connected to the Internet at significantly lower costs than in conventional calls.

5.3.5 Subjective methods of measuring quality

The quality of the degraded voice samples is evaluated statistically by a larger group of people. This group of people responds to a questionnaire from ITU-T Recommendation P.82 and expresses an opinion on the given voice sample. This measurement is time consuming and costly. The resulting value is the true value of the MOS-LQS (Mean Opinion Score) (Listening Quality Subjective). When recording the sample, the requirements of ITU-T Rec. P.800. These requirements include: a specific recording room, high quality recording equipment, microphone distance, recording noise level, simple sentences and fluency, and others. Even when evaluating a sample, requirements such as: a specific listening room (it should have the same features as a recording room), high quality speakers, listeners did not take this test for more than 6 months, and others [84].

5.3.6 Objective methods of measuring quality

Evaluating MOS is complex, time-consuming and costly. In practice, direct evaluation of MOS replaces estimates made by machines or algorithms. Algorithmic evaluation methods should be objective because they do not rely on opinions people, but many of them are trying to estimate the MOS parameter, which is derived from the average of individual tests (opinions). Depending on the input data, objective algorithms, evaluating quality, are divided into the following groups [82]:

- View models - these models have various factors at the input, including delays and other network performance parameters. The result is an estimate of the conversational MOS. The E-model is the most significant and most widely used model of opinion.
- Models on speech layer - these input models require speech signals. The result is a generated estimate of MOS listening.
- Models on the packet layer - in this case, IP packets are a output is an estimate of MOS listening. These models are suitable for monitoring speech quality at intermediate network points where speech signals are not available.

5.3.6.1 E-model is the computational model, which uses transmission parameters to predict subjective voice quality. It was standardized in Recommendation G.107 and was also adopted by the European Telecommunications Standards Institute (ETSI) and the US Telecommunications Industry Association (TIA) as a network planning tool.

The result is in the R-factor, evaluating speech quality on a scale of 0 for bad and 100 for perfect. R-factor values can be theoretically converted to a MOS scale. To its limitations, the maximum value for R achievable by conventional PCM networks is 94. For VoIP systems, R-factor values are usually worse.

The assumption of the E-model is that the effects of individual degradation can be summarized in the equation [82] [85]:

$$R = R_o - I_s - I_d - I_e + A \tag{17}$$

- $\mathbf{R_o}$ — Signal to noise ratio,
- $\mathbf{I_s}$ — deterioration created linear distortion,
- $\mathbf{I_d}$ — deterioration created linear distortion,
- $\mathbf{I_e}$ — deterioration created by distorted voice
- **A** — factor of expectation, takes into account unsatisfactory performance with the new service.

Table 13: E-model

R-factor	**MOS**	**Label**
100-90	4.5-4.34	Excellent
90-80	4.34-4.03	Great
80-70	4.03-3.60	Good
70-60	3.60-3.10	Fair
60-50	3.10-2.58	Poor
49-0	2.6-1.0	Bad

5.3.6.2 PSQM This model is the basis for the ITU-T Recommendation P.861. The idea is to measure deterioration due to distortion in the psycho-acoustic domain, unlike normal frequency domains. Personal speech quality measure (PSQM) works by comparing the deteriorated and original signal. This means that PSQM testing is always performed with a reference. The result of the algorithm is an estimate of MOS, which is sometimes called an objective MOS (OSMO), because it is calculated using the algorithm instead of averaging subjective evaluations [82] [84].

5.3.6.3 PESQ Designed to assess the quality of voice codecs. Not suitable for voice networks, especially for IP telephony, because it is unable to address packet loss and variable latency. These problems are solved by the Perceptual Analysis Measurement System (PAMS) algorithm, which is the predecessor of the PESQ (Personal evaluation of speech quality) algorithm. PESQ has been standardized in ITU-T Recommendation P.862. This is the latest and most accurate objective quality model on the speech layer. PESQ combines a time alignment algorithm with the PSQM psycho-acoustic model [84].

5.4 Data services

This service, part of the Triple Play service, aims to share, receive and send data over the Internet. For the purpose of this communication, use FTP to manipulate files that are located in different parts of the network.

5.4.1 FTP protocol

FTP (File Transfer Protocol) is used to transfer files between end-users in the data network and is not dependent on the operating system that is at the end user. The protocol works on the application layer OSI/ISO model, and works on a client-server basis when using TCP for its transmission. It uses two TCP / 20 and TCP / 21 ports for communication. The FTP server on port 21 listens to an incoming connection from an FTP client. Only data is transferred to port 20, and commands are also on port 21.

You can connect to an FTP server in two modes, either passive or active, when passive mode is safer.

- Active Mode - In this mode, the server connects to a data connection and the client listens. The problem occurs when a client connects from a private network and his IP address has to be translated.
- Passive mode - In this mode, the client connects to the connection, which, when compiled, sent the server the IP address and the TCP port on which it listens..

5.4.2 HTTP protocol

The Hypertext Transfer Protocol is an application-level protocol for distributed, collaborative, hypermedia information systems. It is a generic, stateless, protocol. This specification defines the protocol referred to as "HTTP/1.1", and is an update to RFC 2068. Web-based communication takes place via HTTP or HTTPS (Secure) protocol [11].

6 Practical measurement

In this part of the diploma thesis I will familiarize myself with the measurement itself, will introduce used equipment and describe the important implementation of devices.

6.1 Description of experimental workplace

The chapter contains a description of the individual devices and elements I used to implement the wiring. Set up measuring devices and configure the individual devices for Triple Play services.

The workplace is equipped with GEPON technology testing. There are two Races in the lab that contain both active and passive devices. The active devices include the Abacus Server, the L3 switch, the ONU drive. The racks contain different passive elements. Racks are associated with individual benches using fiber optics.

All laboratories and the creation of diploma thesis was realized in the lab VŠB - Technical University of Ostrava at the Department of Telecommunications Technology.

They are linked as follows:

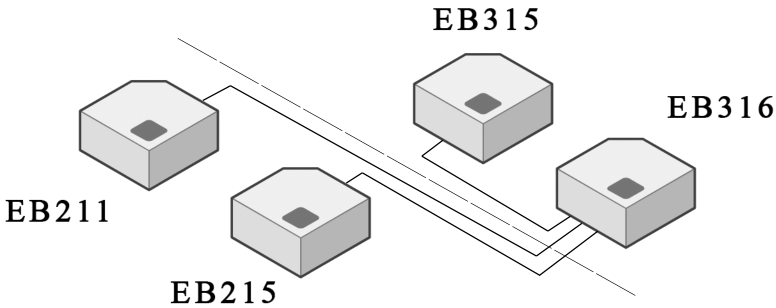

Figure 26: Interconnection scheme between laboratories

Figure 26 show laboratories and their interconnections. **EB316** - OLT unit Allied Telesis, Triple play servers, **EB315**- connection with Data Distribution. Optical path (5, 10, 15, 20, 25, 30, 35 km), **EB215**- VDSL2 DSLAM ZyXEL, **EB211**- Spirent line management simulator.

6.2 Network topology

As shown in Figure 26, we used the laboratory topology throughout the new FEI building. I used OLT to do this after modem. These devices are only the most important to measure I have been using more. All other devices described below are listed in the chapter below.

6.3 Devices used

Several devices and software were used to implement the path and measurement. This chapter gives a brief description.

6.3.1 OLT Allied Telesis iMAP 9102

OLT is the foundation of every topology. It consists of four slots, three of which are intended for the installation of participant cards and the fourth one for the unit management. The management card is referred to as CFC12 and has four Ethernet ports. Configuration OLT is done using the CONSOLE port. Also important is the EPON2 (TN-118-B) card that connects the PON network through the SFP connector.

OLT by Allied Telesis model iMAP 9102. This model is displayed on Fig. 27.

Figure 27: Allied Telesis iMAP 9102 [57]

On the front are slots for modular cards. There are CFC12, FX 20 BX and EPON2 cards. **Card CFC12** it is made up of ports like **Console**, which is used to connect to a PC in front of a UTP cable. There is a port to connect to **MGMT** and **Fast Ethernet**. The last port is for fiber optic connection for the **WAN** interface. Connection is made via SFP modules. **Card FX 20 BX** is made up of 20 ports for optical connectors. **Card EPON2** it contains two SFP modules that are used to connect to an optical passive network.

6.3.1.1 Configuration

Configuration of the OLT unit was performed by linking serial RS-232 cable with CONSOLE port on OLT MiniMAP 9102. Using the PuTTY client is possible log into OLT unit configuration mode. The client had to set up several parameters and select a Serial connection.

```
# Login credential
  Username: "officer"
  Password: "officer"
```

Listing 1: Login credential Allied Telesis iMAP 9102

In the topologies, devices requiring IP addresses were used for each other communication. To set the EPON IP address and ONU registration, use the command:

```
# set the EPON IP address
  SET INTERFACE=2.0 EPON IPADDRESS=192.168.0.12
# registration ONU
  CREATE ONU=ONU1 ONUID=1 INTERFACE=2.0 MAC=00:15:77:73:98:68
```

```
# useful commands
  SHOW INTERFACE
  SHOW ONU
  SHOW system
  SHOW card
```

Listing 2: EPON Interface Configuration

Create and assign a VLAN with a defined name.

```
# Creta Vlan with name vlan300 and assignment ONU
  CREATE VLAN=vlan300
  ADD VLAN=vlan300 INTERFACE=2.0.1
# Create QoS profile
  CREATE QOSPOLICY=100Mbit MAXUPSTREAMRATE 100M MAXDOWNSTREAMRATE 100M
  ADD QOSPOLICY=100Mbit INTERFACE=2.0.1 BIDIRECTIONAL VLAN=vlan300
# group broadcasting
  ADD QOSPOLICY=100Mbit INTERFACE=2.0.1 BRUUM
  ENABLE IGMPSNOOPING VLAN=VLAN300
# useful commands
  SHOW QOSPOLICY
  SHOW VLAN
```

Listing 3: EPON VLAN and QoS Configuration

6.3.2 ONU Allied Telesis AT-ON1000

The endpoint is designed to support data services up to 1 Gbit/s via a passive optical network. It is equipped with an optical and metallic connector and converts the optical signal to an electrical signal and conversely.

The device is for GEPON networks. The WAN connectivity is secured using the PX20 optical SC connector. On the client side, the device is equipped with an RJ-45 connector (10/100/1000 base-T). There is VLAN support, IGMP DHCP server and more.

Figure 28: Allied Telesis AT-ON1000 [58]

6.3.3 Fiber optic splitter

Fiber optic splitter is used to divide the optical signal into multiple directions. In the work is used divider in ratio 1:2.

As next I use two Wavelength Splitters (WDM splitters) to split the optical signal according to wavelength. I split wavelength on S, C and L (1460 - 1620 nm) band and O (1260 -1360 nm) band. On fig. 35 shows the wavelength division of used WDM splitter.

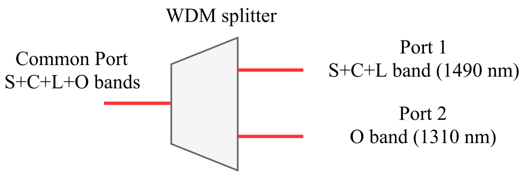

Figure 29: WDM splitter

6.3.4 ZyXEL IES-5005

It contains five slots, the first of which is for managment and the remaining four for cards terminating lines. The lab is equipped with MSC1000G and VLC1224G-41 cards. The first one includes a 10/100M MGMT interface, two 1000/100 Ethernet interfaces, four SFP interfaces, and one console interface. The VLC1224G-41 is a VDSL2 line card that contains 24 ports with a maximum bit rate of 65 Mbit/s for downstream and 35 Mbit/s for upstream.

This card uses the DSLAM configuration or links to other networks. The second card is already a VDSL2 splitter, whose 12 ports are fed into the patch panel above the unit. The only need was to allow IGMP Snooping permission to do so handling multicast traffic.

6.3.5 Server Abacus

The device manufacturer is Abacus, a server that has four network cards with a maximum bit rate of 100 Mbit/s (Full Duplex).

Server uses a VSphere operating system based on Linux to serve as a server virtualization tool. Hard disks include system, virtual machines, and data.

Specifications:

- Processor: Intel Xeon E5-2620, 2.00 GHz
- Network adapter: 2x Intel Corporation 82575EB Gigabit Network Connection. 2x Intel Corporation I350 Gigabit Network Connection
- 2x HDD (465,5 GB; 460,75 GB)

6.3.5.1 Configuration

On the server, I updated virtual machines with Triple Play Services (Data, VoIP, IPTV). Virtual machines run on Ubuntu Linux. Machines have been configured to local IP.

Access to the server is ensured by using software VMware VSphere Client. This PC mus be on the same network.

```
IP address of server: 158.196.81.21
Name: "root"
Password: "n311kat44c"
# set on All VM
  ifconfig eth0 10.0.0.0/8
  ip route add to 224.0.0/4 dev eth0
```

Listing 4: Login data to the Abacus server

On the server are three virtual machines: IPTV server, VoIP server and Data server. All servers can be logged on using the same information.

Configuration IPTV

For streaming videos were used by the VLC program. VLC player is suitable for IPTV service. Installing VLC on OS Linux Ubuntu is executed using the terminal by:

```
sudo apt-get update
sudo apt-get install vlc
```

Listing 5: Install VLC on Ubuntu

```
#Server side
  Media/Open Network Stream
    #In File section add video example stream
    #Then click on Stream and choose RTP/MPEG Transport Stream and we will
        add
      Address: "224.1.1.90"
      Base Port: "5004"
      Activate Transcoding: "False"
#Client side
  Media/Open Network Stream
    #In Network section
      Please enter a network URL: "rtp://@224.1.1.90.5004"
```

Listing 6: Configuration IPTV

Configuration VoIP

The Asterisk PBX is used for VoIP. Asterisk is installed on the server side. Asterisk is a open source and free framework for building communications applications.

On the client side, the Grandstream GXV3140 is used. These phones are registered as SIP accounts in the asterisk control panel

```
apt-get update
apt-get install asterisk
#Restart asterisk
  /etc/init.d/asterisk restart
#Start Asteris
  asterisk -cvvv
```

Listing 7: Install Asterisk

Parameter **-c** defines the opening of the console. Parameter **-v** defines the listing detail level. After registering your phone (or VoIP app) to the same network as the asterisk switchboard. The phones are automatically registered. This registration may take several minutes. After signing up will appear in the CLI asterisk registration statement.

The **sip.conf** file is used to configure all connections to the device using the protocol SIP.

```
[general]
port = 5060
allow = all
binaddr = 0.0.0.0
[1000]
type = friend
host = dynamic
secret = 1000
[1001]
type = friend
host = dynamic
secret = 1001
```

Listing 8: Configuration Asterisk /etc/asterisk/sip.conf

The **extensions.conf** file defines the PBX numbering plan. Numbered plans are part of contexts.

```
[default]
exten=> 1000, 1, Dial(SIP/1000)
exten=> 1001, 1, Dial(SIP/1001)
```

Listing 9: Configuration Asterisk /etc/asterisk/extensions.conf

Configuration Data

Web server was in Abacus server on a virtual machine. The user can do that server to connect remotely by entering the IP address in the Internet browser. On the web server is located for download and upload section.

```
apt-get update
apt-get install apache2
```

Listing 10: Install Apache

Configure file for apache and web server is located in **/var/www**.

```
<HEAD><TITLE>DATA SERVER</TITLE></HEAD>
<BODY>
<h1>Download</h1>
<a href="file4.mkv">File 18100Mb</a>
<h1>Upload</h1>
<b>File to upload:</b>
<FORM ACTION="index.php" METHOD="post" ENCTYPE="multipart/form-data">
<input type="HIDDEN" name="MAX_FILE_SIZE" VALUE=100000000>
<INPUT TYPE="file" NAME="file" SIZE="40">
<INPUT TYPE="submit" NAME="ok" VALUE="Upload">
</FORM>
<P>Files on server:</P>
<a href="file1.avi">file1.avi</a><br>
</BODY>
```

Listing 11: HTML Data web server

6.3.6 Spirent DLS-6900

DLS-6900 is a lead simulator. Can simulate cable models PE04 and 26AWG. For type PE04, it can simulate at a distance of 7 km in step 100 meters. It offers bandwidth up to 30 MHz. It is useful for testing VDSL, VDSL2, ADSL, ADSL2 + and SHDSL. This line simulator also includes an internal noise generator that allows the simulated line to provide white noise at levels -90 dBm/Hz to -140 dBm/Hz. The control is very intuitive, all the parameters set are shown on the front panel.

6.3.7 ZyXEL P-870MH

VDSL2 modem that supports QoS and is suitable for Triple Play service providers. The device is equipped with an RJ-11 port for VDSL and four LAN 10/100 ports for RJ-45 connectors. Supports transfer rates up to 65 Mbit/s downstream and 35 Mbit/s upstream. IGMPv2 was required to be enabled here.

6.3.8 L3 switch ZyXEL XGS-4528F

This L3 switch with dual ports supports multi-layer functionality (L3/L3/L4) at 10/100/1000 Mbit/s and 10 Gbit/s connection rates. The ZyXEL XGS-4528F offers 20 ports 1000BaseT and 4 Gigabit dual ports.

6.3.8.1 Configuration

On the L3 switch, the DHCP server was set. I got to the configuration web graphical interface via IP address 192.168.1.1. The graphical interface offers a wide range of settings, it was necessary to activate the IGMP protocol, which is necessary for the multicast function.

```
IP L3 switch: 192.168.1.1
name: "admin"
password: "1234"
```

Listing 12: Login for L3 switch ZyXEL XGS-4528F

The graphical interface offers a wide range of settings, it was necessary to activate the IGMP protocol, which is necessary for the multicast function. I set up Full Duplex in the ADVANCED APPLICATION folder.

6.3.9 EXFO FTB-500

A universal platform that includes 8 slots for the measuring modules themselves. It uses the Windows XP operating system.

The following modules were used for measurement:

6.3.9.1 FTB-5240B is an Optical Spectrum Analyzer (OSA).

6.3.9.2 FTB-5500B is the Polarization Mode Dispersion (PMD) module that allows polarization of the dispersion to be measured. The wavelength range is 1260 to 1675 nm and the sensitivity is -47 dBm.

6.3.9.3 FTB-5800B is the Chromatic Dispersion (CD) module, which is used to measure the chromatic dispersion. The range of wavelengths is 1530 to 1625 nm / 1200 to 1700 nm.

6.3.9.4 FTB-7200D is an Optical Time Domain Reflectometer (OTDR) module. It is optimized for single and multi-threaded fiber testing. Single wave strands use wavelengths of 1310 nm and 1550 nm, for multi-threaded fibers, wavelengths of 850 nm and 1300 nm are used. The dynamic range is up to 36 dB with a dead zone less than 1 meter.

6.3.10 EXFO FTB-1/FTB-880 NetBlazer

FTB-1 is a platform for FTTx and Ethernet testing, are the optimal choice for field technicians. FTB-880 module allows quick and easy verification of Ethernet services. Contains USB ports 30.

Figure 30: EXFO FTB-880 [59]

6.3.10.1 Configuration

The FTB-1 platform works under Windows. After signing in to the system the FTB-880 NetBlazer module will start. To connect to the XGS-4528F switch and act as a remote unit to perform a Dual Test RFC. It was also connected to a DSL modem for the EtherSAM test.

Use the UTP cable to connect NetBlazer. I statically set the IP address so that it is in the same range as Loopback.

```
#Set IP address with Loopback unit
  Discover/Remote: "Scan Subnet, must be set Loopback"
#RFC2544 -> Interface -> Port
  Port: "Port1"
  Interface Type: "10/100/1000 Mbit/s Electrical"
  Auto Negotiation: "Manual"
  Speed: "100 Mbit/s"
  Duplex: "Full"
  Flow Control: "None"
#RFC2544 -> Interface -> Network
  Ip address: "Define IP subnet"
```

Listing 13: Configuration EXFO FTB-1/FTB-880

6.3.11 EXFO AXS-200/635

Provides Triple Play service solution for VDSL2 and Ethernet 10/100M. IPTV testing measures packet loss parameters, PCR jitter, packet jitter, MDI, PID viewer. Monitor VoIP call statistics. Verifies they are fulfilled QoS requirements.

Figure 31: AXS-200/635 [78]

6.3.11.1 Configuration

Cable is connected to WAN port. For the IPTV test, select the IPTV Analysis sub menu and go to the IGMP tab. Monitor where we join with the desired multicast address. For data and VoIP services, there was a problem with capturing the required data stream services, other techniques have been used to monitor these services.

```
# DSL/IP Tests -> Connection Setup -> Selected Profile
  Line Mode: "Ethernet"
  Access Mode: "Routed Ethernet DHCP"
# In IPTV Analysis can be set length of monitoring
  IGMP Monitor: "Join multicast address"
```

Listing 14: Configuration EXFO AXS-200/635

6.3.12 EXFO PPM-350B-EG/PPM-350C

Are PON power meters device for measurement optical power in EPON networks and establishes a new FTTx testing benchmark. Filtered measurements, providing distinct power values for each signal (1310 nm, 1490 nm and 1550 nm). The device has two ports one for OLT and the other for ONT. Features include: Easy-to-Access Data Storage, Simultaneous Display of All PON Signals, Quick and Efficient Visual Inspection, Automated Pass/Warning/Fail Assessment and Rugged and Weatherproof Design [91] [92].

Figure 32: PON Power Meter [92]

6.3.13 Grandstream GXV3140

A hardware IP phone enabling VoIP communications. It includes NETWORK (10M/100M) ports and PC, equipped with 4.3 digital TFT display, CMOS camera.

Supports network protocols:

- SIP RFC3261, TCP/UDP/IP, PPPoE, RTP/RTCP, SRTP, HTTP/HTTPS, ARP, ICMP, DNS, DHCP (client), NTP/SNTP, TFTP, Telnet, UPnP.
- Voice code: G.711, G.722, G.729, G.723.1, GSM-FR, G.726-32, L16-256.

It also includes a variety of applications like Skype, Web Browser, RSS feeds, calendar, and more.

6.3.13.1 Configuration

Grandstream GXV3140 phones can be configured directly, using phone keys or network connectivity, and obtain an IP address from a DHCP server via a web interface.

```
# Configuration with web interface
  IP address: "from DHCP"
  Username: "admin"
  Password: "admin"
# General setting set Account
# Codec Settings settings with video codec
# Call settings DialPlan: "{x+|*x+}"
```

Listing 15: Configuration GXV3140

It is very important, every time you change it is necessary to restart the device and it may take several minutes.

6.4 Software used

This chapter describes the measurement software used.

6.4.1 VLC Player

VLC is a open source and free cross-platform multimedia player and framework that plays most multimedia files as well as DVDs, Audio CDs and streaming protocols various.

VLC Plays everything - Files, Discs, Webcams, Devices and Streams. Plays most codecs with no codec packs needed - MPEG-2, MPEG-4, H.264, MKV, WebM, WMV, MP3. Runs on all platforms - Windows, Linux, Mac OS X, Unix, iOS, Android.

6.4.2 IxChariot

A tool for assessing the performance of complex networks. It allows you to create up to 100,000 connections between endpoint pairs representing hundreds of thousands of end-users. It supports TCP, UDP, RTP, IPv4 and IPv6 protocols and can simulate applications from downloading torrents to Google Play.

Work in IxChariot is done using a user console that displays simulation and results. It consists of two APIs. This tool is created in the C programming language.

6.4.3 MSU Video Quality Measurement Tool

Program for objective video quality assessment. It offers the ability to analyze a single video and also explore two videos. Provides metric values for each slide, average values for the sequence of frames, and metric values for a particular color component.

There are 20 objective metrics (such as PSNR, SSIM, MSU, etc.), supports over 20 video formats. There is a freely downloadable version that does not allow the HD video to be evaluated or a PRO version paid. The work is for HD video used PRO Demo version, which, however, calculates the metrics in the random part of the image, the values are therefore at least indicative.

7 Network Integrity

Network integrity testing was performed using the RFC 2544, ITU-T Y.1564 EtherSAM and RFC 6349 tests. Before integrity testing, it was necessary to decide how long to choose. On VDSL2 I chose the profile on port 12, which had a downstream speed of **63968 kbit/s** and upstream of **35968 kbit/s**.

Topology for test measurement is displayed on figure 33 and 34.

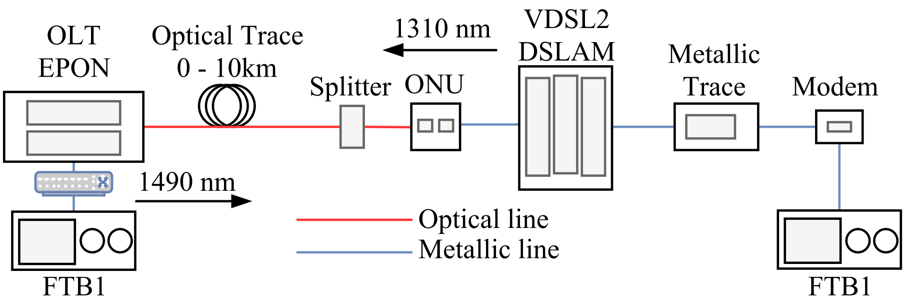

Figure 33: Topology network test integrity according to RFC 2544 without SOA

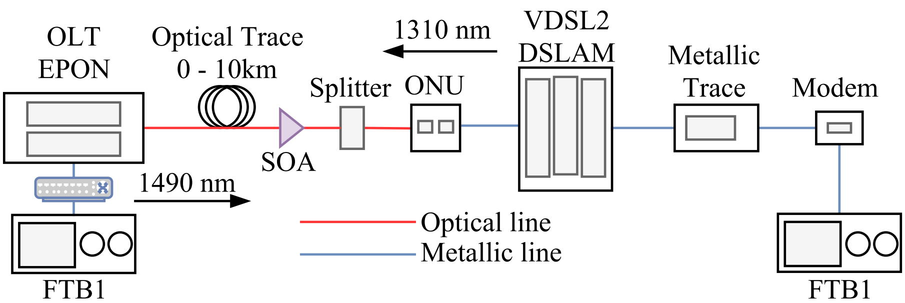

Figure 34: Topology network test integrity according to RFC 2544 with SOA

The test results are described later in chapter. The basic combinations of path lengths are shown in the table 14. There are combinations of the optical and metallic parts shown here. The real results on the metallic part are shown in the table 29.

Table 14: Selected combinations of path lengths

<table>
<tr><th>Max Down/UP
64/36 Mbit/s</th><th colspan="13">VDSL2 profile 12</th></tr>
<tr><td>Optical path [km]</td><td colspan="3">0</td><td colspan="3">10</td><td colspan="2">10 (SOA)</td><td colspan="3">20</td><td colspan="2">20 (SOA)</td></tr>
<tr><td>Noise [dB]</td><td colspan="3">-140</td><td colspan="3">-140</td><td colspan="2">-140</td><td colspan="3">-140</td><td colspan="2">-140</td></tr>
<tr><td>Metallic path [km]</td><td>0</td><td>0.6</td><td>1</td><td>0</td><td>0.6</td><td>1</td><td>0</td><td>1</td><td>0</td><td>0.6</td><td>1</td><td>0</td><td>1</td></tr>
</table>

The display of modules in FTB1 can be seen in Figure 35. This is the initial interface after logging on to the module on the tab test application.

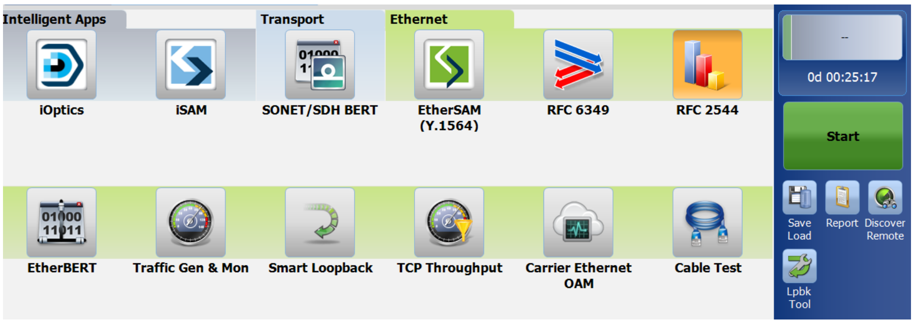

Figure 35: Menu FTB1/880v2 in the test application

7.1 RFC 2544

RFC 2544 has been developed for testing in laboratory conditions. This standard describes tests for network devices. It is used to measure the performance characteristics of the device. During testing, real-time traffic must be stopped, then the device generates specific frames. The measurement method includes - Throughput, Back-to-Back, Frame Loss and Latency. This method does not support testing multiple services at once.

7.1.0.1 Throughput Test

Where target is to specify the maximum number of frames per second that the device can process and send without breaking or loosing the frame. First, a number of frames are sent at a certain speed. If the number of received and sent frames is the same, the throughput and retesting will be increased. This is done until the number of received frames is less than the number of sent frames. At this point, the test is terminated and the resulting throughput is the last value at which the frame was lost.

7.1.0.2 Back-to-Back Test

Examines the capacity of the buffer by transmitting burst operation at the highest speed and then the longest burst is measured when no packets were dropped. The first step is to send bursts of frames with minimal spaces between frames. Then, the number of sent and received frames is compared. If the values are equal, a longer burst is sent if the received frames are less than sent, the burst duration is shortened.

7.1.0.3 Frame Loss Test

To determine the rate of loss over the entire frame rate and frame sizes. The Loss Ratio is calculated using the following formula:

$$FLR = \frac{FS - FR}{FS} \cdot 100\,[\%] \tag{18}$$

where: FLR- loss in percentages, FS- number of input frames, FR- number of frames at output

7.1.0.4 Latency Test

Used to determine the time from sending the frame to its acceptance. Typically, the frame is provided with a time stamp and placed in the middle of the burst. First, it is necessary to determine permeability for each defined frame size, then the frame stream is sent at a given rate. The time at which the frame is sent is stored, which is a time stamp A. The receiver must recognize the specific frame and record the reception time of the frame. This time is a time stamp B. Delay is the difference between mark B and mark A [56].

7.1.1 Testing RFC 2544 for EPON/VDSL2

The measurement was performed with the so-called Dual Test, testing in both directions. The length of the optical path varied during measurements. In the case of ADSL DSLAM, a length of 0 or 1 km of noise was added to the network. Supported Interfaces/Rates: 10M to 10G LAN/WAN. Test has to be executed in conjunction with a remote module. The remote module can be either in loopback configuration for unidirectional testing or in RFC 2544 Dual Test Set mode for bidirectional testing [98].

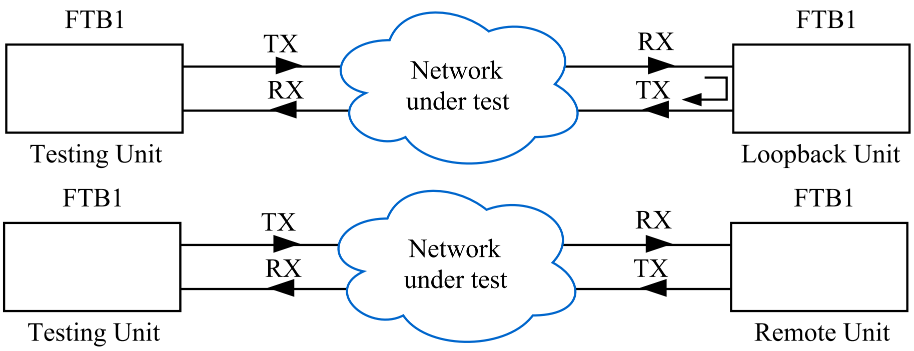

Figure 36: Typical RFC 2544 test applications [98]

Test setup for RFC 2544 I set up some parameters:

Subtests allows to individually enable the Throughput, Back-to-Back, Frame Loss, and Latency subtests. Estimated Time (H:MM) indicates the estimated time required to complete each subtest at best conditions.

Frame Size (Bytes): For RFC 2544 distribution, gives predefined frame size distribution values. For User Defined distribution, enter up to seven frame size values.

The settings and the in process test can be seen in the figure 37 and 38.

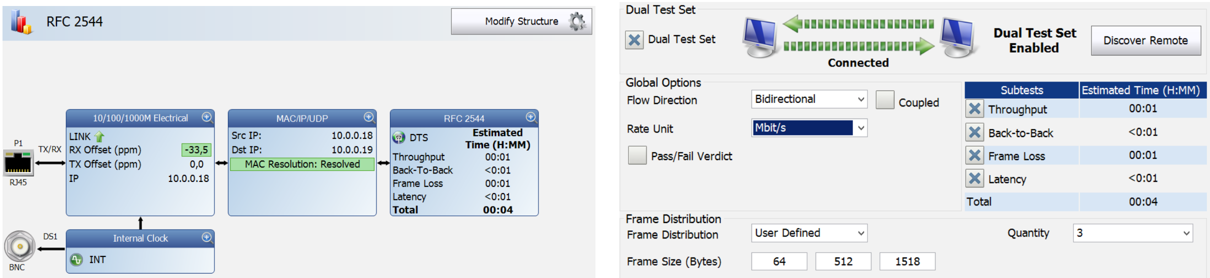

Figure 37: RFC 2544 Test configuration in menu

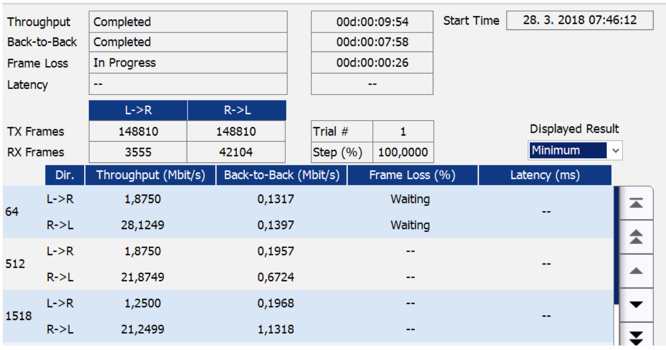

Figure 38: RFC 2544 In Progress test applications

7.1.1.1 RFC 2544 VDSL2 - Throughput

Figures 39 and 40 show the throughput for VDSL2 for downstream and upstream. The length of the optical path was less affected by the measurement of throughput, but the metallic track was more influential. There was a very poor connection on the 20 km optical path even at 0 km on the metallic path. The difference at the ideal optical path and the metallic path 0 km and 1 km and frame size 1518 was 38.12 Mbit/s. These are the average measurement values that were measured from the RFC 2544 Througput. The measurement was repeated and selected the most satisfactory.

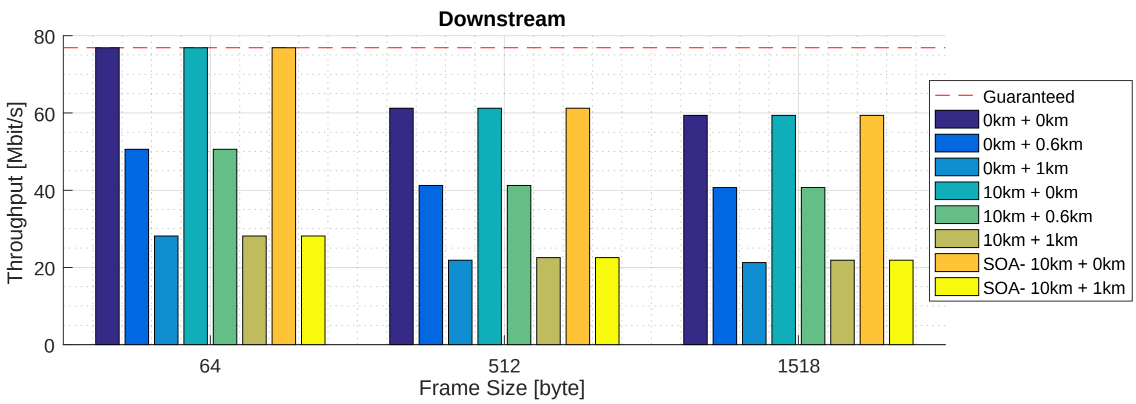

Figure 39: RFC 2544 VDSL2- Throughput- Downstream

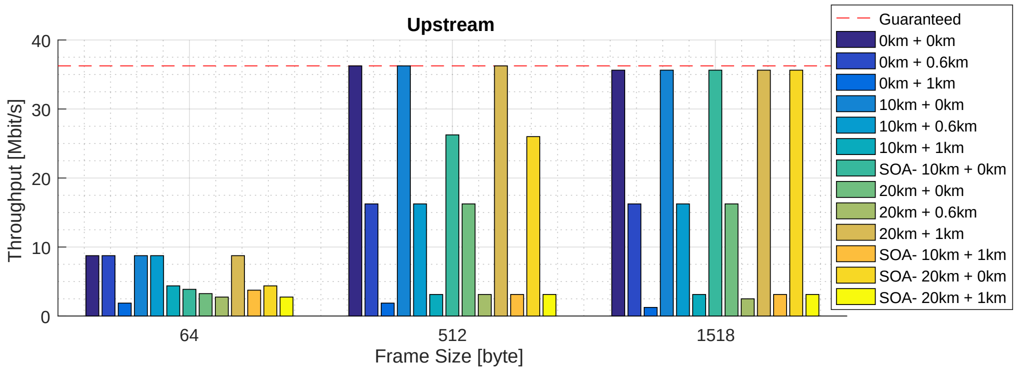

Figure 40: RFC 2544 VDSL2- Throughput- Upstream

7.1.1.2 RFC 2544 VDSL2 - Back-to-Back

From Figures 41 and 42 it can be seen that Back-to-Back is low. Values are low due to the use of active elements on the network. In the case of downstream Back-to-Back, it increases with a larger frame size, but in the case of upstream, the Back-to-Back value is the lowest of 64 bytes. Other frame sizes are values approximately the same. Measurements were repeated and the best results were then selected.

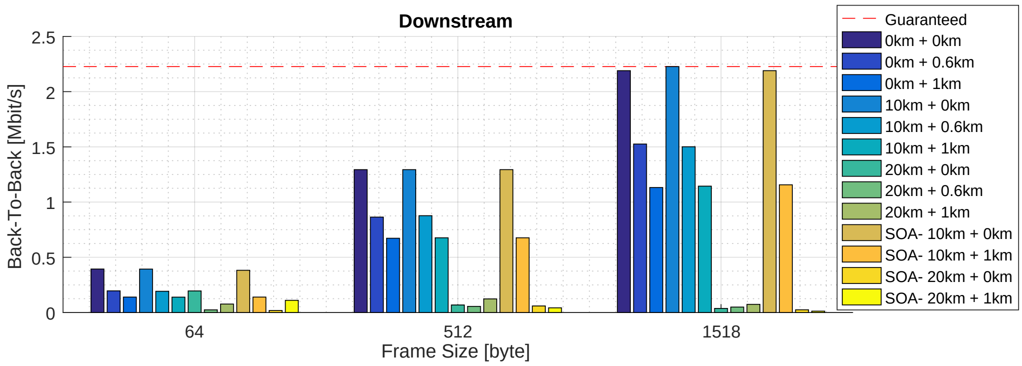

Figure 41: RFC 2544 VDSL2- Back-to-Back- Downstream

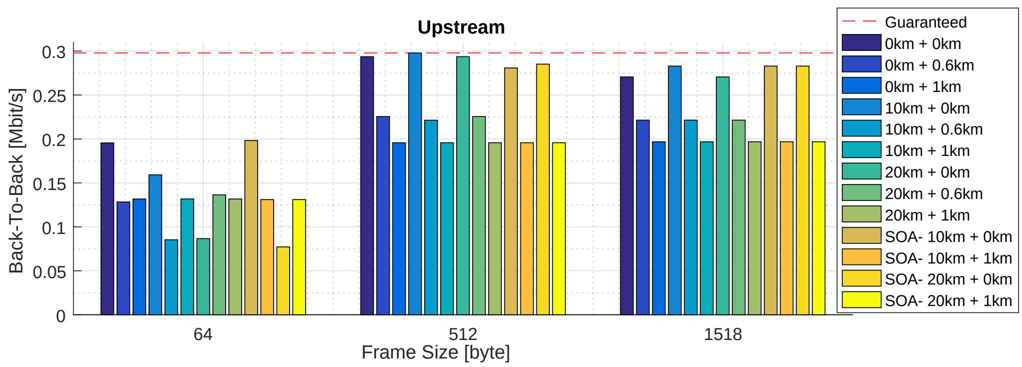

Figure 42: RFC 2544 VDSL2- Back-to-Back- Upstream

7.1.1.3 RFC 2544 VDSL2 - Frame Loss

Frame Loss is shown in Figures 43 and 44. The values are very high and are caused by a noise that is set to -140 on the simulator. Frame Loss increases with a larger frame size. For downstream, the frame loss is approximately 23% for 0 m of metallic wire at frame sizes of 64 bytes. The highest frame loss was measured in the case of 0.6 and 1 km of metallic conduction, reaching values of around 70% and 100% for the largest frames. In the case of upstream, fracture loss values were greater. For the largest frames, values were measured around 100% for 0.6 and 1 km of metallic conductors. On the length of the optical path was not affected by the loss measurement.

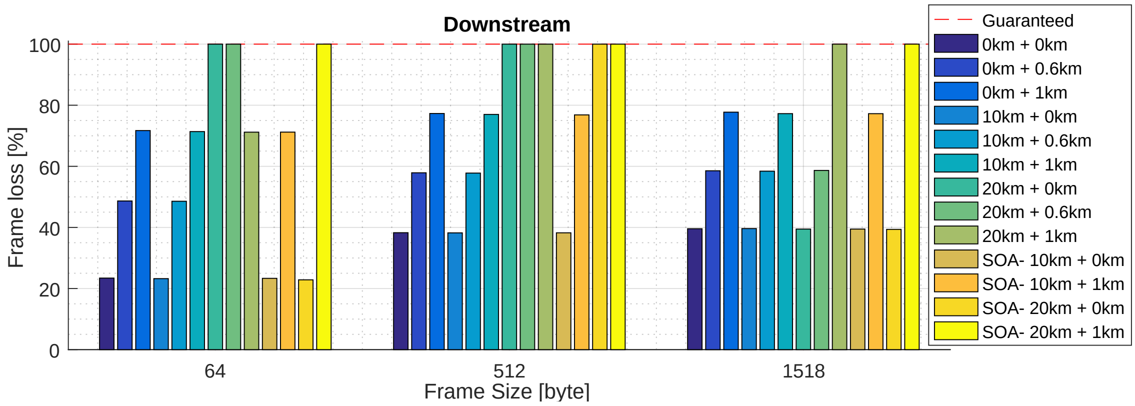

Figure 43: RFC 2544 VDSL2- Frame Loss- Downstream

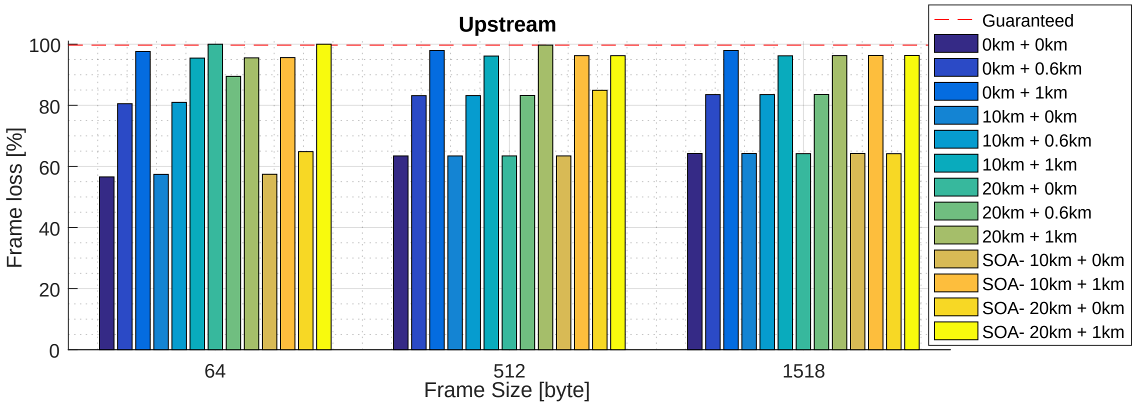

Figure 44: RFC 2544 VDSL2- Frame Loss- Upstream

High values ranged more than 20 km in a rink. Where a large loss has already occurred. the values are averaged from several measurements and then selected.

7.1.1.4 RFC 2544 VDSL2 - Latency

Figure 45 shows latency. The size of the metallic trace was affected by the measurements. The highest measured delay value was 1km for the metallic route and 0km for the optical path of 29ms. At a distance of 20 km latency was aborted.

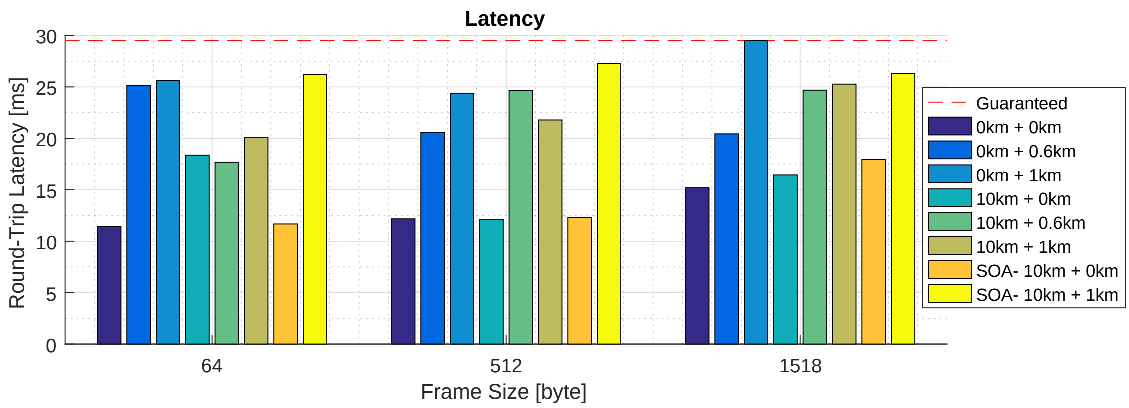

Figure 45: RFC 2544 VDSL2- Latency

7.2 EtherSAM (ITU-T Y.1564)

The Standard was adopted in 2011 and can simulate all types of services that will run on the network and simultaneously qualify all key SLA parameters for each of these services. It contains all the tools needed for quick and efficient performance verification. For all services, it tests the throughput, loss, delay, and jitter variability. Operation is divided into three classes, each class having a specific color. Moreover, it validates the QoS mechanisms provisioned in the network to prioritize the different service types [98].

Verification of transmission parameters takes place in two phases:

- **Check your network configuration settings**: a ramp test is performed for each service when 3 phases of the data stream are sequentially generated. The first phase is from the minimum bitrate to the CIR, the second phase is between CIR and EIR, and the third is in the band over EIR. The result of this test is the suitability of the network for the service and the suitability of the CIR and EIR configuration of the service. The test time is about 1 minute for each service.

- **Checking QoS Quality Parameters Settings**: in the test, all services are generated at the same time as CIR speeds to the network. It checks the service quality for each defined service and evaluates whether it meets the Service Level Agreements (SLA) parameters. The length of the test is user-selected.

This newer standard is used to simulate Triple Play services. Compared to RFC 2544, it is faster and more accurate, and it can also test Ethernet SLAs (Service Level Agreement), which is a binding contract between a service provider and a customer that guarantees minimal performance. EtherSAM Traffic Classes:

- **Commited Information Rate** (CIR, green traffic) - this is the bandwidth whose availability is guaranteed for any circumstances.
- **Excess Information Rate** (EIR, yellow traffic) - is the bandwidth above CIR that can be available to the customer. It depends on the use of the network. This value is not guaranteed.
- **Discarded Traffic, red traffic** - traffic over CIR and EIR. If this band is used, other services will be canceled, so this traffic is discarded.

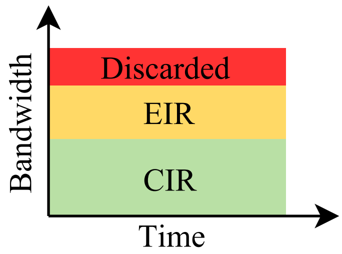

Figure 46: EtherSAM traffic classes

Specific traffic characteristics, indicating the minimum performance of a particular traffic profile, are called KPIs (Key Performance Indicators). In green traffic, the network must guarantee that these minimum requirements meet all traffic. Typical KPIs include [95]:

- **Bandwidth** - indicates the maximum amount of data that can be delivered. This is the ratio of the total amount of shipped traffic within one second. More services share bandwidth, services need to be limited to avoid interference.
- **Frame Delay** - is the measurement of the time span between sending the packet and receiving it. In most cases, it is a back and forth journey when simultaneous measurements are made in both directions.
- **Frame Loss** - it may appear for many different reasons, such as transmission errors or network congestion. An error may be caused by a physical phenomenon during transfer, which may result until the frame is dropped.
- **Packet Jitter** - in this case, it is the variability between packet delivery times. The packets are often queued in the network and sent to other devices in the form of bursts. If packets are handled with a different priority, packets are moved at different speeds. Therefore, packets are received at irregular intervals. This type of delay is particularly sensitive to audiovisual applications.

7.2.1 Testing EtherSAM (ITU-T Y.1564) for EPON/VDSL2

In this case, the measurements are performed on the same topology as the RFC 2544 measurements. The length combinations are also the same. The modem was connected with the FTB1 multifunction device and OLT with the second FTB1, which served as a feedback unit. The duration of the test is determined by the size of the network, the recommended length of the test is 15 minutes for services in smaller networks of one operator, 2 hours for larger networks, and 24 hours for services over several network operators. We should test download and upload.

The EtherSAM (ITU-T Y.1564) test has to be executed in conjunction with a remote module. The remote module can be either in loopback configuration for unidirectional testing or in EtherSAM Dual Test Set mode for bidirectional testing. Supported Interfaces/Rates: 10M to 10G LAN/WAN [98].

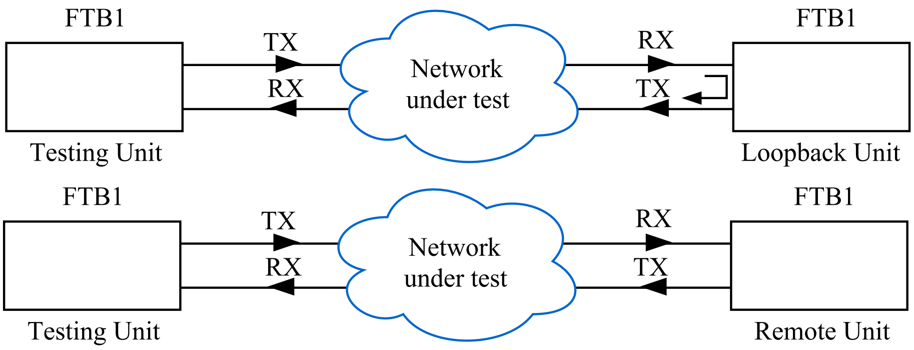

Figure 47: Typical EtherSAM (ITU-T Y.1564) test applications [98]

Test setup for EtherSAM (ITU-T Y.1564) I set up some parameters:

The Burst settings are only available for configuration when the Burst Test check box is selected. The burst configuration parameters are defined globally for all services but CBS, EBS, and Burst Max Rate parameters are as per each service configuration.

Dual Test Set (DTS) check box when selected enables EtherSAM Dual Test Set. Alternatively it is possible to use the Discover Remote button to connect to a remote module and automatically enable the Dual Test Set.

Service Configuration Test verifies if the network configuration is correct for each service before starting a long term test. To test the network configuration, a ramp test and/or a burst tests is/are generated for each configured service.

Service Performance Test verifies that the SLA parameters (The Service-Level Agreement (SLA) parameters allow enabling and defining the pass/fail verdict thresholds for the service) are met over time by running multiple services simultaneously. The maximum Jitter, Latency, Frame Loss, and average throughput are measured and compared to the configured thresholds to declare pass/fail verdicts.

The profile for the EtherSAM (ITU-T Y.1564) was set according to the table 15. The first service was set for IPTV with codec (HDTV MPEG4). The second service was set for VoIP with codec (G.723.1) and third Data Service.

Table 15: Set profile to EtherSAM ITU-T Y.1564 - VDSL2

Service	**Type x number**	**Service Profile**	**CIR [Mbit/s]**	**Frame Size [bytes]**
1	IPTV x 1	HDTV MPEG-4	10.5927	1374 (Fixed)
2	VoIP x 5	G.723.1	0.1316	82 (Fixed)
3	DATA x 1	Data	20	Random

The values in Table 15 were set according to recommendations to match real measurements.

The settings and in process test can be seen in the figure 48 and 49. Set individual services on FTB1. They are shown in the figures 50 and 51.

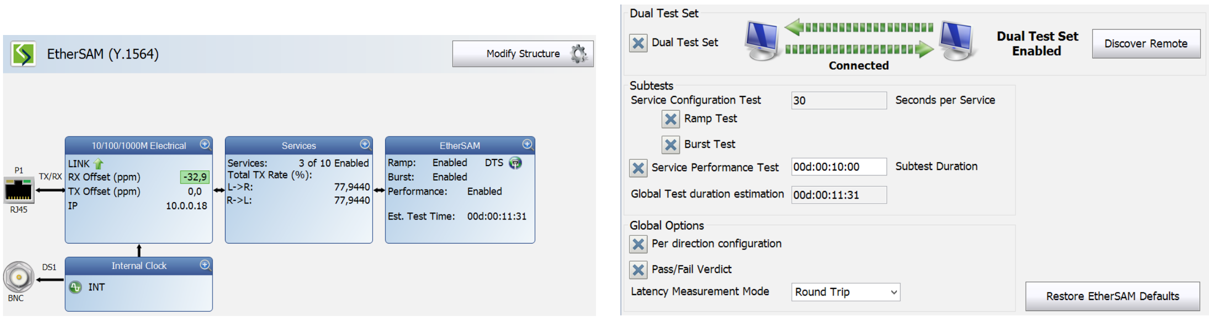

Figure 48: EtherSAM (ITU-T Y.1564) Test configuration in menu

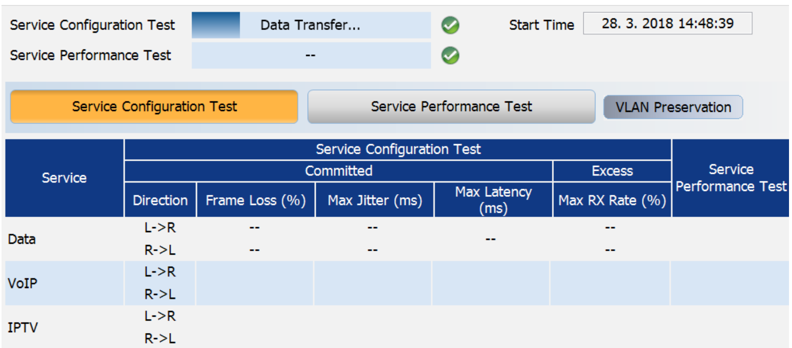

Figure 49: EtherSAM (ITU-T Y.1564) In Progress test applications

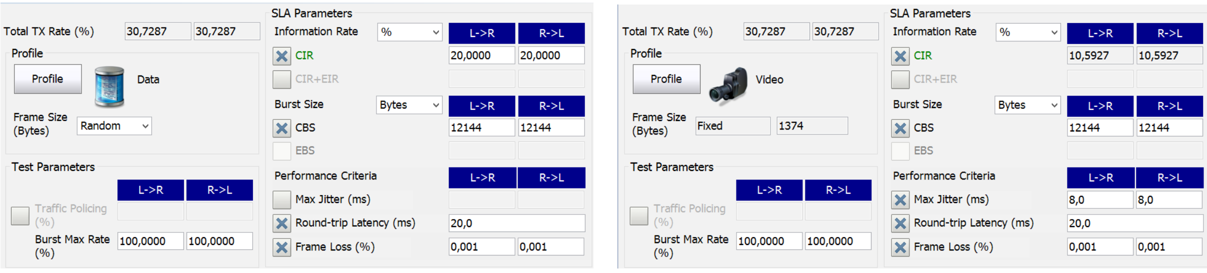

Figure 50: EtherSAM (ITU-T Y.1564) Configure Data and IPTV service

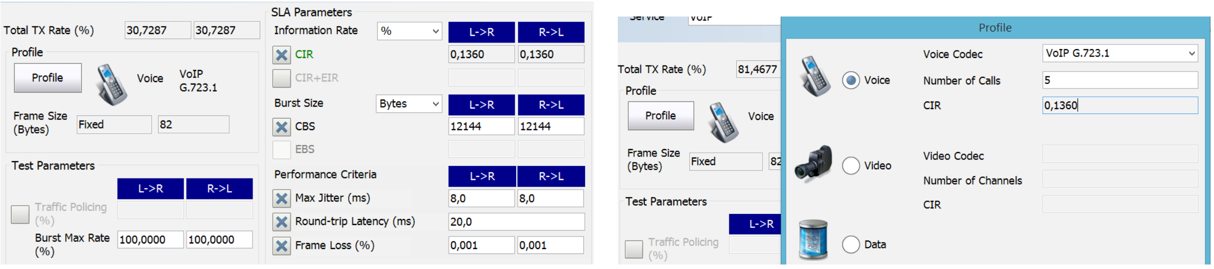

Figure 51: EtherSAM (ITU-T Y.1564) Configure VoIP service

7.2.1.1 EtherSAM (ITU-T Y.1564) VDSL2 - Average Throughput

The figures 52 and 53 shows average throughput for all services on profile 12 set on DSLAM. For IPTV, where the MPEG-4 HDTV codec was set, it had a bandwidth of around 10 Mbit/s. Changing the length of the metallic path to 1 km, this value dropped by approximately 9 Mbit/s. For data, the average throughput was 20 Mbit/s, and for a 1 km metallic path, this value dropped by 18 Mbit/s. For VoIP, the throughput is very low, and the change in path length has not had a major impact on this service.

The throughput of services is affected by the length of the metallic line. The impact of SOA is not noticeable. The length of the optical path does not have a big impact on the resulting values. Guaranteed values are set to 20, 0.136 and 10.5927 Mbit/s.

Average throughput is only 20 Mbit/s maximum since the CIR has been set for testing. The results from the measurements are also shown below.

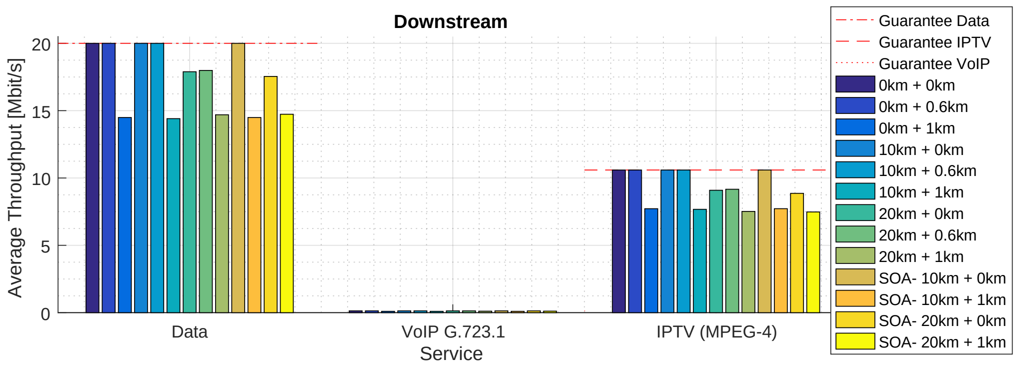

Figure 52: EtherSAM VDSL2- Average Throughput- Downstream

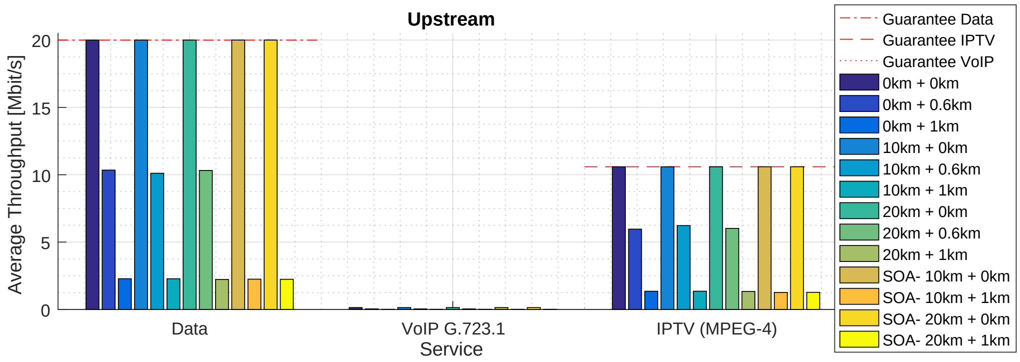

Figure 53: EtherSAM VDSL2- Average Throughput- Upstream

7.2.1.2 EtherSAM (ITU-T Y.1564) VDSL2 - Average Jitter

The figure 54 and 55 shows the average jitter of each service. Optical network along with SOA does not affect jitter quality. The lowest average jitter was measured by VoIP in downstream and upstream. The maximum jitter for all services was around 3 ms and at least 0.1 ms. Some values are low and do not have a major influence on the service function.

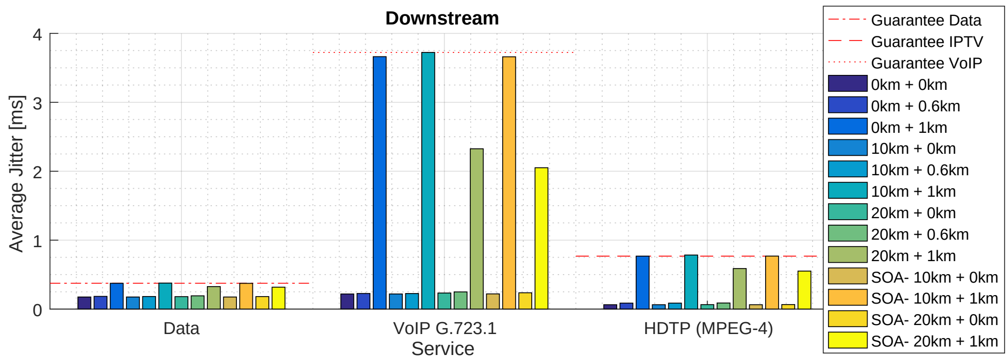

Figure 54: EtherSAM VDSL2- Average Jitter- Downstream

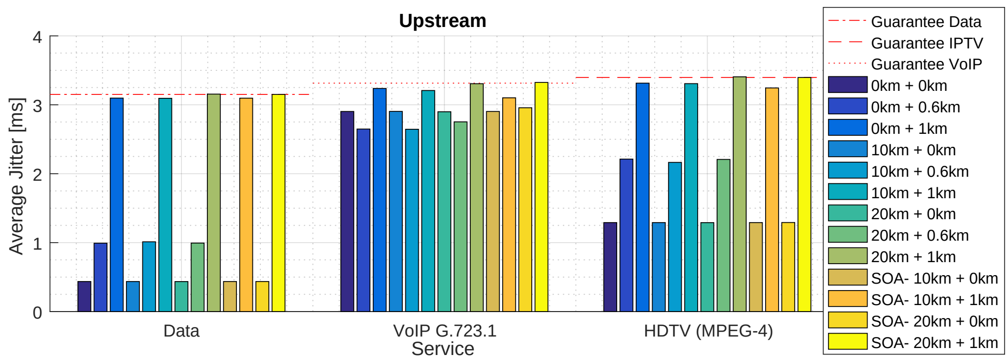

Figure 55: EtherSAM VDSL2- Average Jitter- Upstream

7.2.1.3 EtherSAM (ITU-T Y.1564) VDSL2 - Frame Loss

Figures 56 and 57 show frame loss of individual services. Fiber length has the greatest impact on video and data transfer. Downstream is a frame loss above 20% and upstream of over 80%. For a 0 km non-SOA metallic trace, the frame loss is 0%. Values not shown on the figure are 0% and are not visible.

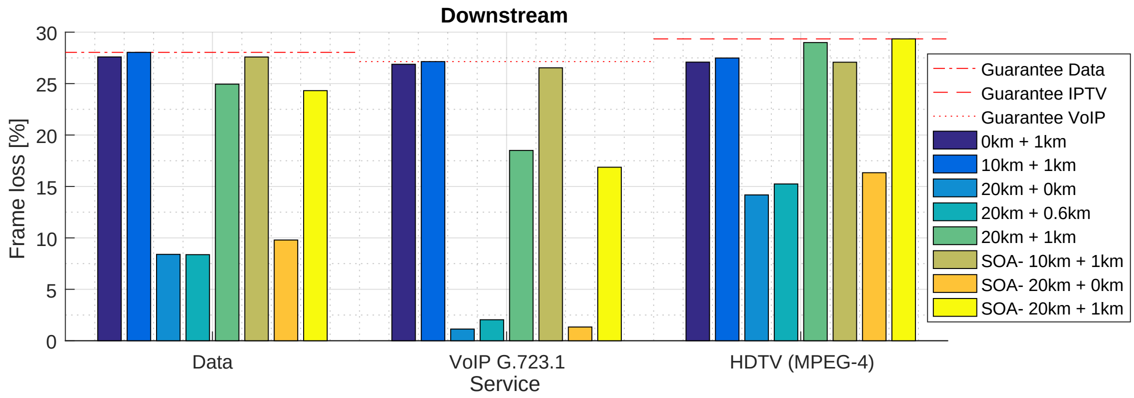

Figure 56: EtherSAM VDSL2- Frame Loss- Downstream

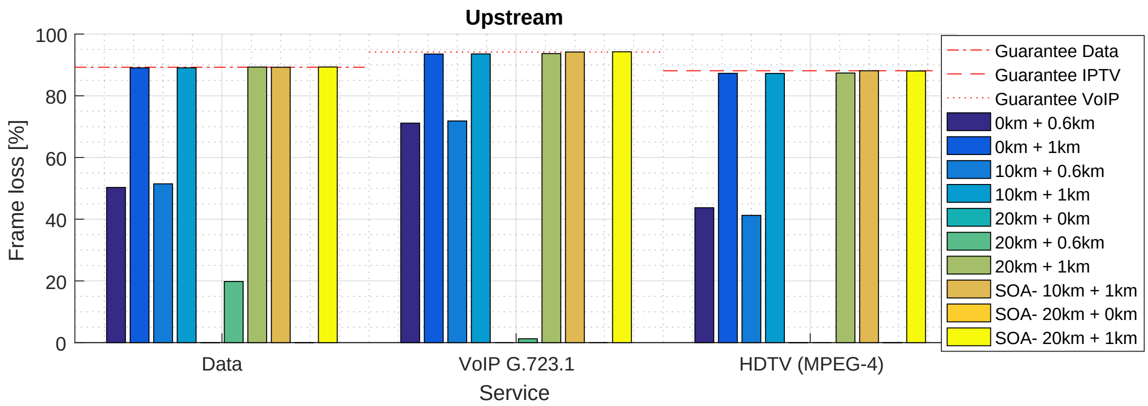

Figure 57: EtherSAM VDSL2- Frame Loss- Upstream

7.2.1.4 EtherSAM (ITU-T Y.1564) VDSL2 - Round-trip Latency

Figure 58 shows latency. Values for 0 km and 0.6 km metallic path are low and thus do not affect the operation of the set services. We can see that when setting a metallic path for 1 km, the delay is over 100 ms.Us of all three services, the latency is about 20 ms in the case of metallic lines 0 km and 0.6 km. With the introduction of SOA, latency is the same as without SOA. Service quality requirements are sufficient because the latency value is not over 150 ms.

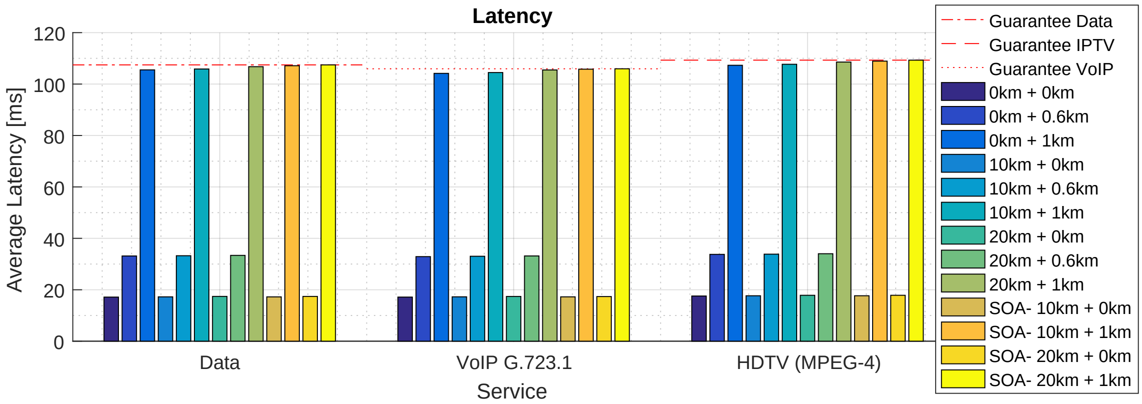

Figure 58: EtherSAM VDSL2- Round-trip Latency

7.3 RFC 6349

RFC 6349 is a framework used to confirm that the Ethernet service is able to properly carry TCP traffic published in 2011 that describes a practical methodology for measuring end-to-end TCP throughput in a managed IP network. IP Network operators and Service providers need to verify that their networks are well performing and meet the Service Level Agreements (SLA) with the customers. Standards RFC 2544 or ITU-T Y.1564 use to testing at Layer2 or Layer3 (Ethernet or IP layers).

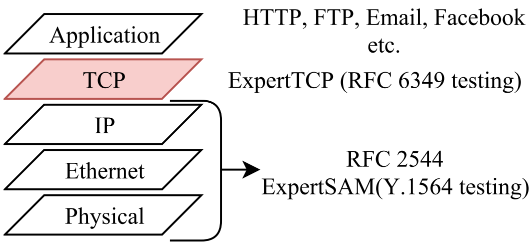

Figure 59: RFC 6349 layer structure

Though these tests are necessary, they are not sufficient, because they do not cover testing at TCP layer. Most web based applications like Http, FTP etc. run over TCP. Even many modern web applications like Facebook, Youtube, and the like use TCP.

RFC 6349 is based on the to measure TCP throughput, RTT and optimal window size. It has the capability to Generate and Analyze up to 12 UDP streams of traffic of various packet lengths. It also performs bi-directional TCP throughput measurements in combination with another unit at the remote location (other end of the network), that acts as the TCP server. Many real-world networks are not symmetrical [93].

The metrics that should be calculated from the measurement are [94]:

- Transfer time ratio – the ratio between the Actual TCP Transfer Time versus the Ideal TCP Transfer Time

- TCP Efficiency – the percentage of Bytes that were not retransmitted
- Buffer delay – represents the increase in RTT during a TCP Throughput test versus the inherent or baseline RTT.

7.3.1 Testing RFC 6349 for EPON/VDSL2

The RFC 6349 has to be executed in conjunction with a remote compatible module in RFC 6349 Dual Test Set mode allowing bidirectional testing. The Dual Test Set test provides independent results for each test direction. Supported Interfaces/Rates: 10M to 10G LAN. [98].

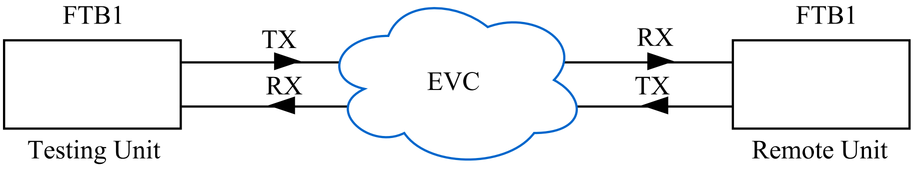

Figure 60: Typical RFC 6349 test application [98]

Test setup for RFC 3649 I set up some parameters:

The Local-to-Remote CIR and Remote-to-Local CIR represent the Committed Information Rate of the Ethernet Service under test. The CIR is not used to actually transmit frames at this rate but to calculate a Bandwidth Delay Product (BDP) which in turn is used to set the Max Window Size of the TCP connections. Rate Unit the unit used to display the rate values default set is Mbit/s we can set Gbit/s.

Max MTU (bytes) determines the Maximum Transfer Unit (MTU) to use when the client is generating TCP traffic toward the server.

TCP Throughput duration is the duration of the TCP Throughput phase per direction (1 minute to 30 days). Threshold (% of ideal) allows to enter the TCP hroughput as a percentage of the Ideal L4 Throughput.

The settings and in process test can be seen in the figure 61 and 62.

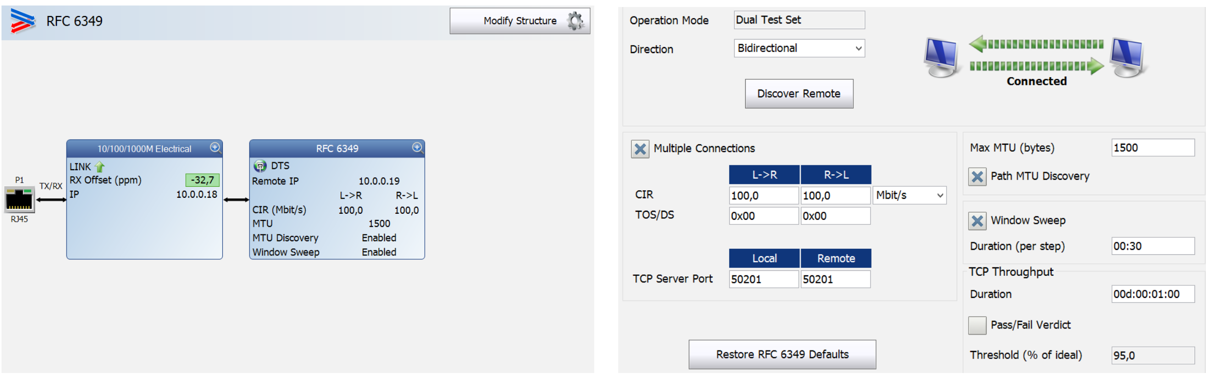

Figure 61: RFC 6349 Test configuration in menu

Figure 62: RFC 6349 In Progress test applications

7.3.1.1 RFC 6349 VDSL2 - TCP Throughput

The Figure 63 and 64 can see TCP Throughput on L4 (Transport layer). In Downstream, Throughput is larger than in Upstream. My conditions are fine for my testing. For comparison, I added the ideal throughput to the graph. When introducing SOA into the path and changing the length of the optical path, values significantly did not differ from SOA. The metallic trace was greatly negative, with 0.6 km and noise of -140 dB.

I tested the L4 network functionality and all the tests went well.

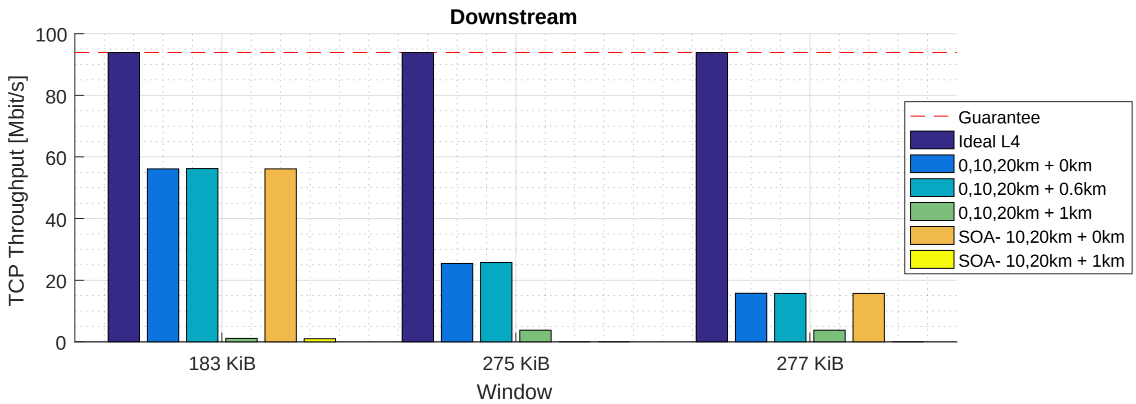

Figure 63: RFC 6349 VDSL2- TCP Throughput- Downstream

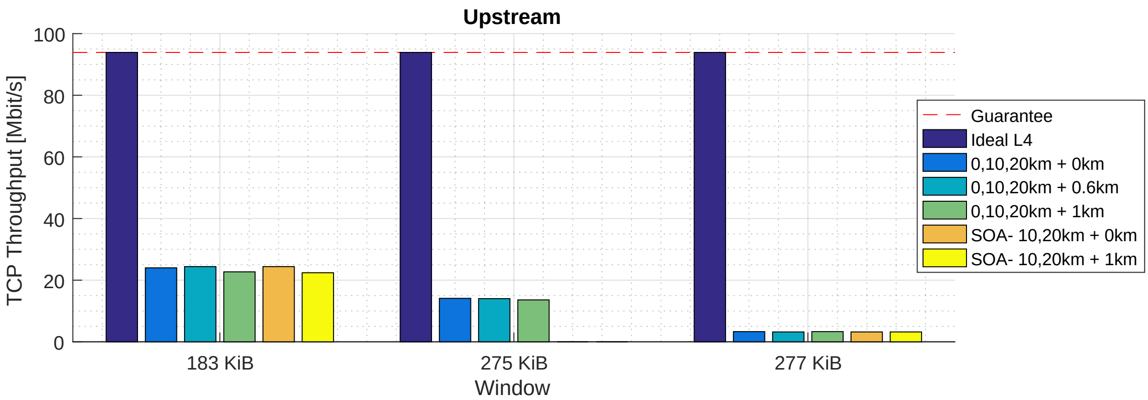

Figure 64: RFC 6349 VDSL2- TCP Throughput- Upstream

7.4 Network Integrity – Summary

The total measured values are show in the previous chapters and all test results are stored in the attachments. Some tests have proven to be inappropriate, but for my testing, the hybrid access network is sufficient. I made the measurements from EPON that was connected to the switch then into the optical path to the SOA and ONU unit.

Thats one about the optical part. The metal part was connected to VDSL where it went on to the simulator and then to the end user in the modem.

Tests were not performed on longer optical paths because the ONU unit was interrupted with EPON, so I chose testing only within 20km. For the metal part, I chose the best results.

With a 10-kilometer optical path, the test results were very poor, the neutral attitude they had with SOA. The biggest lead for all tests was the optical path I changed from 0, 0.6 and 1 km. Constant noise was set to -140 dB for greater approach to real routes.

8 Measured results of Optical topology

In this chapter, I present measured results for the optical part. I test here the quality measurement of Triple Play services. The practical measurements of the Tests were carried out in the network as mentioned in the previous chapters.

8.1 Optical Fiber Loss Measurement

This chapter is it uses measurement on the optical path. I measure the different paths of each path to determine the path quality. The path is very old and I have to to take care of aging of optical fibers.

8.1.1 Optical Time Domain Reflectometer (OTDR)

Measurement of optical path attenuation was performed using the OTDR method. The Rayleigh scattering method. Transmitted optical signal is scattered back in different points of the fiber back to the fiber entry. The optical fiber used is of type G.652.A. The optical paths are composed of optical fibers of length 5, 10, 15, 20, 25, 30, 35 km. These fibers are more than 20 years old and are interconnected a series of welds and connectors. The attenuation values are significantly higher.

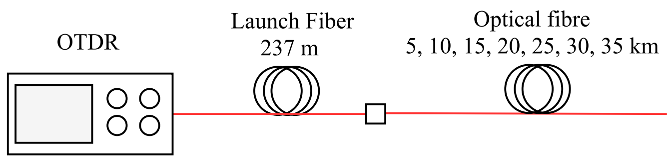

Figure 65: Topology for OTDR measurement

The average value of the specific attenuation calculated from all individually measured optical paths. For the measurement, a precursor fiber of 237 m is required to suppress the input dead zone. The 16 table shows the measured optical path values.

Table 16: Attenuation of optical paths

	1310 nm		1550 nm	
L $[km]$	IL $[dB]$	α $[dB/km]$	IL $[dB]$	α $[dB/km]$
5.0	2.863	0.543	3.854	0.731
9.8	5.695	0.567	5.809	0.578
14.7	7.211	0.481	8.920	0.655
19.7	7.297	0.369	11.049	0.558
24.7	14.277	0.573	10.883	0.437
29.5	16.989	0.569	12.971	0.470
34.4	19.078	0.548	14.009	0.405

A reflectogram is shown in the image 65. The first part is used launch fiber with a length of 237 m. This is followed by a 27 m long fiber that is used to interconnect the EB315 and EB316 laboratories, followed by an optical path. Events 4 to 13 are fiber connections. Event 14 is a reflection at the end of the path.

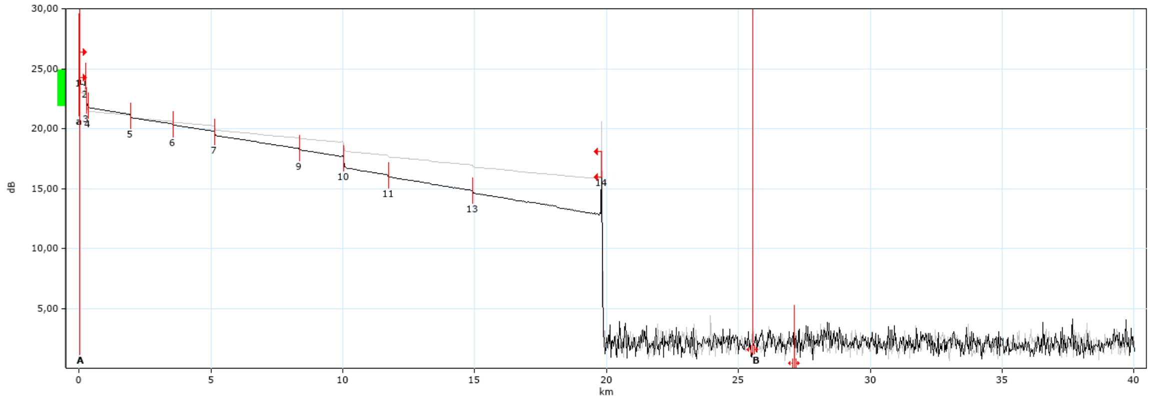

Figure 66: Reflectogram for optical path 19.8 km

8.1.2 Splitter Insertion Loss Measurement

I use in topology many kind splitters. For example 1:2 splitter (SFT-SWB-02x02-50-CM1-NPC-NPC) show in table 17. Attenuator (27.62 dB) is used to protect the end unit from high power and gain. I used Method 1C, the scheme is shown in Fig. 67 and practice is show on Fig. 68.

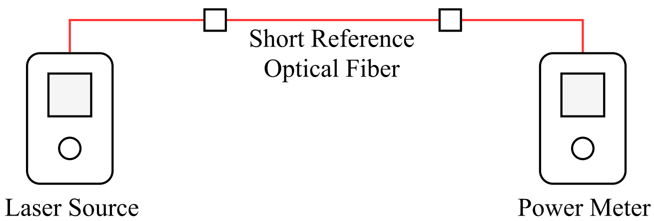

Figure 67: Topology for method 1C measurement

Table 17 shows the average values between ports 1, 2 and 3 in Splitter 1:2.

Table 17: Measured insertion loss between Splitter ports

	Insertion Loss [dB]			
	Splitter 1:2		**Splitter 1:2**	
λ [nm]	**1 ->2**	**1 ->3**	**2 ->1**	**3 ->1**
1310	3.55	3.98	3.08	4.01
1550	3.48	3.67	3.54	3.98

The WDM Splitter is used with SOA and is used to divide wavelengths 1410 for download and 1310 for upload.

The last of the splitters used is the coupler dividing into $\sum$, 90% and 10%. This coupler is used to simplify the use of OSA devices in order to see downstream and upstream concurrently. I used the 4 couplers. Coupler split 90:10, where 10% of the optical power travels to OSA and 90% of the optical power travels to continue along the path.

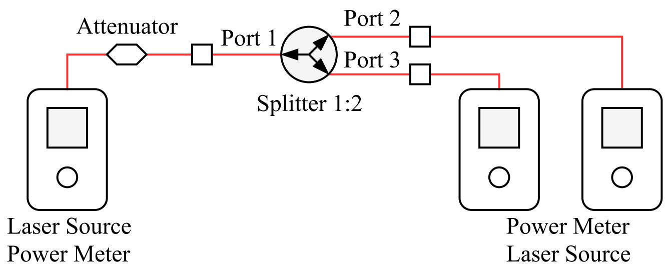

Figure 68: Topology for measure insertion loss of optical splitters 1:2

The following attenuation is measured by topology 69 and 65. The measured values can be seen in the table 18. The last table 19 shows the attenuation of only one component. All measurements are performed on both sides and averaged.

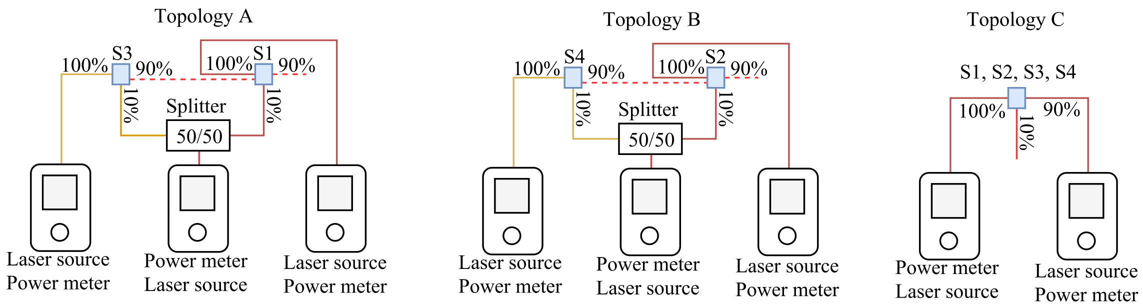

Figure 69: Two topology for measuring attenuation within the WDM splitter

Table 18: Measured insertion loss between track couplers and splitters see on topology A, B

	Insertion Loss [dB]			
λ **[nm]**	**S1->50/50**	**S2->50/50**	**S3->50/50**	**S4->50/50**
1310	14.0	13.2	12.9	11.9
1550	13.6	13.3	12.7	11.9

Table 19: Measured insertion loss between four couplers see on topology C

	Insertion Loss [dB]			
λ **[nm]**	**S1**	**S2**	**S3**	**S4**
1310	0.53	0.94	0.12	0.13
1550	0.44	0.82	0.28	0.31

To find out of gain G I have found the total attenuation **S1** and **S3**, which is about **12.72 dB** and **S2** and **S4** is **13.00 dB**. All values are the average of the tables 18 and 19.

8.1.3 Chromatic Dispersion Measurement (CD)

The CD is created because of the different speeds of different spectral components in the optical fiber. This means that some spectral components will be delayed over others. It is made only for single-threaded fibers. The CD module (FTB-5800B) and the broadband laser source (FLS-5800A) were used for the measurements. The topology is shown in Fig. 70.

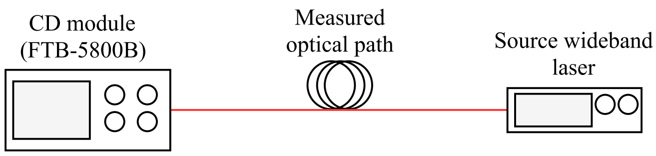

Figure 70: Topology for CD measurement

For each optical path, the fiber length was set, which I measured at OTDR. The table 20 shows measured CD for each path at 1550 nm.

Table 20: Measurement of CD on optical paths λ 1550 nm

L $[km]$	**CD** $[ps/(nm \cdot km)]$	**Slope** $[ps/nm^2]$
5.0	16.708	0.302
9.8	16.423	0.580
14.7	16.863	0.873
19.7	16.535	1.152
24.7	16.608	1.448
29.5	16.483	1.773
34.4	16.677	2.070

This measurement of the Chromatic Dispersion is a phase shift method recommended as reference method to CD measurement. Figure 71 shows the curve of chromatic dispersion coefficient and relative group delay in dependence on wavelength.

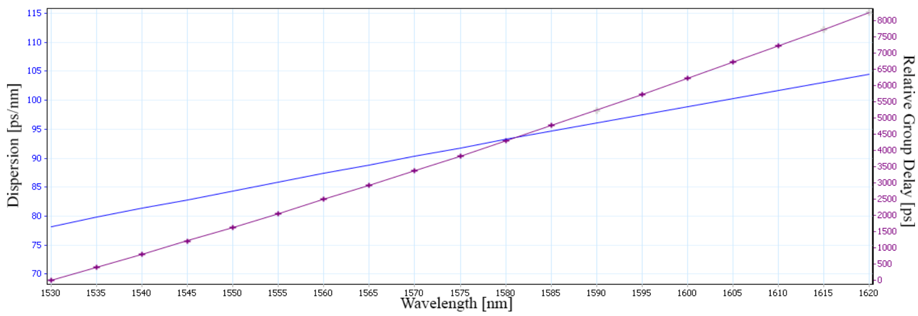

Figure 71: Result from CD analyzer on 5 km

8.1.4 Polarization-Mode Dispersion Measurement (PMD)

Polarization-Mode Dispersion created because of the different path lengths of both polarization plane modes when passing through the fiber. PMD was measured on the same optical paths as the CD, and the PMD module (FTB-5500B) and broadband laser source (FLS-5800A) were used for the measurement. For all optical trace has to be set the length of the fiber from OTDR reports. The topology for PMD measurement is shown in Figure 72.

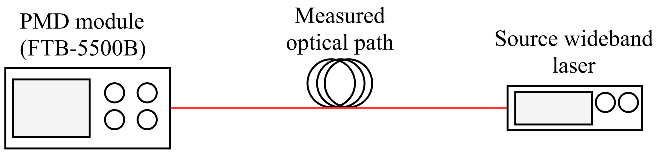

Figure 72: Topology for PMD measurement

Table 21 shows the measured polarization-dispersion values for each path. The guaranteed maximum value for PMD is 0.5 $ps/km^{1/2}$ but fibre is old more 20 years and many splices creates high values.

Table 21: Measured PMD coefficents and PMD value

L $[km]$	**PMD value** $[ps]$	**PMD coefficient** $[ps/nm^{1/2}]$
5.0	1.934	0.862
9.8	0.345	0.110
14.7	1.963	0.512
19.7	1.807	0.407
24.7	2.214	0.445
29.5	1.713	0.315
34.4	2.380	0.4055

The Figure 73 shows the interferogram from the PMD analyzer (FTB-5500B) for the fiber length of 5 km. Interferogram show delay $[ps]$ and intensity $[\%]$.

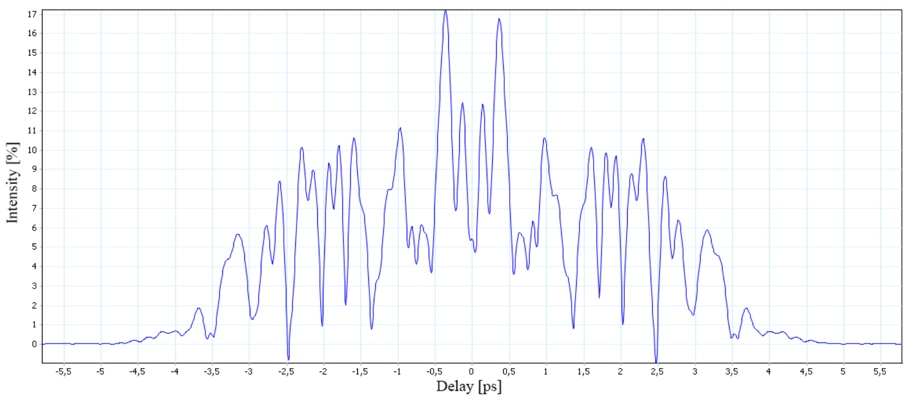

Figure 73: Result interferogram from PMD analyzer on 5 km

8.2 Spectral analysis of the optical network

This chapter focuses on testing the optical spectrum between OLT and ONU at different distances. I use here WDM splitters for downtream 1490 nm and upstream 1310 nm.

The first measurement took place on the optical path between OLT and ONU. where I measured PON Power Meter optical performance. The measurement results can be seen in the table 22.

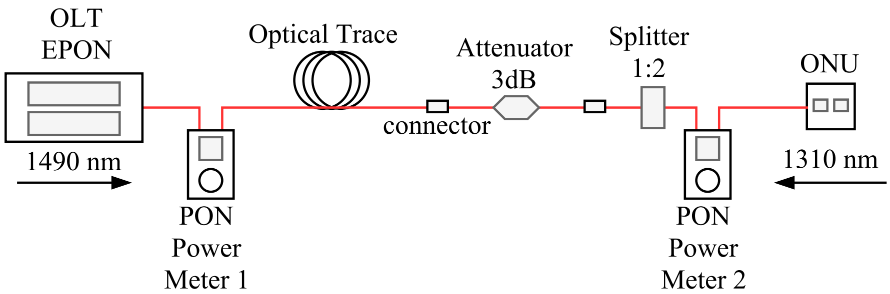

Figure 74: Topology for optical spectrum measurement without SOA

Table 22: Measured values from PON power meters (PPM) between OLT and ONU

	Optical Power [dBm]			
	PPM1		**PPM2**	
L [km]	**λ 1490 nm**	**λ 1310 nm**	**λ 1490 nm**	**λ 1310 nm**
0	3.7	-7.8	-6.7	3.3
5	3.7	-13.6	-11.9	3.3
10	3.7	-16.7	-14.4	3.3
15	3.7	-21.1	-17.4	3.3
20	3.7	-22.1	-18.4	3.3
25	3.7	-25.3	-20.5	3.3
30	3.7	N/A	-22.7	N/A
35	3.7	N/A	-25.8	N/A

Spectrum analyzer EXFO FTB-5240B was used to measure the optical spectrum. I used a 90:10 optical divider to measure. This means that 90% of the signal is transmitted over the network and 10% of the signal is into an optical spectral analyzer. The topology for measuring the optical spectrum is shown in Figure 75.

Table 23 show measured PON Power meter 1 and 2 between OLT and ONU. Figure 76 and 77 show Optical spectrum in Downstream and Upstream from OSA 1. Distance 30 and 35 km in wavelength 1310 nm Upstream is unknown.

The values from PPM1 and PPM2 were measured on a real path. Values are calculated with attenuators on the optical path. The downstream optical spectrum is not symmetrical. This is caused by a source from OLT, where it is not always accurate. This is only due to the optical path 1 because of poor results in optical path 2.

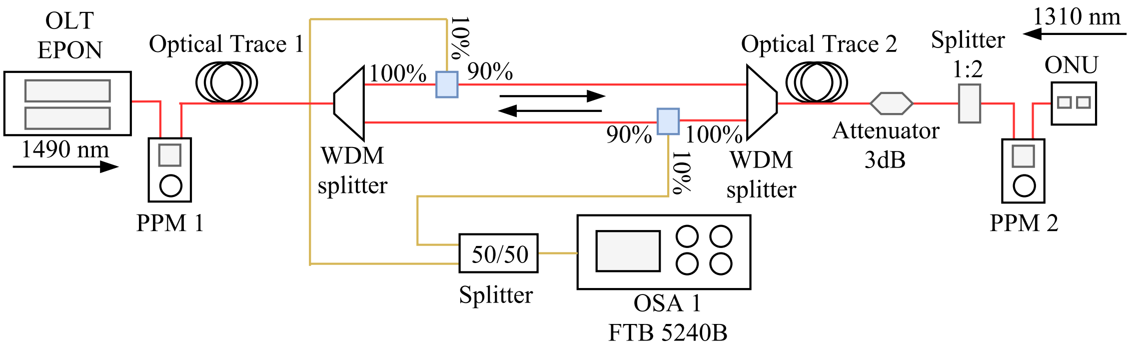

Figure 75: Topology for optical spectrum measurement without SOA

Table 23: Measured values from PON power meters (PPM) between OLT and ONU with WDM splitters

	Optical Power [dBm], Optical Trace 1				Optical Power [dBm], Optical Trace 2			
	PPM1		PPM2		PPM1		PPM2	
L [km]	λ 1490 nm	λ 1310 nm	λ 1490 nm	λ 1310 nm	λ 1490 nm	λ 1310 nm	λ 1490 nm	λ 1310 nm
0	3.7	-10.6	-10.2	3.3	3.7	-11.1	-10.7	3.3
5	3.7	-15.1	-14.3	3.3	3.7	-15.4	-14.5	3.3
10	3.7	-17.5	-16.3	3.3	3.7	-17.3	-15.9	3.3
15	3.7	-22.1	-19.7	3.3	3.7	-22.5	-19.8	3.3
20	3.7	-23.1	-20.6	3.3	3.7	-22.8	-20.7	3.3
25	3.7	-28.6	-24.2	3.3	3.7	-27.9	-23.6	3.3
30	3.7	N/A	-26.1	N/A	3.7	N/A	-25.3	N/A
35	3.7	N/A	-29.6	N/A	3.7	N/A	-29.2	N/A

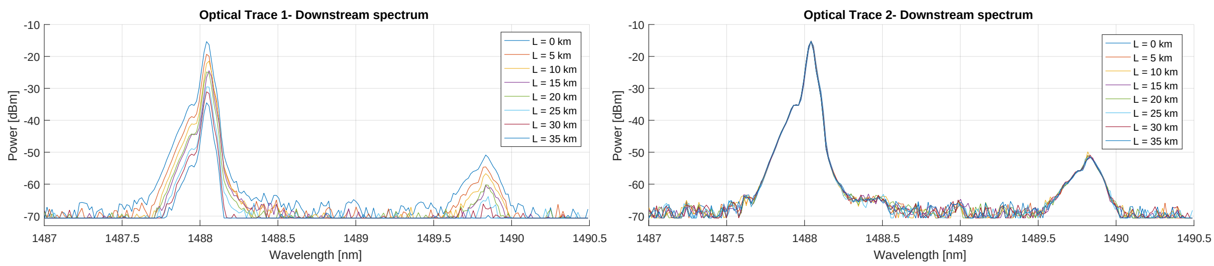

Figure 76: Downstream optical spectrum - optical trace 1 and 2 (0 - 35 km)

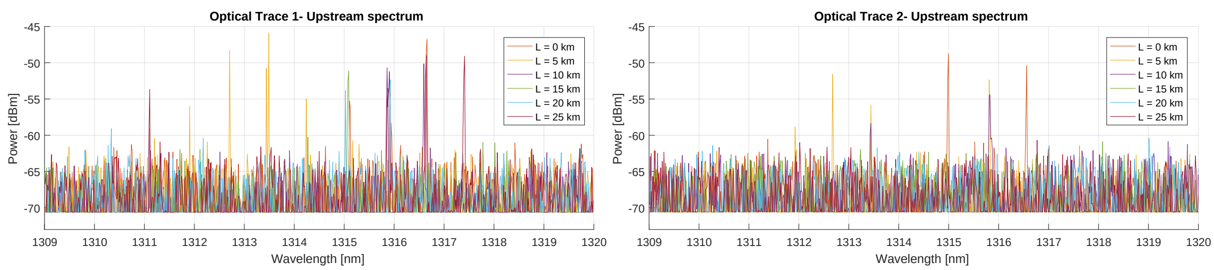

Figure 77: Upstream optical spectrum - optical trace 1 and 2 (0 - 25 km)

8.2.0.1 Summary Spectral analysis of the optical network

where path before and behind the WDM splitter is from the perspective of the PON Power meter almost identical. The wavelength signal at 1310 nm is more attenuated than the signal at a wavelength of 1490 nm. This is given by the WDM splitter, which divides the band O-band and L-band. The ONU responded to a maximum distance of 25 km.

It can be seen from the table 22 and 23 that the attenuator (3dB) together with the WDM splitter and clutches has reduced the ultimate Optical Power. From figure 76 and 77 it is evident that the optical trace 2 does not affect the beam of the spectrum at the wavelength of 1490 nm because the path is after the optical spectral analyzer. The optical spectrum on the optical trace 2 is the same at 1490 nm and 1310 nm is bad. Optical traces can be seen from the topology fig. 75.

8.2.1 Semiconductor Optical Amplifiers analysis

To amplify the optical signal, they were used two Semiconductor Optical Amplifiers. One type of S-band SOA that amplifies downstream direction and the second type of O-band SOA that amplifies upstream.

Sample of SOA functional control in Windows. The figure 78 shows the found SOA ports for S-band and O-band. With this software, SOA wakes up. Graphically, temperature, Optical Power and forward voltage are displayed. Both SOAs have different settings for maximum current limit.

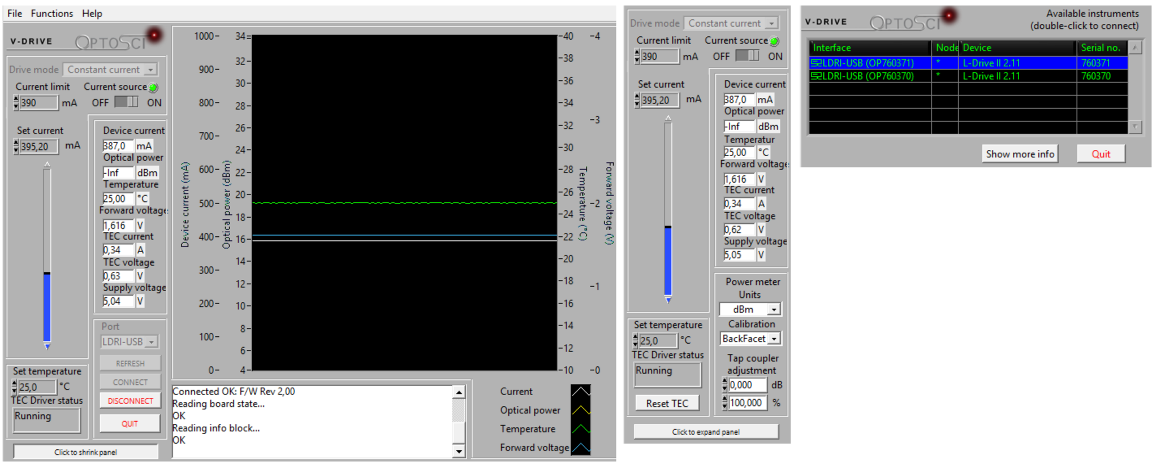

Figure 78: Real time setting SOA

SOA the O-band (λ 1310 nm) is manufactured by Thorlabs with part number BOA1017S. SOA the S-band (λ 1490 nm) is manufactured by Aeon with part number SASH 24P151. The SOA properties obtained from the datasheets are listed in Table 24.

Table 24: Properties of SOA used

Type SOA	Operating current [mA]	Central Wavelength [nm]	3 dB Bandwidth [nm]	Average Noise Figure [dB]	Saturation Output Power [dBm]	Peak Gain [dB]
SASH 24P151	390	1470.0	64.5	7.0	11.6	29.4
BOA 1017S	600	1319.1	79.5	6.5	16.9	30.3

Topology of the overall connection The SOA optical parts are shown in Figure 79. On one SOA only, you can see the power input and output of OSA for downstream and upstream.

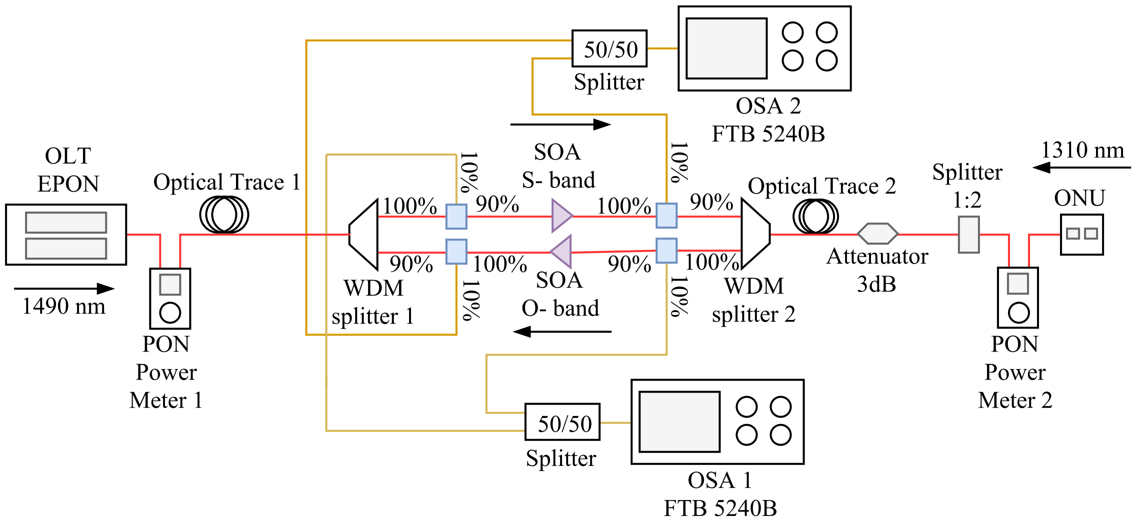

Figure 79: Topology for optical spectrum measurement with SOA

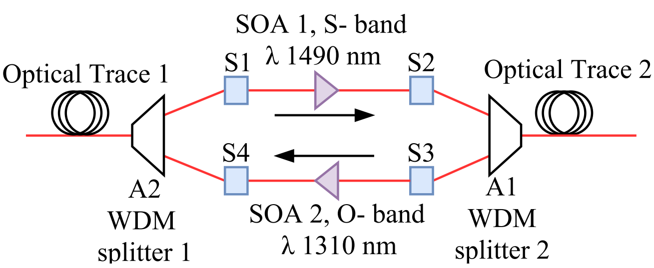

Figure 80: Specific topology inside WDM splitters for SOA measurement

Table 25 represents the values of the PON power meter. PPM1 is at the start of the path and PPM2 at the end behind the attenuator and splitter. The measurement was in progress in two variants on the optical path 1 and the optical path 2 recorded in the diagram 79. Optical path 2 is very inappropriate for further testing, so I will only use option 1 in the measurement.

Table 25: Measured values from PON power meters with OLT, ONU, SOA and WDM splitters

	Optical Power [dBm], Optical Trace 1				**Optical Power [dBm], Optical Trace 2**			
	PPM1		**PPM2**		**PPM1**		**PPM2**	
L [km]	**λ 1490 nm**	**λ 1310 nm**	**λ 1490 nm**	**λ 1310 nm**	**λ 1490 nm**	**λ 1310 nm**	**λ 1490 nm**	**λ 1310 nm**
0	3.7	6.3	3.1	3.3	3.7	5.8	1.2	-16.4
5	3.7	6.1	1.9	-8.2	3.7	6.9	-2.4	-18.9
10	3.7	3.1	1.2	-6.2	3.7	-5.4	-3.7	-20.6
15	3.7	-0.8	0.7	-5.6	3.7	-6.5	-6.7	-25.6
20	3.7	-1.2	0.3	-5.8	3.7	N/A	-7.4	-25.8
25	3.7	-6.6	-0.6	-6.1	3.7	N/A	-10.4	-30.6
30	3.7	-8.1	-1.4	-6.5	3.7	N/A	-12.3	-32.4
35	3.7	-13.2	-2.7	-7.2	3.7	N/A	-15.9	-37.2

Table 26 show the measured values from the 1490 wavelength spectral analyzer. The values are measured using a spectral analyzer. SOA gain is calculated $P_{OUT} - P_{IN}$ values. Optical spectral analyzes are shown for optical traces 1 and 2.

Table 26: Measured values from OSA 1 and OSA 2 with SOA on λ 1490 nm

	OSA 1			**OSA 2**			$P_{OUT} - P_{IN}$
L [km]	**λ [nm]**	**Power [dBm]**	**OSNR [dB]**	**λ [nm]**	**Power [dBm]**	**OSNR [dB]**	**G [dB]**
0	1488	-13.68	30.84	1488	-0.91	33.91	12.77
5	1488	-18.63	29.82	1488	-2.42	31.28	16.21
10	1488	-20.77	28.74	1488	-2.86	28.51	17.91
15	1488	-24.17	23.3	1488	-4.23	25.8	19.94
20	1488	-23.32	30.51	1488	-3.89	28.59	19.43
25	1488	-27.23	29.56	1488	-5.49	26.8	21.74
30	1488	-28.95	29.83	1488	-6.76	25.89	22.19
35	1488	-32.77	28.48	1488	-9.16	25.09	23.61

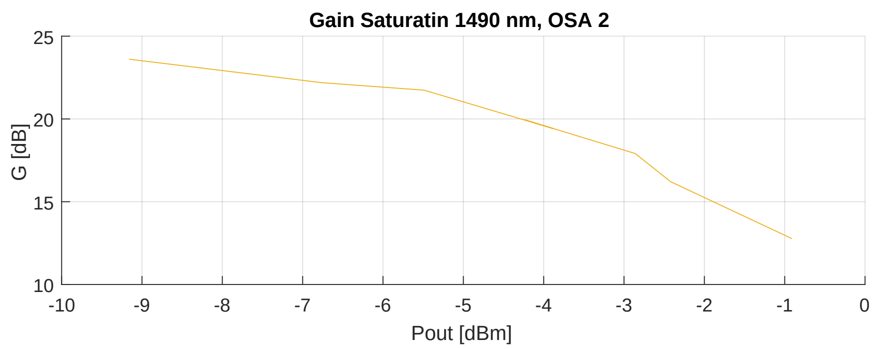

Figure 81: Gain Saturation for downstream λ 1490 nm

Figure 81 how gain saturation for downstream with value of power versus gain on wavelength of 1490 nm. From the optical spectrum 2. These are values P_{out} and G.

In the figure 82 are 4 charts are displayed. Graphs show Optical path 1 and 2 in Downstream direction. Everyone has a length of between 0 and 35 km. Optical path 2 was very unstable, as can be seen in the charts below.

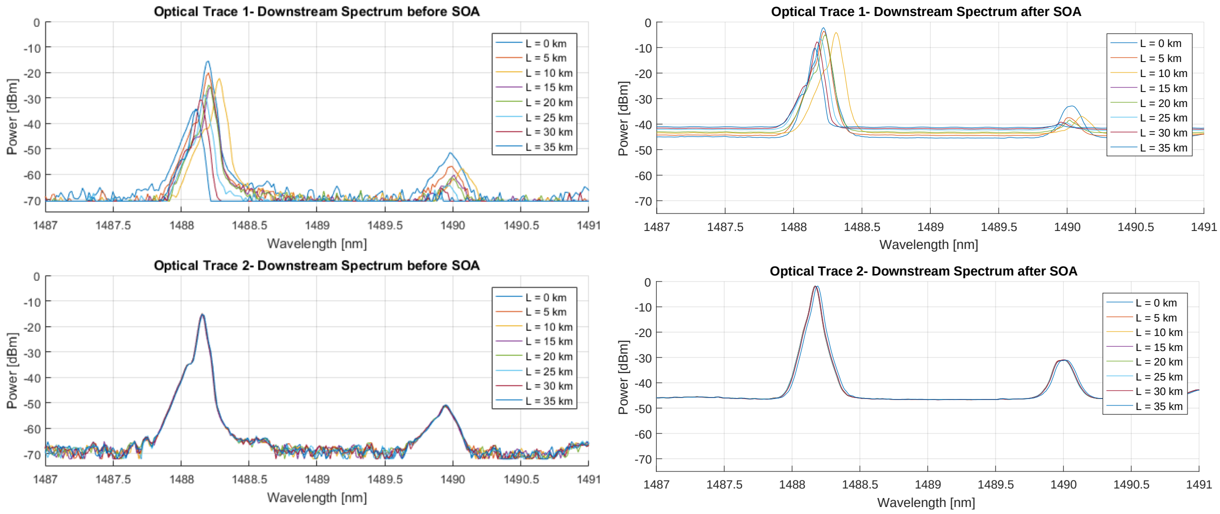

Figure 82: Downstream OSA before and after SOA - optical trace 1 and 2 (0 - 35 km)

Figure 83 shows 4 charts are displayed. The graphs show Optical path 1 and 2 in Upstream direction. Each one over a length of 0 to 35 km. After amplification the signal was increased power. Optical path 2 was unstable as in previous charts, so I will use the optical path just beyond EPON for further measurements.

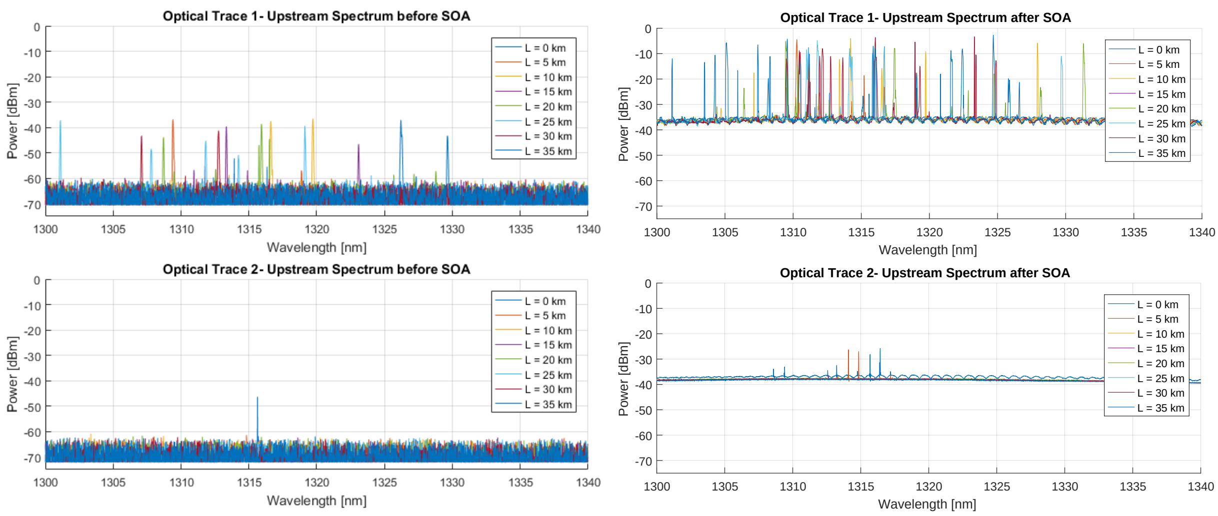

Figure 83: Upstream OSA before and after SOA - optical trace 1 and 2 (0 - 25 km)

8.2.2 Summary with and without SOA on spectral analysis

At increasing the optical path increases noise and attenuation. Even when using amplifiers, optical noise rises when the optical path is lifted. In some cases, the optical paths failed to register UNO with PON. In the case of Optical Trace 2, it was not registered for 10 and more km. This may be caused by a small OSNR when the photodetector is unable to detect the signal. EPON is very limited in upstream, because the downstream has enough energy, so ONU can detect an optical signal at a wavelength of 1490 nm. The table 27 shows the basic damping elements of the entire topology. the attenuation of the optical path depended on the distance.

Table 27: Resulting attenuation

Component	Attenuation [dB]
Attenuator	3
Power meter 1, 2	0.5
WDM splitter 1	0.68
WDM splitter 2	0.65
Splitter 1:2	3.1
Connectors	0.5

I performed two measurements for the optical trace 1 and 2. From the total the more appropriate measurement conditions were based on the optical track 1, the path was first, followed by amplification of the signal. Otherwise, when the signal grew earlier, the losses were greater.

From measurements, the O-band can range between 1260 and 1360 nm and the S-band in the range (1460 - 1530 nm).

The figure 84 shows the optical spectrum without amplification and amplification. The Optical Spectrum on Optical Trace 1 is displayed on a 0 - 35 km path. This is an upstream direction.

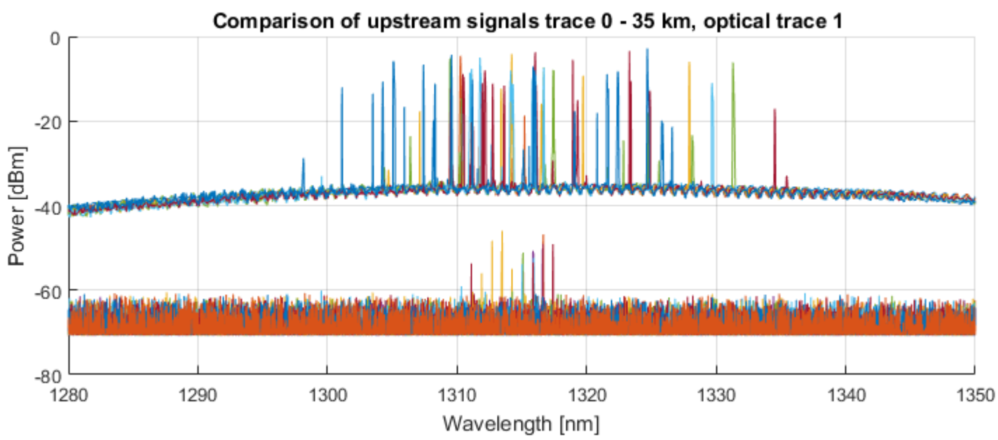

Figure 84: Comparison of upstream signals on optical trace 1 with and without SOA

The figure 85 represents spectral analysis of the optical path 1 for the dowstream direction. The optical spectrum shows directions 0 to 35 km before and after amplification. Optical spectrum in Downstream and Upstream is not symmetrical. This is caused by a source from OLT, where it is not always accurate. Also, a large approach to the spectrum causes and displays all deficiencies.

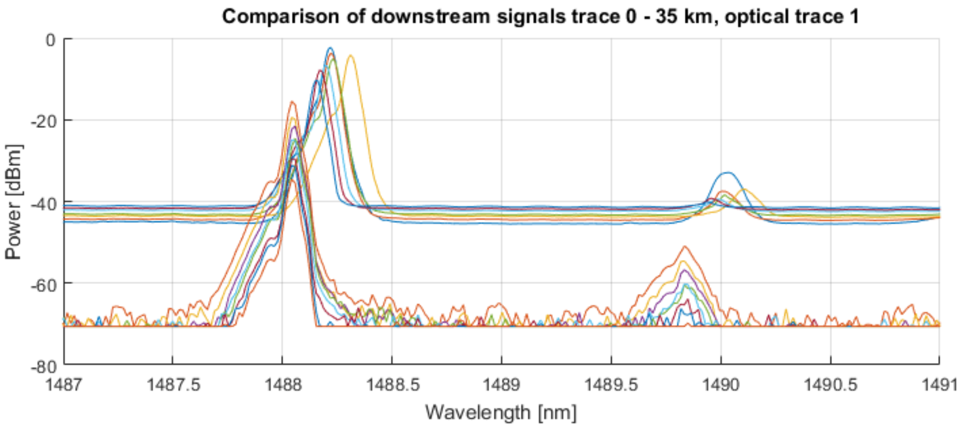

Figure 85: Comparison of downstream signals on optical trace 1 with and without SOA

The figure 86 compares the original optical signals against the amplified optical signals. On the left graph, the optical spectrum measured with the optical trace 1 and the right optical spectrum measured on the optical track 2. You can see the original upstream and downstream optical signal after 5 km of optical trace and amplify the original optical signal. The graphs show the full O-band and S-band together in one chart. Amplified is not only the optical signal but the noise is amplified too. All values are measured on a 0 to 35 km trace.

ONU was unable to register for optical track 2 at 10 km. There was a lot of damping and SOA did not help. On optic trace 2, the ONU captured over 35 km but the real signal for service use is lossy.

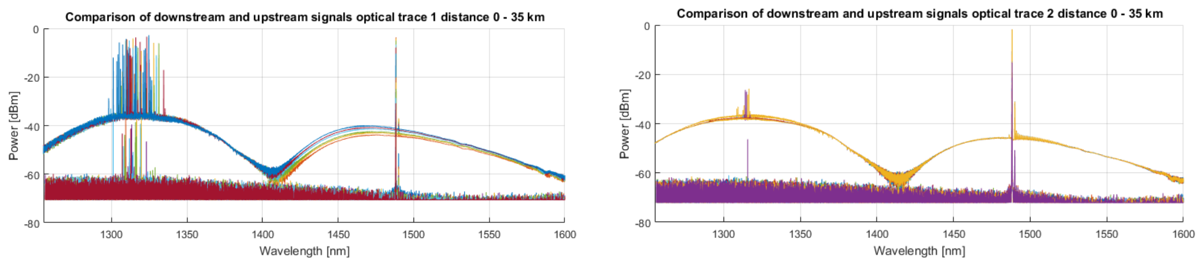

Figure 86: Comparison of downstream and upstream signals on optical trace 1 and 2

8.3 Hybrid xPON/xDSL

In the figure 87 you can see the basic topology for Hybrid xPON/xDSL. The figure 87 shows the optical and metallic parts. OLT, VDSL2 and Managment simulator are located in other laboratories and interconnected. Triple play services have been used in this part of the topology.

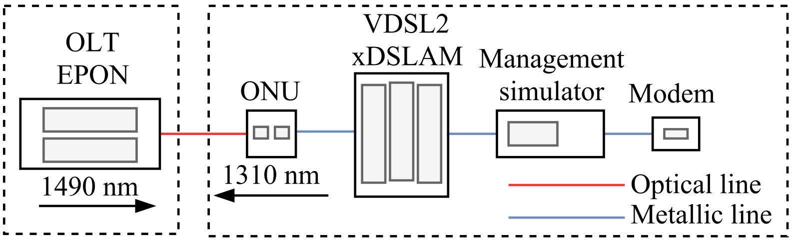

Figure 87: Topology for hybrid xPON/xDSL network

8.3.1 VDSL2 Link Rates

To measure transmission properties, a modem was used to display the maximum transmission speeds to console output. All profiles were measured, which were located on individual ports. The profiles were set on VDSL2 DSLAM. The integrity of the network was performed on port 12 and the subsequent features of each service. The measurement was from VDSL2 via a simulator that contained ports 1 to 12, where 1 is a management port. Modem and then PC console output. The schema is shown in Fig. 87.

In the table 28 you can see the measured speeds on the individual ports. The table 29 shows the measured bit rates when changing the metallic line for the selected profile 12 to VDSL2.

Table 28: Measured profiles for VDSL2 at 0 km and noise OFF

	Set speeds [Mbit/s]											
Ports	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**
Downstream	0.480	0.960	1.952	3.968	7.968	16.352	23.968	31.968	45.408	59.968	60.704	63.968
Upstream	0.480	0.960	1.952	3.952	7.968	16.352	23.968	31.968	35.968	35.968	35.968	35.968

Table 29: Measured transmission speeds for the profile on port 12 for VDSL2

Noise [dB]	**OFF**	**-140**	**-130**	**-120**	**-110**	**-100**	**-90**
L [km]	**Downstream/Upstream [Mbit/s], Port 12**						
0	63.968/35.968	63.968/35.968	63.968/35.968	63.968/35.968	59.264/35.968	42.816/29.952	23.648/16.896
0.2	63.968/35.968	63.968/35.968	63.968/35.968	62.080/35.968	51.072/32.064	33.536/19.968	11.168/5.504
0.4	55.616/34.272	54.560/34.016	54.144/32.832	50.336/26.144	37.120/13.792	17.76/4.032	6.656/0.224
0.6	41.408/17.792	41.280/17.056	40.096/15.744	32.768/7.552	22.528/3.840	12.192/0.480	3.072/0.192
0.8	28.192/7.104	28.160/6.784	27.488/5.984	23.648/3.264	15.328/0.224	6.400/0.210	1.280/0.192
1	22.240/3.908	22.144/3.680	21.312/2.496	18.208/0.256	9.728/0.120	4.000/0.0480	0/0

Transmission speeds decreasing with increasing length metallic line. The higher the noise, the worse the quality. The figure 88 and 89 shows the dependency transmission speeds on length metallic lead for all noise. Measurement was performed using a modem that shows the current speed in downstream and upstream when telnet is connected. The procedure can be seen in Figure 90.

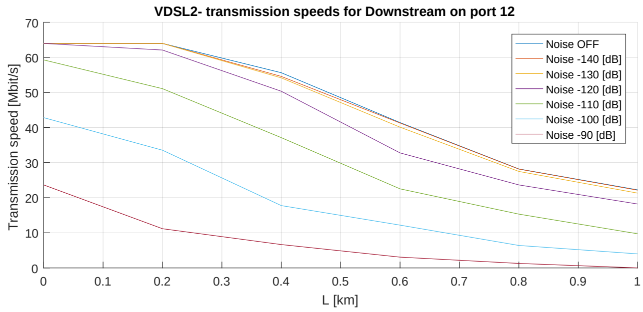

Figure 88: Downstream transmission speeds dependence on length with noise in VDSL2

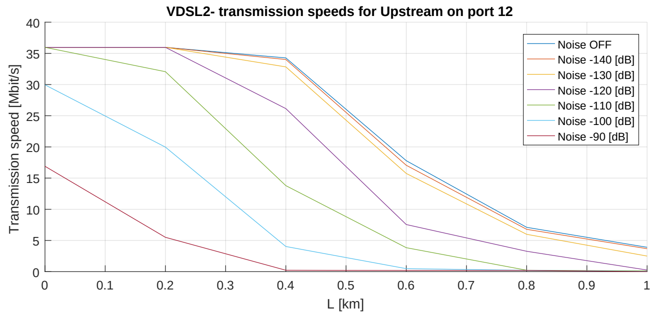

Figure 89: Upstream transmission speeds dependence on length with noise in VDSL2

I found the link rate by connecting to the modem using PuTTy via Windows and using telnet via Linux. After successful connection with using password: 1234, I chose `24. System Maintenance` then `Command Interpeter`, where I got to the command line. The `vdsl status` command displays the information about the modem including transmission speeds. The `DS Payload Rate` and the `US Payload Rate` have found speeds for the downstream and the upstream direction.

VDSL2 uses higher frequencies (up to 30 MHz), there are large attenuations at increased wire length. A slight deterioration can be expected with a 50 m length.

```
Copyright (c) 1994 - 2006 ZyXEL Communications Corp.
P-870MH-C1 Main Menu

Getting Started              Advanced Management
  1. General Setup             23. System Password
  3. LAN Setup                 24. System Maintenance
--------------------------------------------------------------------
              Menu 24 - System Maintenance
              1. System Status
              2. System Information and Console Port Speed
              3. Log and Trace
              7. Upload Firmware
              8. Command Interpreter Mode
--------------------------------------------------------------------
VDSL DSP Firmware Version: 1.60.00-A1
VDSL Line State:      DATA   Total Transmit Power:    14.0 dB
DS Payload Rate:   63968kbps   Local Attenuation:    0.0 dB
US Payload Rate:   35968kbps   Local SNR Margin:   76.0 dB
```

Figure 90: VDSL 2 modem main menu

8.4 Quality measurement of Triple Play services

Measurement of service quality on real routes was done with both hardware and software tools. In the case of IPTV, the MSU Video Quality Measurement Tool and EXFO AXS-200/635 were used. The IxChariot software tool from Ixia was used to measure the quality of voice services. Data service testing was performed using download and sharing rate software.

The Abacus server on which virtual machines are located was used for the measurement. These were used for IPTV, VoIP and Data services. Network topology is shown in figure 91.

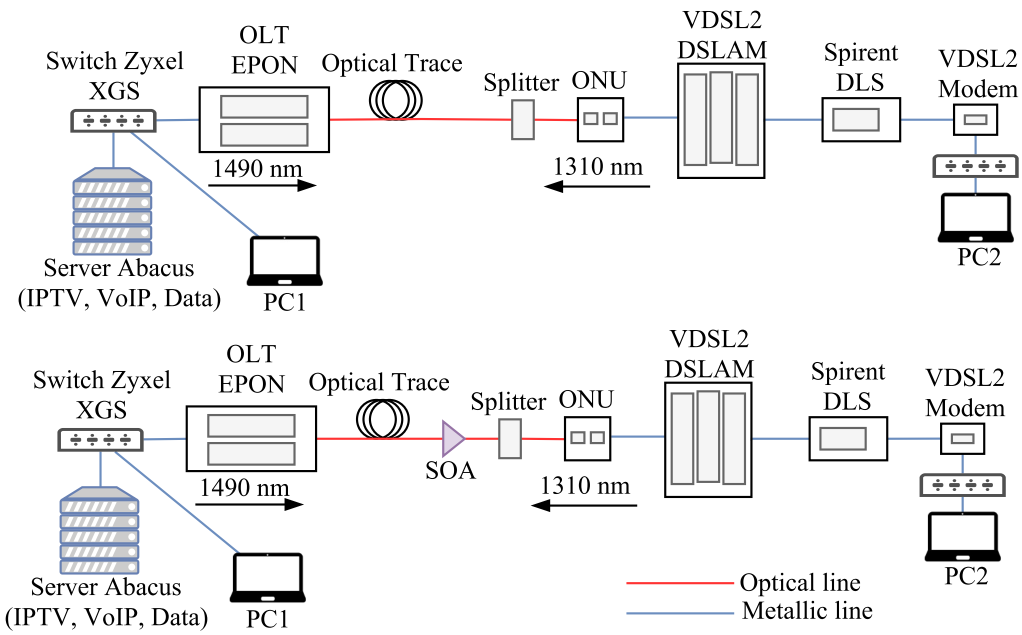

Figure 91: Network topologies with and without SOA for Triple Play services

8.4.1 IPTV

Testing was performed using the MSQ VQMT software. The operation was also measured by the AXS-200/625, which allows multicast traffic testing. The 30 table lists the parameters of the video samples being examined.

Table 30: Samples of videos for IPTV testing

Sample	Format	Frame per second	Resolution	Codec
1080p	MPEG	29	1920x1080	MPEG-2
1080p	MKV	25	1920x1080	H264 (MPEG-4)
720p	AVI	25	1280x720	H264 (MPEG-4)
576p	MPEG	25	720x576	MPEG-2
DVB-T	MPEG	25	720x576	MPEG-2

The measured values also vary according to the current picture, and the same sequence was not found during the measurement. This means that sometimes the image was fast or slower. More directions are shown in the tables in the appendix.

8.4.1.1 Measure IPTV using the MSU Video Quality Measurement Tool

IPTV testing was done using the MSU Video Quality Measurement Tool. The same video samples as in the previous measurement were tested. Testing is ongoing uploading streaming video and then comparing it to the original video. The only problem is that videos must have the same start so that the same frame can be compared.

Two-minute videos were recorded for testing and then compared to the original video. In this test, video from terrestrial broadcasts could not be tested because it was not comparable to the original video. Samples were tested using objective PSNR, MSE, and SSIM methods.

Testing took place on metallic trace 0 km, 0.6 km and 1 km, and on optical paths 0 km, 10 km and 20 km. For 10 km and 20 km SOA was involved. You can see the MSU Video Quality Measurement Tool on the figure 92, where the GUI is on the left and the full-frame scanning is on the right.

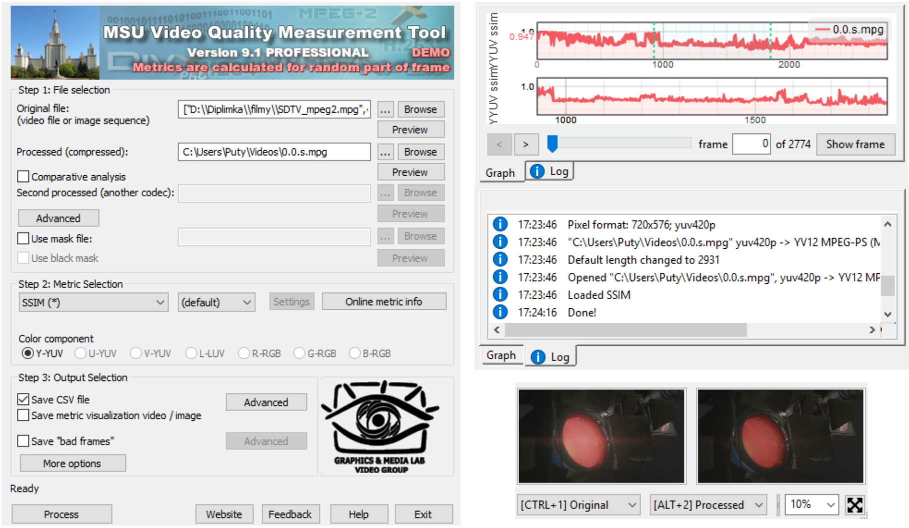

Figure 92: Practical show MSU Video Quality Measurement Tool

The objective MSE method expresses the mean quadratic deviation of the original signal from the captured signal.

In the figure 93 the resulting MSE values are displayed. The highest MSE value was in the MPEG-4 sample (1080p). The value was around 700. The smallest value for MPEG-2 (576p). MPEG-2 codec is less bandwidth and the MPEG-4 encoder is more demanding.

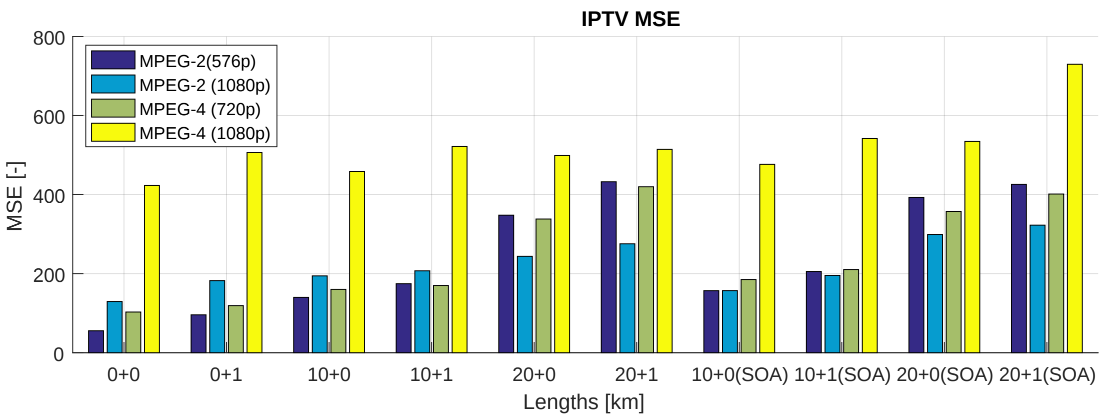

Figure 93: IPTV Values of the MSE parameter

The PSNR parameter represents the ratio between the highest signal value and the MSE parameter. It is given in dB. The higher the value of MSE, the sample is more different from the original video. For the PSNR parameter, the sample is the same as the original video, this is 100 dB. The lower the PSNR, the less tested the samples are.

In the figure 94, the resulting PSNR values are displayed. The PSNR parameter was rated at around 15 dB for MPEG-2 (576p) and MPEG-4 (1080p). A codec MPEG-2 (1080p) was moving around 30 dB.

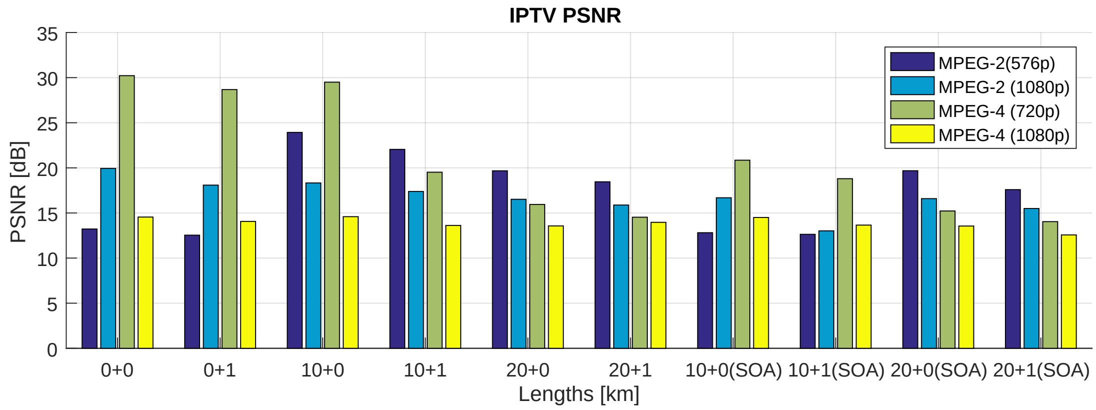

Figure 94: IPTV Values of the PSNR parameter

The SSIM parameter takes into account the human visual system. Value 1 means the match of the samples being compared. The value 0 indicates the zero relation to the original image.

In the figure 95, the resulting SSIM values are displayed. The SSIM parameter ranged around 0.9 for all codecs.

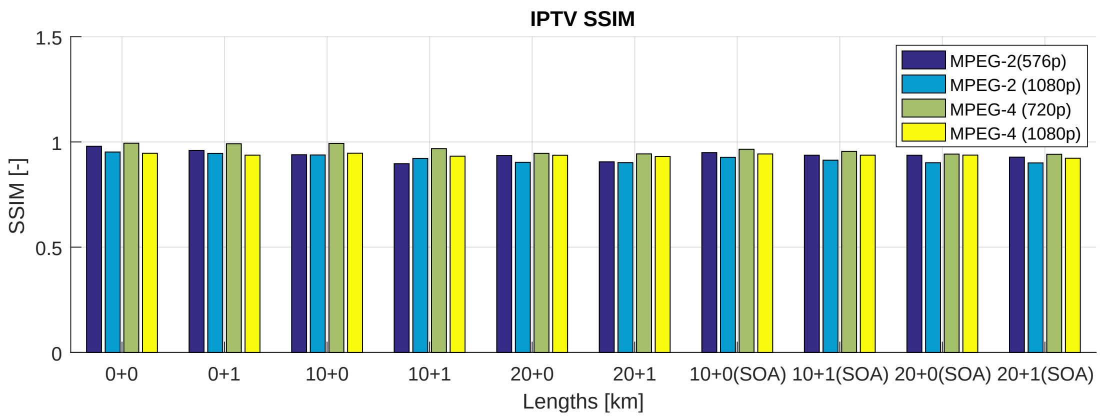

Figure 95: IPTV Values of the SSIM parameter

The change of the optical path from 0 km to 10 km but only 20 km was not very significant. The metallic path had a greater impact on change. The MPEG-2 576p, MPEG-2 1080p, and MPEG-4 1080p samples were the most significant change in the metallic path. All spreadsheet values are stored in the appendix. According to the results, it is better to use MPEG-4 for video streaming.

8.4.1.2 Measure IPTV using EXFO AXS 200/625

In this measurement, the video samples listed in the table 30 were analyzed. Video transmission speeds, packet loss, and jitter parameters were measured. Video samples were measured for two minutes, and the resulting data is analyzed below. The display of the test progress is shown on figure 96.

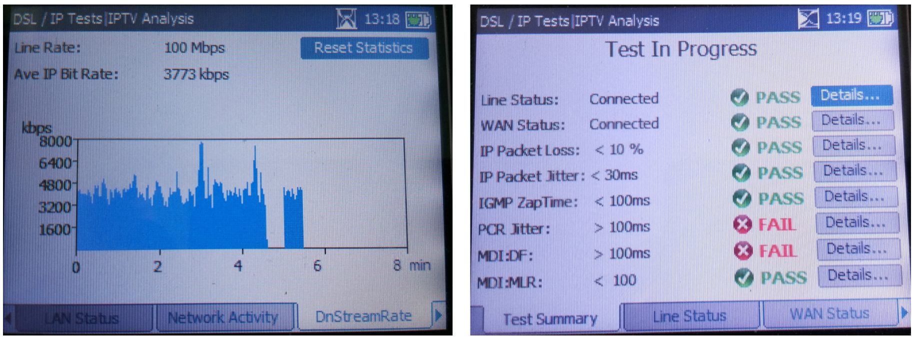

Figure 96: Test run from EXFO AXS 200/625

In the table 31, the average video speeds with different combinations of optical paths lengths (0 km, 10 km and 20 km) and metallic paths (0 km and 1 km) are shown. In the case of a 1080p MPEG-2 video sample that achieved high bit rates, then DVB-T, MPEG-2 (576p) and MPEG-4.

Table 31: IPTV Average IP Bit Rate of selected videos for VDSL2

	Average IP Bit Rate [kbit/s]				
Length [km]	**MPEG-2 (576p)**	**MPEG-2 (1080p)**	**MPEG-4 (720p)**	**MPEG-4 (1080p)**	**DVB-T (576p)**
0 + 0	4172	12902	3299	2604	4351
0 + 1	2824	7367	1053	2004	3548
10 + 0	3855	13700	1524	2553	4051
10 + 1	3300	8610	1416	2226	3404
20 + 0	3540	14489	1250	2309	3846
20 + 1	3500	8693	1200	2105	3654
10 + 0 (SOA)	4005	12838	2152	2915	4156
10 + 1 (SOA)	3453	8164	1651	2668	3545
20 + 0 (SOA)	3969	13927	1336	3211	4054
20 + 1 (SOA)	3110	8539	910	2574	3456

Figure 97 shows a comparison of codecs in a 0 km to 1 km metallic path and 0 km, 10 km and 20 km optical paths. The highest Bit Rate is for MPEG-2 codec (1080p) and the smallest MPEG-4 (720p).

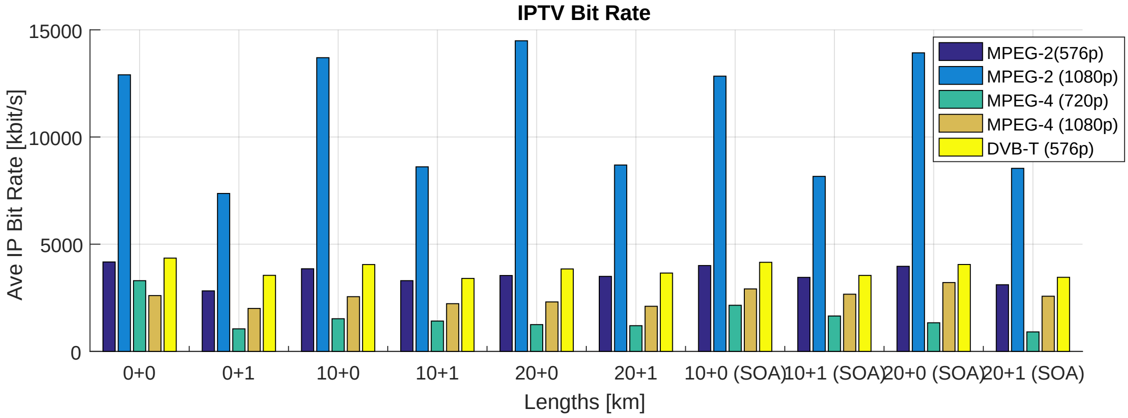

Figure 97: IP bit rate comparison for individual codecs

Comparison of two video sequences of a metallic path 0 km and 1 km. The left picture shows a 0 km optical path and 0 km of a metallic path on the right side of the same 1 km of metallic path.

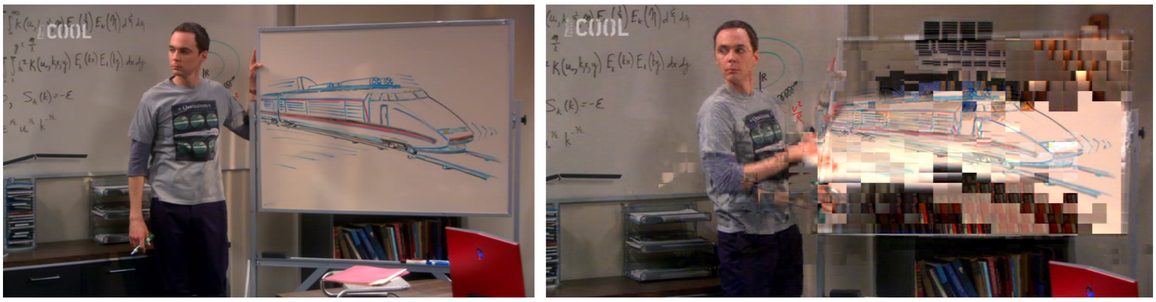

Figure 98: Video sequence comparison for MPEG-2 (576p)

Table 32 shows IP Packet Loss. For a MPEG-2 576p video sample has a low packet loss rate of up to 10 km of optical path and video viewing had no effect. Increasing the optical path to 20 km increased packet loss above 10%. It had a negative impact on the video sample.

For the MPEG-4 1080p video sample, packet loss was almost zero in 1 km of metallic path. This is because the MPEG-2 1080p needs a minimum speed of around 34 Mbit/s.

For the MPEG-4 720p video sample, changing the distance of the metallic path did not have a major effect on packet loss.

For the MPEG-4 1080p video sample, the metallic path had an effect of up to 1 km where the packet loss was measured at 0.43% and had a lesser negative effect.

At other lengths of the metallic path, packet loss was minimal. On the optical path, the loss occurred within about 20 km and approximately 10%.

Table 32: IPTV IP Packet Loss of selected videos for VDSL2

Length [km]	**IP Packet Loss [%]**				
	MPEG-2 (576p)	**MPEG-2 (1080p)**	**MPEG-4 (720p)**	**MPEG-4 (1080p)**	**DVB-T (576p)**
0 + 0	0	1.61	0	0	0
0 + 1	0.78	1.97	0	0.43	0.9
10 + 0	0.43	1.27	0	0	0
10 + 1	0.67	1.98	0	0.45	0.8
20 + 0	13.21	14.29	13.89	11.81	12.15
20 + 1	16.58	15.46	15.28	17	15.4
10 + 0 (SOA)	0.3	1.63	0	0.42	0.2
10 + 1 (SOA)	0.64	2.76	0	0.85	0.9
20 + 0 (SOA)	10.93	14.49	12.05	11.22	9.54
20 + 1 (SOA)	14.29	15.5	13.11	15.81	12.48

Figure 99 shows IP Packet Los, where the largest was the 20 km optical path for all codecs.

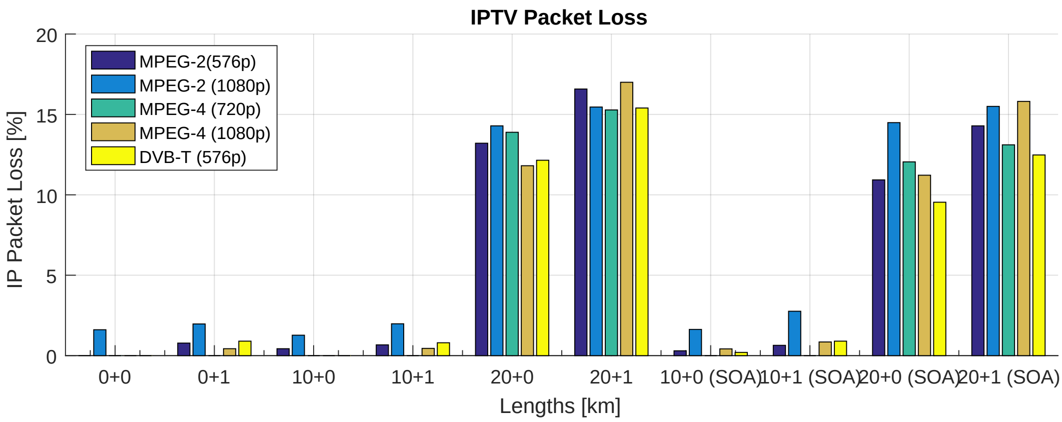

Figure 99: IP bit rate comparison for individual codecs

Figure 100 shows the difference between specific sample images MPEG-4 videos (1080p). The first image is from a video sample at an optical path of 0 km and a 0 km metallic path and the second is from a 0 km optical path and 1 km of a metallic path.

Figure 100: Video sequence comparison for MPEG-4 (1080p)

Table 33 shows Arrival Jitter with different combinations of optical path lengths (0 km, 10 km and 20 km) and metallic paths (0 km and 1 km). The values vary according to the image sequence. The largest jitter had an MPEG-4 codec (1080p) at 20 km optical path. Smallest for MPEG-4 (1080p).

Table 33: IPTV IP Arrival Jitter of selected videos for VDSL2

	IP Arrival Jitter [ms]				
Length [km]	**MPEG-2 (576p)**	**MPEG-2 (1080p)**	**MPEG-4 (720p)**	**MPEG-4 (1080p)**	**DVB-T (576p)**
0 + 0	3.47	1.29	4.73	0.77	3.25
0 + 1	5.87	1.7	13.34	5.66	5.45
10 + 0	4.03	1.17	6.12	4.88	3.98
10 + 1	4.34	1.88	10.27	5.97	4.05
20 + 0	3.03	0.97	7.2	5.71	2.89
20 + 1	4.72	1.36	17.25	6.91	4.54
10 + 0 (SOA)	3.94	1.34	7.08	4.5	3.45
10 + 1 (SOA)	4.8	1.68	9.35	4.77	4.5
20 + 0 (SOA)	3.6	1.14	11.31	5.67	3.2
20 + 1 (SOA)	4.79	1.35	15.07	5.91	4.6

Figure 101 shows the IP Arrival Jitter, representing the table 33.

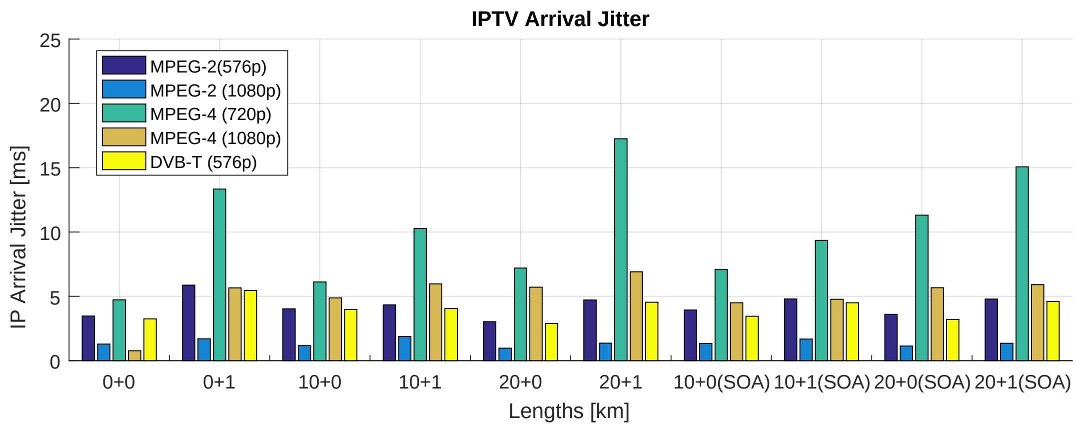

Figure 101: IP bit rate comparison for individual codecs

8.4.2 VoIP

To evaluate the quality of voice services, IxChariot was used, which was installed on the virtual machine via VirtualBox. For testing, 2 endpoints must be set. For each endpoint, ie PC1 and PC2, IXCHariot Platform ENDPoints must be installed, which is freely available. One computer was server-side and the other was connected to the modem side. In the topology they are marked as PC1 and PC2..

An overview of the codecs used is shown in the table 34. I tested codecs that use a different bandwidth. Other codecs are similar to those selected. The measurement results are shown below.

Table 34: Used codecs for testing

Codec	Bandwidth [kbit/s]
G.711 $\mu - law$	64
G.723.1-ACELP	5.3
G.729	8

The demonstration of individual profile settings is shown in the figure 102.

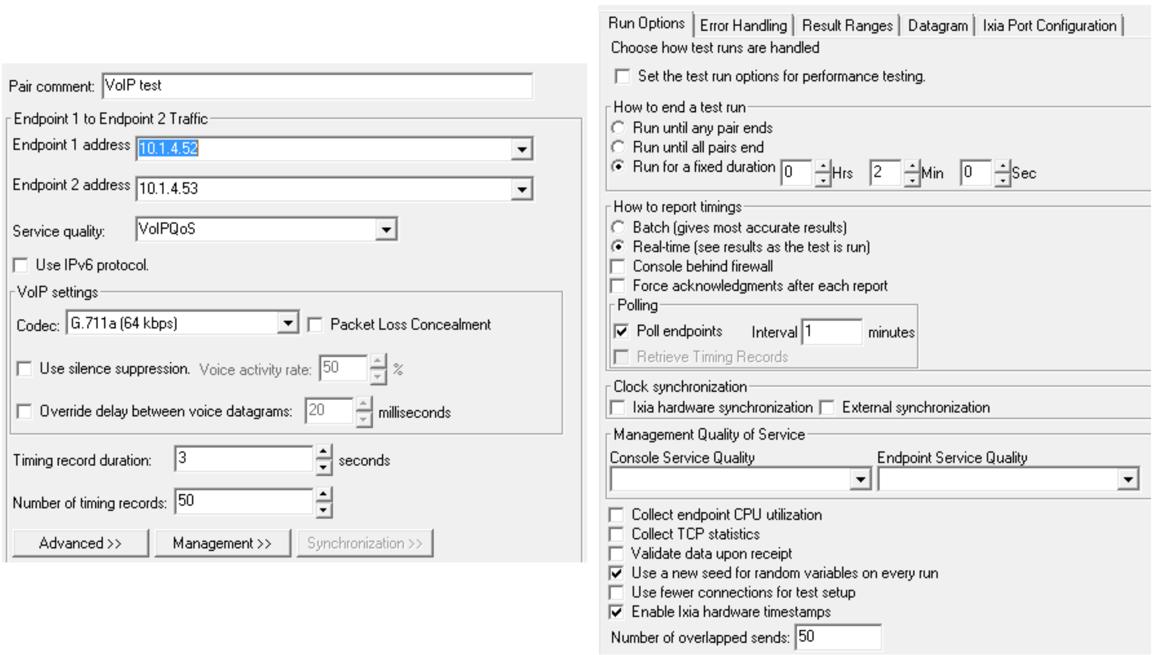

Figure 102: VoIP Run configuration

8.4.2.1 Measurement by IxChariot

Measurements were made on a combination of the lengths of the optical and metallic paths. The bandwidth of the individual codecs is so small that the results are the same and the path change does not affect the quality of the call. Recorded call quality parameters were MOS and R-factor.

The progress of one of the tests is shown in the picture 103.

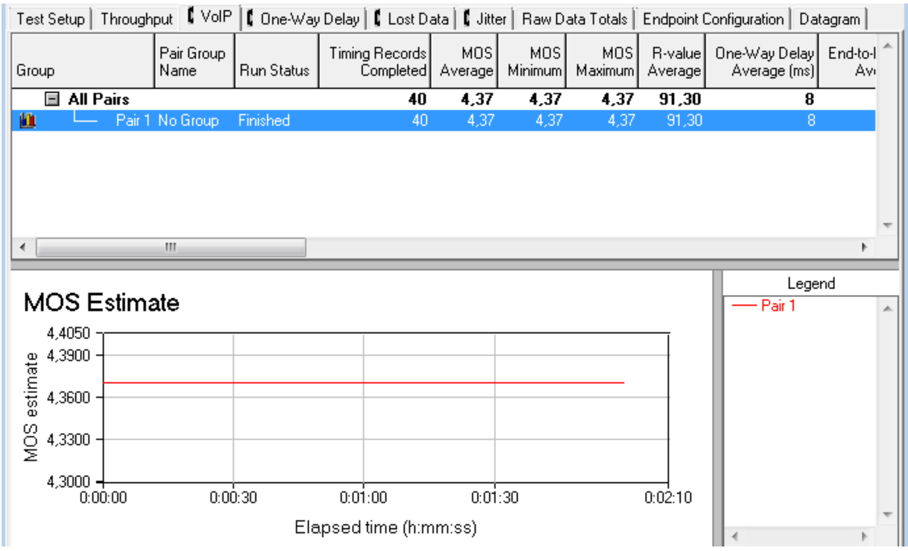

Figure 103: VoIP show complete one test

The table 35 shows the measured MOS and R-factor values for the VDSL2 profile. Changing path lengths has had a significant impact on voice services.

Table 35: Measured MOS and R-factor values for VDSL2 profile

Codec type	G.711 μ-law		G.723.1-Acelp		G.729	
Combination of Lengths	**MOS**	**R-Factor**	**MOS**	**R-Factor**	**MOS**	**R-Factor**
0 km + 0 km	4.37	91.28	3.64	70.93	4.02	80
0 km + 1 km	4.28	88.88	3.57	68.08	3.94	77.89
10 km + 0 km	4.37	91.23	3.64	70.92	4.02	80
10 km + 1 km	4.27	88.41	3.57	69.02	3.94	77.76
20 km + 0 km	4.28	88.9	3.64	70.92	4.02	80
20 km + 1 km	4.27	88.39	3.57	69.06	3.94	77.87
SOA 10 km + 1 km	4.29	88.94	3.57	69.11	3.94	77.92
SOA 20 km + 1 km	4.37	88.95	3.57	69.12	3.95	77.94

The reason why the quality of the call is unaffected, even if the EtherSAM test indicates exactly the opposite, is that when testing ITU Y.1566, performance tests have been evaluated when all Triple Play services are running at one time. When running IxChariot on line, one call is running, which requires minimum throughput.

The quality of the call in terms of user satisfaction is very satisfying for the G.711 codec, with the G.729 codec being around 80, meaning the Satisfied level. R-factor values for codec G.723.1 are above 70, in which case some users may be dissatisfied. All these values, however, are typical for the codecs.

8.4.3 Data

Measurement of data service proceeded on identical topology as measurement of previous services. The virtual machine for Data was turned on, on Abacus. Here a static address was set, and after the address was entered into the client-side search engine, a page was downloaded from which the files could be downloaded or the files could be uploaded.

Speeds were measured using the BWMeter software, which is designed to measure download and upload speeds. The resulting values were recorded in the table 36.

Low download speeds were at first, but after downloading multiple files from the server, download speed has stabilized and showed more acceptable results. There were no defects in the upstream, but they showed relatively small values against the real ones. Figure 104 shows the BWmeter with which the entire measurement was performed. It can record transmission speeds in both directions.

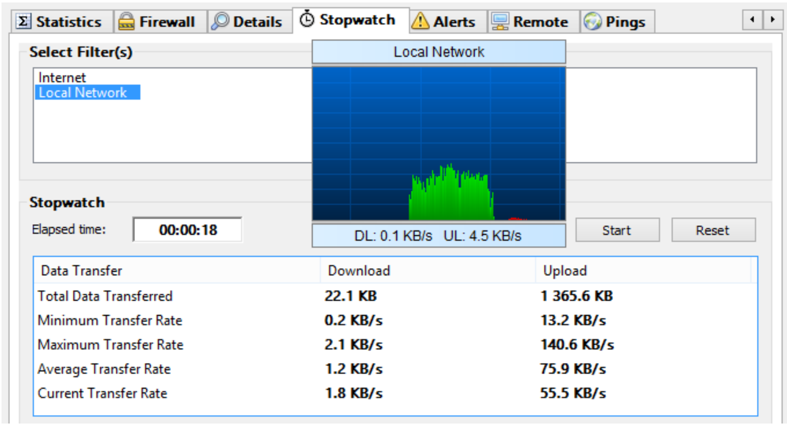

Figure 104: BWmeter in process with Local Network graph

Table 36 shows the measured speeds for the VDSL2 profile. Here is a great drop in speed when uploading and downloading a metallic path. After adding 1 km of metallic path, the speed dropped from 23.3 Mbit/s to 5.9 Mbit/s in Download. Changing the optical path from 0 km to 20 km did not have a significant effect on the resulting values. Similarly, this is true for SOA.

Table 36: Measured data transmission speeds for VDSL2

Length [km]	Download [Mbit/s]	Upload [Mbit/s]	Length [km]	Download [Mbit/s]	Upload [Mbit/s]
0 + 0	23.348	5.941	**-**	-	-
0 + 0.6	11.383	0.983	**-**	-	-
0 + 1	5.999	0.315	**-**	-	-
10 + 0	21.148	4.565	**10 + 0 (SOA)**	21.804	4.368
10 + 0.6	10.481	0.966	**10 + 0.6 (SOA)**	12.250	1.051
10 + 1	4.461	0.373	**10 + 1 (SOA)**	4.594	0.139
20 + 0	20.302	3.833	**20 + 0 (SOA)**	20.476	3.402
20 + 0.6	9.053	0.922	**20 + 0.6 (SOA)**	13.278	0.990
20 + 1	4.006	0.158	**20 + 1 (SOA)**	3.943	0.105

Graphs 105 and 105 show the results of boxplot measurements. For the sake of clarity, I display only the maximum and minimum measured results for 0, 10 and 20 km. With SOA then 10 km and 20 km. On the metallic path 0 km and 1 km.

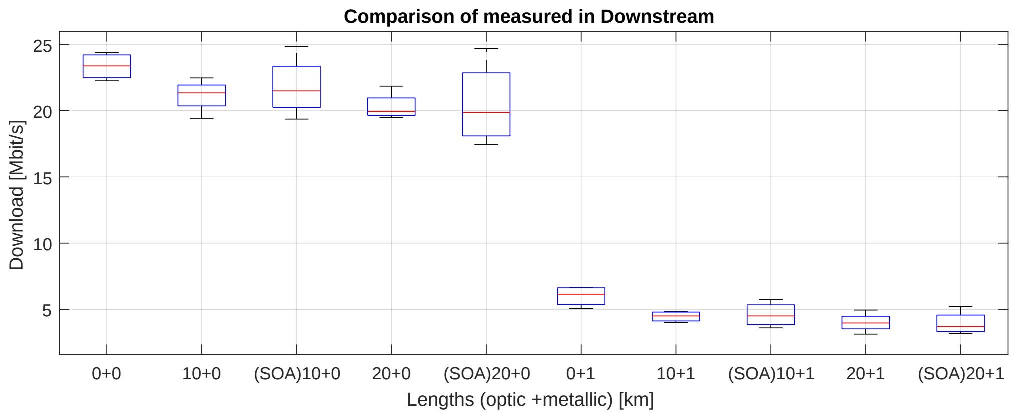

Figure 105: Boxplot for Data Quality measurement in Downstream

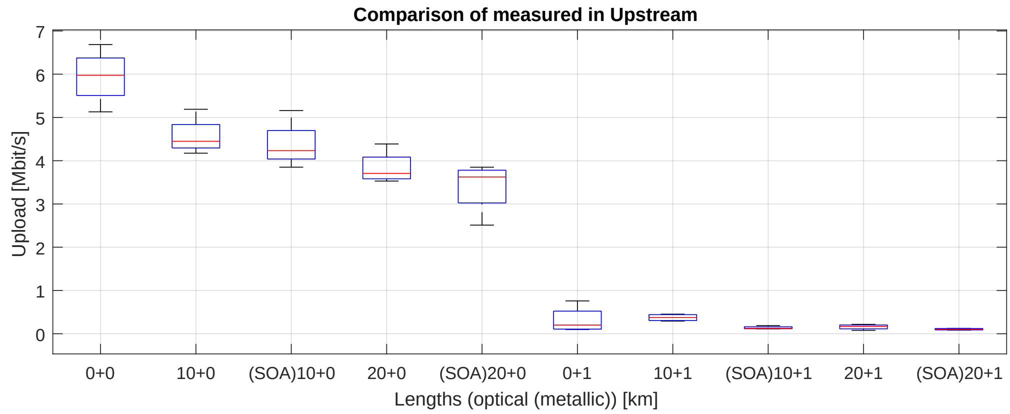

Figure 106: Boxplot for Data Quality measurement in Upstream

9 Simulation of EPON network in Optiwave Software

This chapter deals with optic network design in Optiwave OptiSystem software, which has a large number of optical network design components. For testing, the topology of the optical network was designed to match the topology of the optical part of the network in real-time measurements.

OptiSystem offers a system for the design and planning of an optical communication system at the level individual components and is then able to offer visual analyzes and scenarios. His the ability to integrate with other Optiwave products makes it unique to it in the field of optical communication systems.

Figure 107 shows the topology of the network, which consists of individual blocks, which are described later in the chapter.

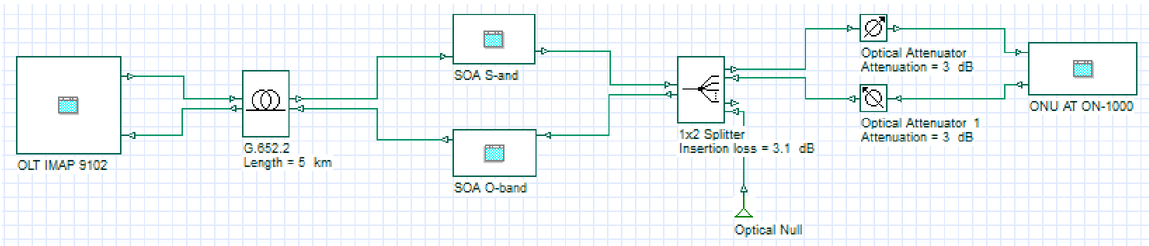

Figure 107: Optical network topology

The Hybrid Access Network is based on a real network that has been measured. It consists of an OLT unit, one optical path section, one 1:2 optical splitter, one AT-ON 1000 terminal block, two optical attentuators, and a set of 2 amplifiers located between 4 band splitters. For the proper functionality of the connection, multiple iterations according to the optical dealy plus one must be set globally.

9.1 Optical Line Termination (OLT)

The layout of the components in the OLT is shown in the figure 108. The transmission power to the access network is composed of a CW laser at a wavelength of 1490 nm and the output power is set at 3.7 dBm. Data generation is made up of the Pseudo-Random Bit sequence and Bit rate is set to 1250 Mbit/s. Random data is encoded into 8B/10B sequences. The sequence generator is connected to the NRZ pulse generator. The optical signal is modulated by the Mach-Zehnder NRZ pulse generator. Both the receiving and transmitting parts are connected by a circulator that connects these different states.

The OLT receiving part is made of InGaAs (Indium, Gallium, Arsenide) Photodiodes. The electrical signal is processed by the Low Pass Bessel Filter, which is set as the low pass filter for high-frequency noise filtering. The 3R Regenerator restores the shape and timing of the signal.

The 3R Regenerator is connected to the BER Analyzer, which shows the status and properties of the signal after browsing the topology.

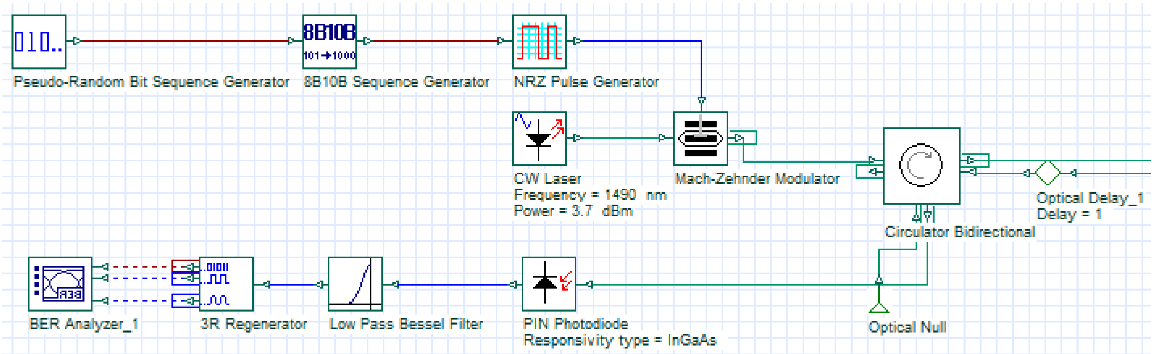

Figure 108: OLT in simulation environment OptiSystem

9.2 Optical Network Unit (ONU)

The ONU consists of a block in which the detection part and the receiving part are inside. The ONU consists of the receiving side of the PIN photodiode, the Low Pass Bessel filter 3R regenerator and the BER analyzer similar to OLT. On the transmitting side, the Pseudo-Random Bit Sequency Generator is set to 1250 Mbit/s. 8B/10B Sequence Generator and NRZ Pulse generator. The CW laser is set at a wavelength of 1310 nm and a power of 1 dBm. The figure 109 shows the ONU schema.

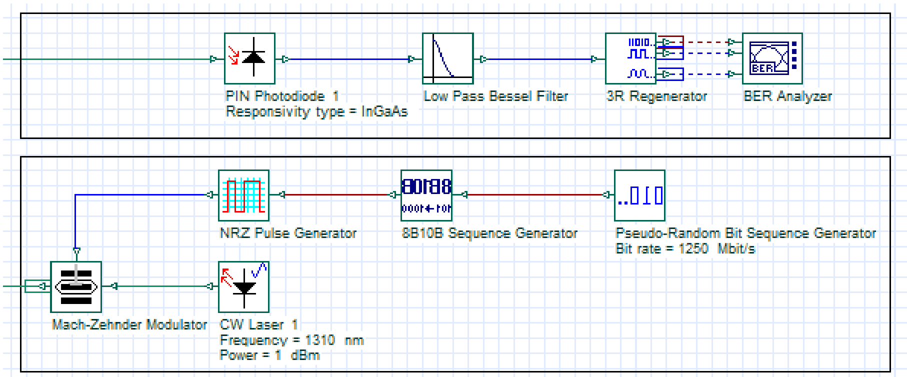

Figure 109: ONU in simulation environment OptiSystem

9.3 Optical Distribution Network (ODN)

ODN consists of an optical path that is placed in front of a pair of WDM multiplexes. WDM multiplex is connected to SOAs that terminate the WDM demultiplex. For SOA, the optical splitter is 1:2 and the attentuator. The entire ODN scheme is shown in the figure 69 and the SOA Detail in the figure 111.

G.652.1 is used as optical bi-directional fiber. The attenuation coefficient is set to 0.6 dB/km. the PMD is set to $ps/km^{1/2}$. The values are averaged and based on real measurements. Measurements took place for 5 to 35 km in 5 km increments. The SOA is located between the optical path and the splitter, where the gain at 1490 nm is set to 20 dB and 130 nm to 20 dB, according to real measurements. Optical delay components are designed for simulation purposes only for proper bidirectional operation.

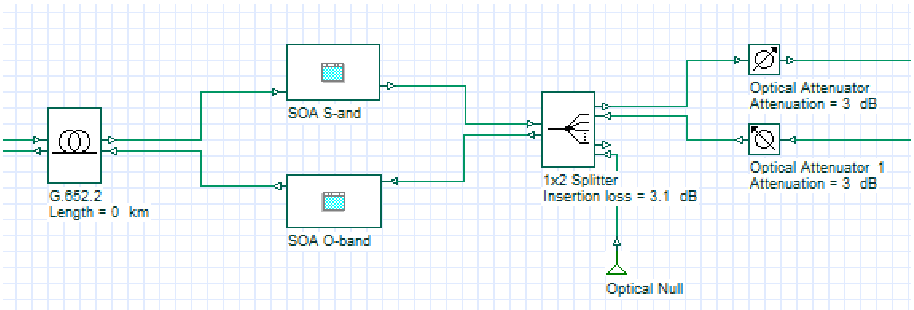

Figure 110: ODN in simulation environment OptiSystem

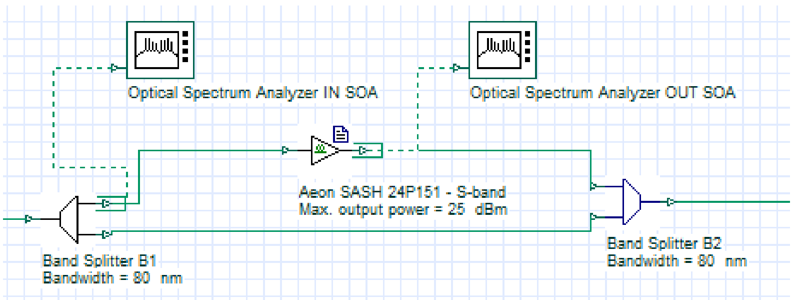

Figure 111: Detail on SOA in ODN

9.4 Measurements in simulation environment

Measurement of the network in the simulation environment is based on the same principle of measuring the real network in the laboratory by changing the length of the optical path. In OptiSystem, eight iterations were set in sweep mode, with each iteration being assigned the optical path length.

In the simulation environment it is possible to evaluate the quality parameters of the communication channels only based on the BER analyzer. The limit value of $1 \cdot 10^{-9}$ has been selected as the critical value of the BER. This value determines whether the channel is still capable of communicating. BER, eye diagram and optical spectrum were recorded for each simulation. Modulation of NRZ is used for modulation of the signal.

Figure 112 represents a graphical representation of bit error rates on a logarithmic scale, depending on the length of the optical path at a combination of 0-35 km in 5 km increments.

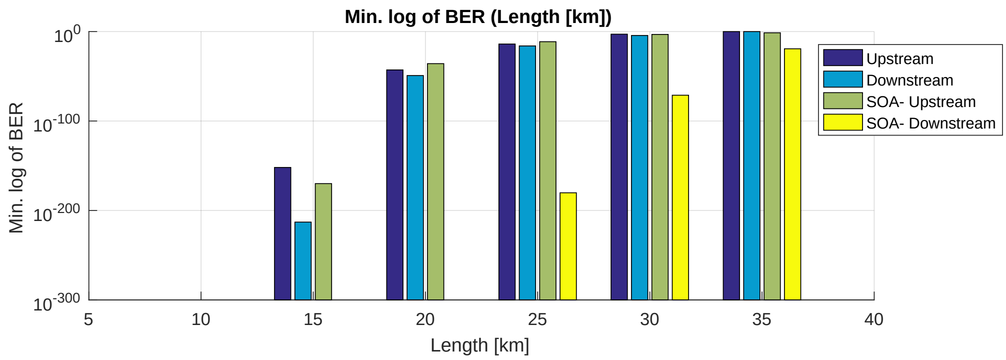

Figure 112: BER values in logarithmic terms depending on the length of the optical trace

It can be seen from the chart that from 25 km or more the boundary is close to the limit value. We can see that SOAs are very effective in Downstream, where the error rate starts up to 25 km. Without SOA, the error rate starts at 15 km. Upstream is worse off, and the error rate can be recorded for 15 km.

9.4.1 Eye diagrams

The following pictures are the result eye diagrams for 5, 30 and 35 km for each direction. We can see that the openness of the eye is maximum in the case of 5 km of the downstream and upstream path. Increasing the optical path over 30 km is the minimum openness of the eye in Upstream in both directions. Over 30 km was not connected without SOA, so the distance is only 30 km without SOA with SOA within 35 km.

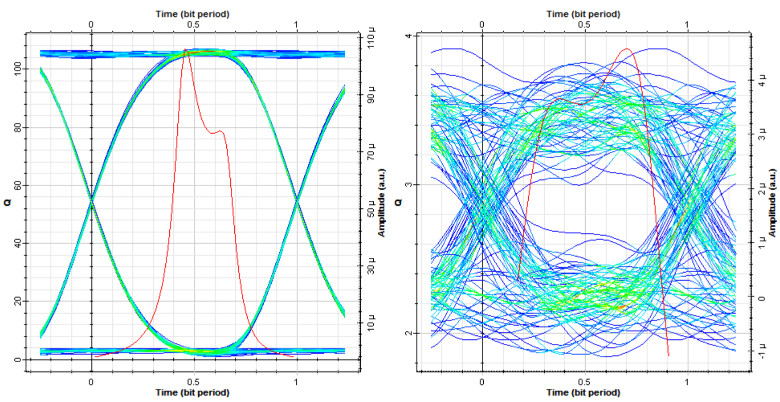

Figure 113: Eye diagram for Downstream on trace 5 and 30 km, Without SOA

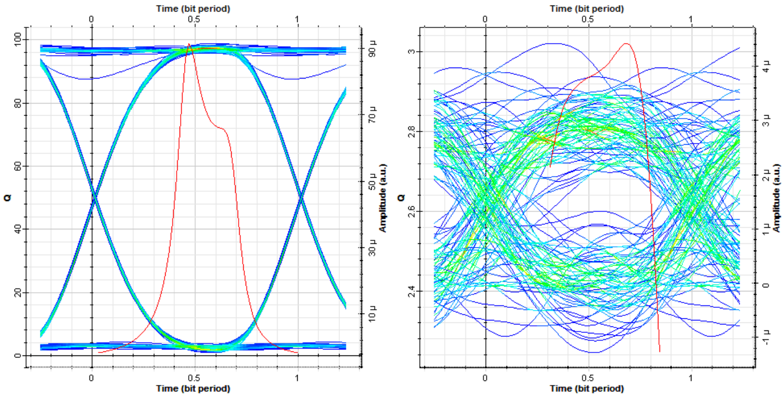

Figure 114: Eye diagram for Upstream on trace 5 and 30 km, Without SOA

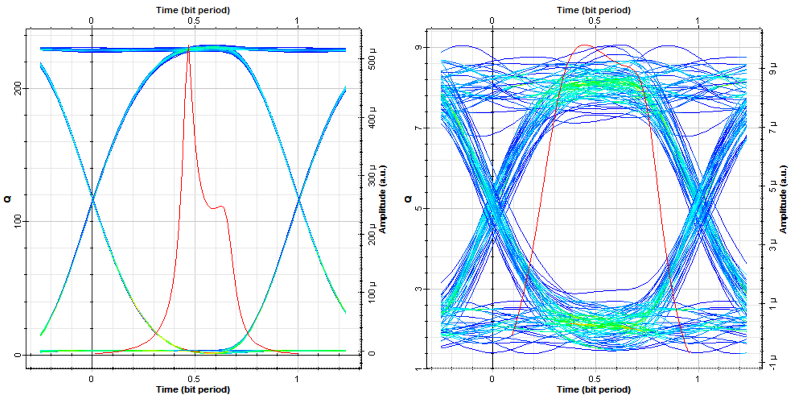

Figure 115: Eye diagram for Downstream on trace 5 and 35 km, With SOA

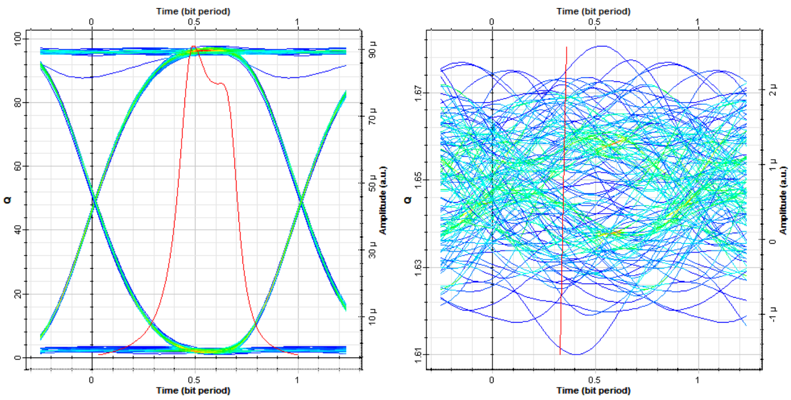

Figure 116: Eye diagram for Downstream on trace 5 and 35 km, With SOA

9.4.2 Spectral characteristics

Figures 117 and 118 show the spectral characteristics of the access network implemented in the Optiwave simulation environment. They show the input and output signals to the field of amplifiers for downstream on the optical path of 10 km. Here you can see the ASE noise represented by the green color that was generated by the SOA amplifier when amplifying the input signal. The optical spectrum at other distances is similar. On the show, I put the spectrum before SOA and SOA.

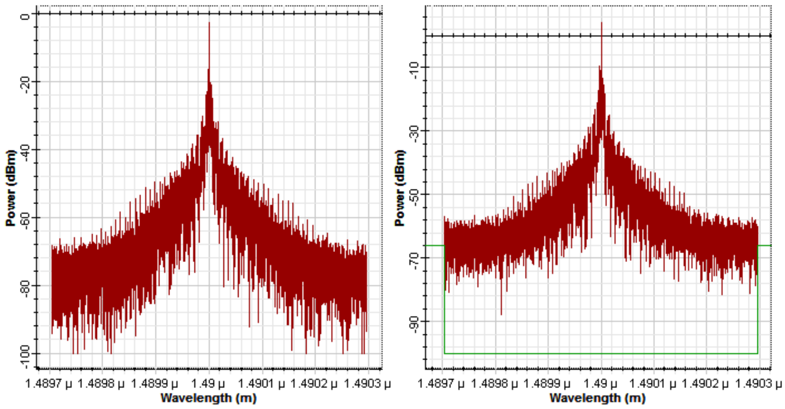

Figure 117: Optical spectrum on 10 km, Without and With SOA, Downstream

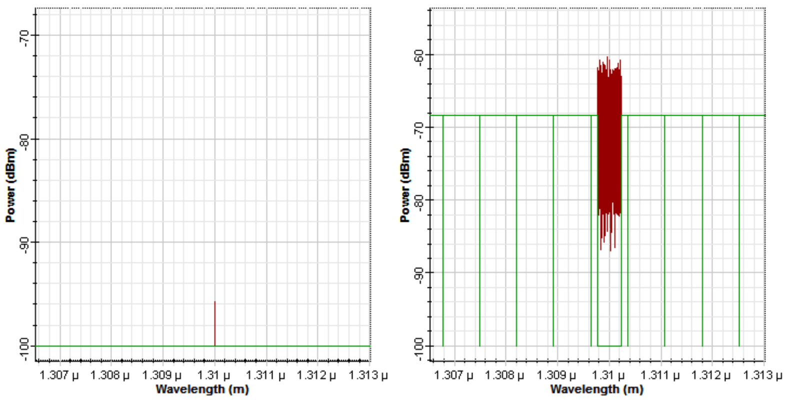

Figure 118: Optical spectrum on 10 km, Without and With SOA, Upstream

9.4.3 Summary Measurements

From the results in which I tried to set the whole topology according to real measurements, it can be seen that the optical transmission between OLT and ONU is similar to real measurement. The values with SOA are significantly better, as can be seen in the chart above. The amplification was performed on the S-band and the O-band at the same time, and due to the gain, the distance was increased. I measured the values from 0 km to 35 km with a 5 km pitch.

The simulation permits ideal conditions, even if the data is set to approximately real values, so the actual experimental values do not match the measured ones. In order to prevent these deficiencies, it is necessary to adjust the sensitivity of the used photodetectors, each connector and others.

9.5 ADSL ITU-T G.992.1

Optisystem is designed for fiber optics, so I will describe at least the individual blocks of the digital part of the ADSL modem.

A summary of ADSL is presented in Figure 119. In the first block the signals are put together in frames. These frames are transmitted over two different paths where the bit order is changed and coded.

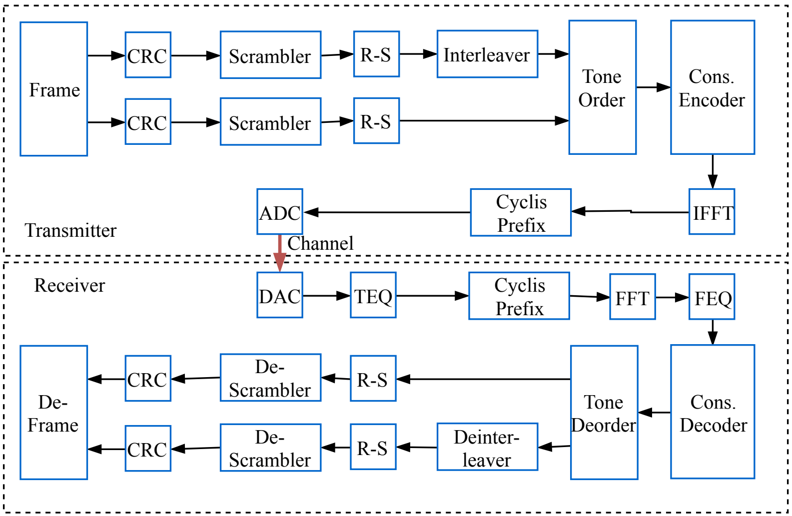

Figure 119: The ADSL block structure. All major parts such as framing, encoding, distributing in different frequencies, filters and decoding are described in their respective block.

The next main block is where the bits are divided into frequencies different. The modulation technique that is used in ADSL is called Discrete Multitone (DMT). The method divides the copper line into smaller frequency bands that are used independently.

ADSL can transmit data in both directions at the same time and on the same frequencies. Handle the overlapping frequencies an echo canceller is used to cancel the signal that is transmitted in the wrong direction. It is possible to transmit the data on different frequencies [100].

Framer

The first block is the framing block. In this block, a lot of different incoming data channels are put together. The different incoming data channels can be used one and one or many together. The framer puts the bits in frames and the frames in superframes.

Each frame of data corresponds to one ADSL symbol. The frame is divided in a fast path and an interleaved path. In the framer the cyclic redundancy check (CRC) is appended. CRC is a control of that the transmission is ok.

Cyclic redundancy check (CRC)

Cyclic redundancy check (CRC) is a short sequence that is evaluated for each superframe. Two CRC results are transmitted. One from each of the fast and the interleaved path. In ADSL the CRC-8 with the polynomial $G(x) = 1 + x^2 + x^3 + x^4 + x^8$ is used. That means that eight bits are appended to each superframe. These bits are in the receiver used to decide if the transmission is correct. CRC can only detect errors and it can accordingly not correct errors.

Scrambler

The scrambler in ADSL is applied to a serial data stream. Frames and synchronization are not important for this function to scramble and also not for descramble.

The idea with scrambling is to mix up the bits to avoid very long bursts of ones or zeros. A better mix gives a better performance. Another reason to have scrambling is that it ensures that the spectrum from the transmitted bits are more like noise.

Reed-Solomon coding

The idea this block is to append redundancy to the transmitted bits that can be used to correct bit errors and detect that can occur during the transmission.

Interleaver

The idea with the interleaver is to spread bursts of errors that can appear to different many code words. The interleaver has two important parameters, the interleave depth, D and the number of bytes per codeword, N. D is always a power of two and is decided during the initialization.

Tone Order

The tone ordered assigns bits to different carriers. The most important thing to know is how many bits that can be assigned to each symbol. That information is stored in the bit table. The bit-table is calculated in the initialization process and for each symbol the tone order gets so many bits from the previous block.

Constellation encoding

The constellation encoding is a process that is independent for each tone. The number of points for each tone depends on the number of bits that are assigned to each tone. The smallest tones with two bits can give four different constellation points and the biggest with 15 bits can give 32768 different points.

FFT

The QAM-constellations are mapped on carriers using the inverse discrete Fourier transform (IDFT). This transform and the symmetry in the constellation make the result real-valued. The transformation to real values makes it possible to convert the digital signal to an analog signal in a later step. In the receiver a discrete fourier transform (DFT) is used for demodulating the signal.

Cyclic prefix

A cyclic prefix is appended to the fourier transformed symbol. The idea with this block is to make the equalization procedure easier and make it possible to destroy most of the components from the ISI.

DAC

A digital to analog converter (DAC) converts the signal from digital values to the analog world.

Channel

Between the two modems in ADSL is the channel which in this case is the telephone line. It must be pretty short because the performance decreases with the length to the telephone station. A telephone line can have a lot of different forms. It can be one very long cable or it can be a cable with a lot of crossings that disturb the signal.

ADC

An analog to digital converter (ADC) converts the signal from a time- and amplitude continuous signal to a time and amplitude discrete signal.

Time Domain Equalizer (TEQ)

The main idea with the TEQ filter is to mitigate the intersymbol interference (ISI) that appears between two different symbols. ISI appears because two symbols overlap each other. The main idea with a TEQ filter is to push the ISI to a small range. If the range is shorter than the cyclic prefix all the problem can be removed.

Frequency Domain Equalizer (FEQ)

The Frequency Domain Equalizer is a vector with complex values that is multiplied row wise with the subchannels, if the cyclic prefix is sufficiently large and the TEQ removes the ISI completely. That also means that if the prefix length is longer than the length of the channel impulse response, then the FEQ is only one single complex coefficient for each subchannel.

10 Conclusion

The aim of the master's thesis was to create a heterostructural network made up of GEPON Allied Telesis iMAP 9102 and xDSL technologies. Subsequently, the analysis of the impact of changes in the lengths of optical and metallic paths on network integrity and the deployment of Triple Play services. Another aim of the thesis was the study of hybrid heterostructure networks.

In the theoretical part, they are first described optical access networks from the point of view of functional units. The next chapter is focused on PON networks, specifically PONs, which use time division multiplexes. These include xPON, collisions have been solved, and GPON has also mentioned its architecture. It was followed by EPON, a passive optical network using Ethernet on the second layer.

The theoretical chapter was dedicated to Triple Play services, mainly VoIP, IPTV and Data. VoIP and IPTV introduced the most commonly used codecs, as well as the protocols used by these applications. There are also methods for testing the quality of these services.

The practical part consisted of several parts. It is mentioned the interconnection of laboratories and the construction of an optical path that was necessary for the creation of an xPON/xDSL experimental workplace. There is a description of all the devices that were used in practical use, including configurations. Here is detailed instructions for setting OLT and all devices. I first measured the built-in hybrid network Tests then I measured the optical section and then the Triple Play service.

The next part was testing the integrity of the experimental network by measuring the RFC 2544 and ITU-T Y.1564 and RFC 6349 tests. Different combinations of path lengths and their effect on the parameters measured by these tests were explored.

ITU-T Standard Y.1564, the values of the measured parameters reach stable values irrespective of the peak power value of the given channel. Increased frames loss was caused boundary values where the link between OLT and the UN was before decay. As regards the EPON system, of all combinations of optical paths, several cases have been reported, when the ONU was in the UP and DOWN modes. Downstream EPON broadcasts a higher power level than Upstream at ONU. RFC 2544 requires a more stable connection than previous ITU-T Y.1654. Because he is older and his successor is ITU-T Y.1564. The RF 6349 Test tested TCP / IP throughput with good results across the hybrid network. Of the total testing, that if the line was stable throughout the test, the resulting values are applicable.

Exploration and measurement of the whole solution took place in several phases, first of all I measured the optical path quality using OTDR, PM and CD methods. Then the quality of the metallic path. Then I joined SOA with which I conducted tests on the whole topology. I used Optical spectral analyzer and PON Power Meter. Two SOA semiconductor optical amplifiers were placed in this integrated network. In the real application, it was necessary to provide suitable passive elements for the distribution of 2 communication bands to separate optical fibers. It happened to separate the individual transmission bands to independently branches.

Downstream is a large crosstalk at a wavelength of 1490 nm, which is the EPON2 channel on the band divider output. It was designed for a wavelength range of 1525-1640 nm. Based on the information provided, it has been found that the dividers are designed with low insulation boundaries.

Another point of the master thesis was the introduction and subsequent testing of multimedia services. IPTV traffic was analyzed using the EXFO AXS-200/625. Bitrate and packet loss indicator were also investigated here. Videos were tested using objective assessment methods. Specifically, it was MSE, PSNR and SSIM. The evaluation was done using the MSU Video Quality Measurement Tool. A big problem occurred when capturing video and recording. Each video sample needed to be uploaded and then compare it in the MSU Video Quality Measurement Tool. It was difficult to capture the same sequence from which they were evaluated MSE, PSNR, and SSIM parameters by athematical calculations. The final evaluation of the samples obtained was evaluated according to the standards.

VoIP was tested using a virtual machine with IxChariot software installed by Ixia, which examined effects on the quality of the MOS and R-factor between the two endpoints. The software in the virtual machine was very difficult to connect with the local network and find two endpoints represented as 2 PCs.

Downstream and upstream data rates were examined using BWMeter software. The data was analyzed from the web browser where I downloaded and uploaded the files. Using full network capacity, I downloaded multiple files at the same time, and I upload multiple files also. I averaged the results and put a boxplot in the chart.

The last practical part was to engage and implement a functional topology in the Optiwave simulation environment. This section records and compares values as in the practical measurement where the measurement with SOA and without SOA. Simulated parameters were used to measure optical paths and their chromatic and polarization dispersion values. The evaluation was performed using the eye diagram and the bit rate error at the optical path length of 5 to 35 km in 5 km increments. With increasing distance, bit error rate increased.

When evaluating the results on the network topology, it was found that changing the length of the optical path from 0 km to 10 km did not have a large impact on the measurement. Greater influence on some services had an optical length of 20 km, where it did not respond to the ONU. The measurement was more strongly influenced by changing the lengths of the metallic route. This is because when using the optical path, we are able to bridge larger distances at higher transfer rates than in the case of metallic conduction.

All the points of the diploma thesis have been fulfilled. There have been many problems during measurement and preparation. But with sufficient material preparation, everything has been accomplished. I think the master thesis on Triple Play Measurement Services in the Hybrid Network could be a useful aid in further processing of topics for students. They could make measurements for multiple path combinations when using other profiles for VDSL2. Change QoS parameters and analyze the impact of Triple Play Services on created hybrid network.

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A List of Appendices on DVD

Part of the masters thesis is DVD. Its directory structure is as follows:

- Chromatic Dispersion Measurements
- OTDR - optical traces
- PMD measurements
- Spectral Analysis
- Simulation - OptiSystem
- EtherSAM results
- RFC6349 results
- RFC2544 results

B Tables

In this chapter only table values will be listed.

Measurement results using IPTV MSU

Tables 37, 38 and 39 shows the results of the MSE, PSNR, and SSIM from IPTV.

Table 37: The values of the MSE parameter

	MSE			
Length [km]	**MPEG-2 (576p)**	**MPEG-2 (1080p)**	**MPEG-4 (720p)**	**MPEG-4 (1080p)**
0 + 0	55.49132	129.70587	102.89239	423.0218
0 + 1	95.69874	182.3387	119.20301	506.1221
10 + 0	140.21828	194.27431	160.45456	458.1191
10 + 1	174.38591	207.03578	170.25555	521.5297
20 + 0	348.1604	244.13107	338.28287	498.6921
20 + 1	432.39227	275.37262	419.73151	514.6274
10 + 0 (SOA)	156.89766	157.1306	185.29915	476.9783
10 + 1 (SOA)	205.73378	195.8544	210.58344	541.6992
20 + 0 (SOA)	393.40283	299.28571	357.90707	534.4249
20 + 1 (SOA)	426.39258	322.89136	401.5611	729.7809

Table 38: The values of the PSNR parameter

	PSNR [dB]			
Length [km]	**MPEG-2 (576p)**	**MPEG-2 (1080p)**	**MPEG-4 (720p)**	**MPEG-4 (1080p)**
0 + 0	13.22293	19.93965	30.21836	14.54757
0 + 1	12.54105	18.0858	28.67666	14.06338
10 + 0	23.92984	18.32881	29.49807	14.58956
10 + 1	22.0485	17.38655	19.51874	13.61798
20 + 0	19.67365	16.51523	15.94564	13.56685
20 + 1	18.45625	15.87895	14.53003	13.96217
10 + 0 (SOA)	12.81335	16.68615	20.84776	14.50073
10 + 1 (SOA)	12.63341	13.0192	18.79916	13.6551
20 + 0 (SOA)	19.68214	16.58856	15.22297	13.55129
20 + 1 (SOA)	17.58712	15.49426	14.03891	12.56898

Table 39: The values of the SSIM parameter

	SSIM			
Length [km]	**MPEG-2 (576p)**	**MPEG-2 (1080p)**	**MPEG-4 (720p)**	**MPEG-4 (1080p)**
0 + 0	0.97911	0.95188	0.99364	0.94575
0 + 1	0.95943	0.94524	0.99144	0.9369
10 + 0	0.93932	0.93794	0.99268	0.94604
10 + 1	0.89674	0.92149	0.96816	0.93192
20 + 0	0.93533	0.90283	0.94551	0.93642
20 + 1	0.90561	0.90169	0.94328	0.93085
10 + 0 (SOA)	0.9493	0.92655	0.96477	0.94292
10 + 1 (SOA)	0.93665	0.91305	0.9549	0.93679
20 + 0 (SOA)	0.93652	0.90128	0.94205	0.93716
20 + 1 (SOA)	0.9274	0.90042	0.94105	0.92222

Measurement results using IPTV AXS 200/625

Tables 40, 41 and 42 shows the results of the IP packet loss, Average IP Bit Rate, and IP Arrival Jitter from IPTV.

Table 40: IP packet loss from IPTV

	IP Packet Loss [%]				
Length [km]	**MPEG-2 (576p)**	**MPEG-2 (1080p)**	**MPEG-4 (720p)**	**MPEG-4 (1080p)**	**DVB-T (576p)**
0 + 0	0	1.61	0	0	0
0 + 0.6	0.63	1.32	0	0.49	0.5
0 + 1	0.78	1.97	0	0.43	0.9
10 + 0	0.43	1.27	0	0	0
10 + 0.6	0.6	1.93	0	0.44	0.5
10 + 1	0.67	1.98	0	0.45	0.8
20 + 0	13.21	14.29	13.89	11.81	12.15
20 + 0.6	14.07	15.09	14.95	13.24	13.5
20 + 1	16.58	15.46	15.28	17	15.4
10 + 0 (SOA)	0.3	1.63	0	0.42	0.2
10 + 1 (SOA)	0.64	2.76	0	0.85	0.9
20 + 0 (SOA)	10.93	14.49	12.05	11.22	9.54
20 + 1 (SOA)	14.29	15.5	13.11	15.81	12.48

Table 41: Average IP Bit Rate from IPTV

	Average IP Bit Rate [kbit/s]				
Length [km]	**MPEG-2 (576p)**	**MPEG-2 (1080p)**	**MPEG-4 (720p)**	**MPEG-4 (1080p)**	**DVB-T (576p)**
0 + 0	4172	12902	3299	2604	4351
0 + 0.6	3466	11238	1590	2320	3684
0 + 1	2824	7367	1053	2004	3548
10 + 0	3855	13700	1524	2553	4051
10 + 0.6	3713	10231	1471	2581	3841
10 + 1	3300	8610	1416	2226	3404
20 + 0	3540	14489	1250	2309	3846
20 + 0.6	3527	10272	1210	2231	3745
20 + 1	3500	8693	1200	2105	3654
10 + 0 (SOA)	4005	12838	2152	2915	4156
10 + 1 (SOA)	3453	8164	1651	2668	3545
20 + 0 (SOA)	3969	13927	1336	3211	4054
20 + 1 (SOA)	3110	8539	910	2574	3456

Table 42: IP Arrival Jitter from IPTV

	Average IP Bit Rate [kbit/s]				
Length [km]	**MPEG-2 (576p)**	**MPEG-2 (1080p)**	**MPEG-4 (720p)**	**MPEG-4 (1080p)**	**DVB-T (576p)**
0 + 0	3.47	1.29	4.73	0.77	3.25
0 + 0.6	4.27	1.55	9.45	4.75	3.95
0 + 1	5.87	1.7	13.34	5.66	5.45
10 + 0	4.03	1.17	6.12	4.88	3.98
10 + 0.6	4.34	1.55	10.14	5.24	4.12
10 + 1	4.34	1.88	10.27	5.97	4.05
20 + 0	3.03	0.97	7.2	5.71	2.89
20 + 0.6	3.27	1.2	9.76	5.75	3.1
20 + 1	4.72	1.36	17.25	6.91	4.54
10 + 0 (SOA)	3.94	1.34	7.08	4.5	3.45
10 + 1 (SOA)	4.8	1.68	9.35	4.77	4.5
20 + 0 (SOA)	3.6	1.14	11.31	5.67	3.2
20 + 1 (SOA)	4.79	1.35	15.07	5.91	4.6

Measurement results using MOS and R-Factor in VoIP

Table 43 show the results of the MOS and R-Factor from VoIP.

Table 43: MOS and R-factor values from VoIP

Codec type	**G.711 μ-law**		**G.723.1-Acelp**		**G.729**	
Length [km]	**MOS**	**R-Factor**	**MOS**	**R-Factor**	**MOS**	**R-Factor**
0 + 0	4.37	91.28	3.64	70.93	4.02	80
0 + 0.6	4.37	91.25	3.64	70.87	4.02	79.95
0 + 1	4.28	88.88	3.57	68.08	3.94	77.89
10 + 0	4.37	91.23	3.64	70.92	4.02	80
10 + 0.6	4.31	89.91	3.64	70.87	4.02	79.92
10 + 1	4.27	88.41	3.57	69.02	3.94	77.76
20 + 0	4.28	88.9	3.64	70.92	4.02	80
20 + 0.6	4.27	88.26	3.64	70.86	4.02	79.92
20 + 1	4.27	88.39	3.57	69.06	3.94	77.87
10 + 0 (SOA)	4.37	91.28	3.64	70.91	4.02	81.12
10 + 1 (SOA)	4.29	88.94	3.57	69.11	3.94	77.92
20 + 0 (SOA)	4.37	91.31	3.64	70.93	4.02	80.01
20 + 1 (SOA)	4.29	88.95	3.57	69.12	3.95	77.94

Measurement results Data from Triple play

Table 44 show the results Downstream and Upstream from data.

Table 44: Measured data download and upload for Data

Length [km] (Optical + Metallic)	**Download [Mbit/s]**	**Upload [Mbit/s]**
0 + 0	23.348	5.941
0 + 0.6	11.383	0.983
0 + 1	5.999	0.315
10 + 0	21.148	4.565
10 + 0.6	10.481	0.966
10 + 1	4.461	0.373
20 + 0	20.302	3.833
20 + 0.6	9.053	0.922
20 + 1	4.006	0.158
10 + 0 (SOA)	21.804	4.368
10 + 0.6 (SOA)	12.250	1.051
10 + 1 (SOA)	4.594	0.139
20 + 0 (SOA)	20.476	3.402
20 + 0.6 (SOA)	13.278	0.990
20 + 1 (SOA)	3.943	0.105

C Set up services and applications

Configurations and settings for each service are shown in this appendix. You can see the initial login window in the figure 120. The following figure 121 shows an overview of all services in the form of virtual boxes.

Figure 120: Login screen for Triple Play services

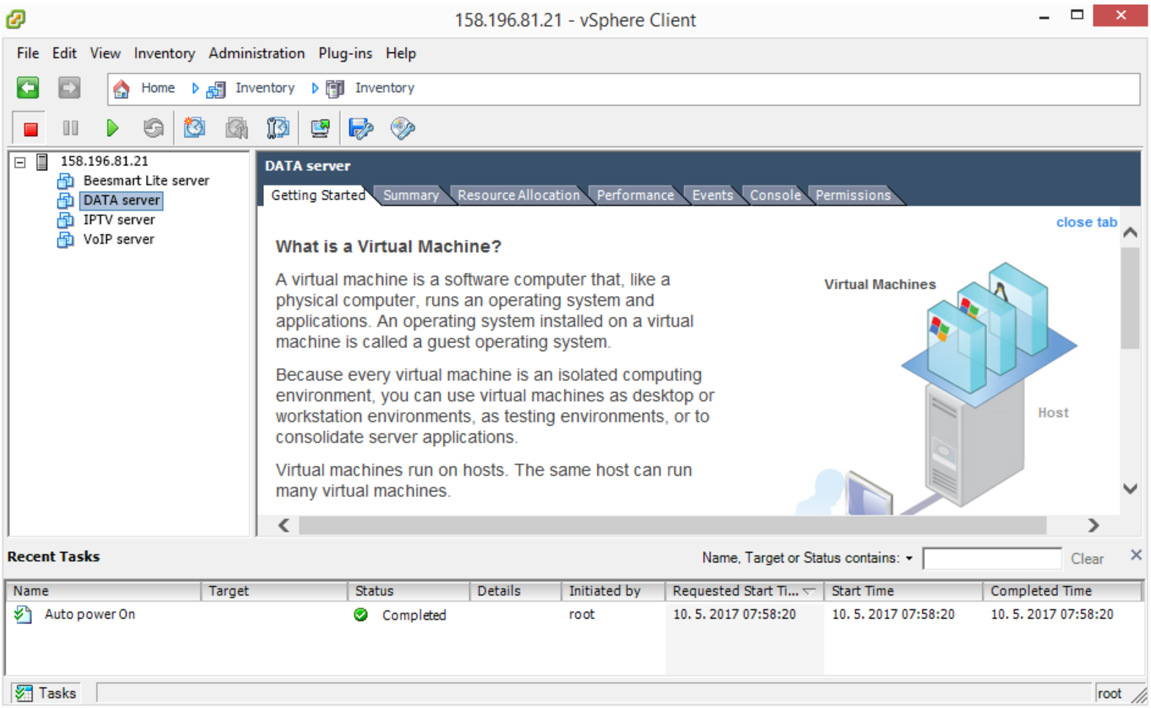

Figure 121: vSphere Client with Virtual machine

C.1 IPTV Set up

Video Broadcast Settings and receiving videos.

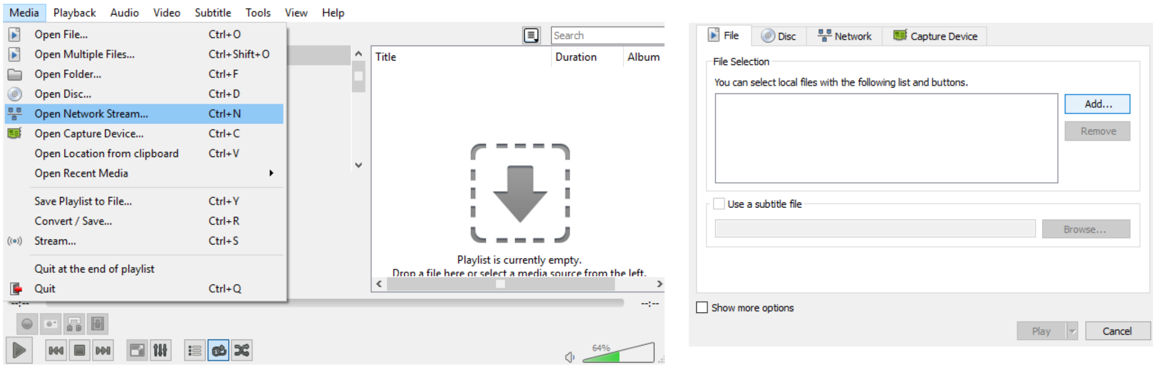

Figure 122: Set Video Broadcasts Step 1 and 2

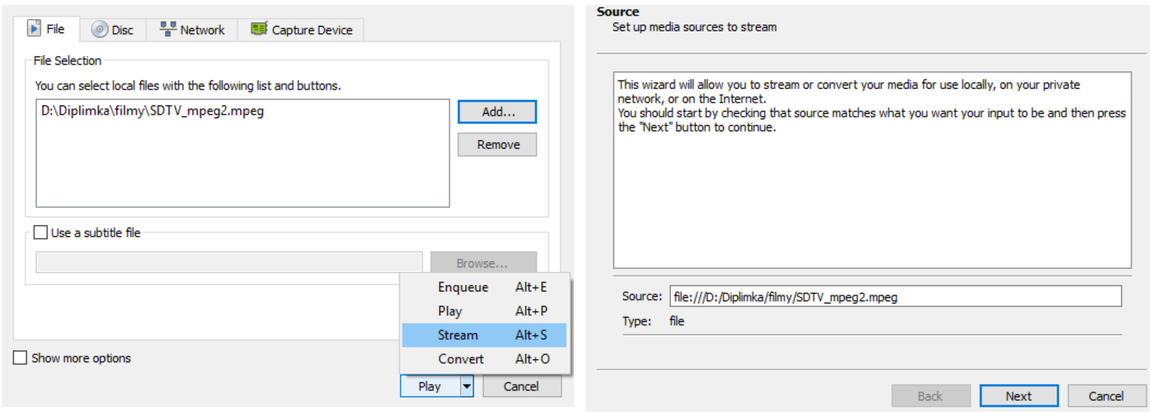

Figure 123: Set Video Broadcasts Step 3 and 4

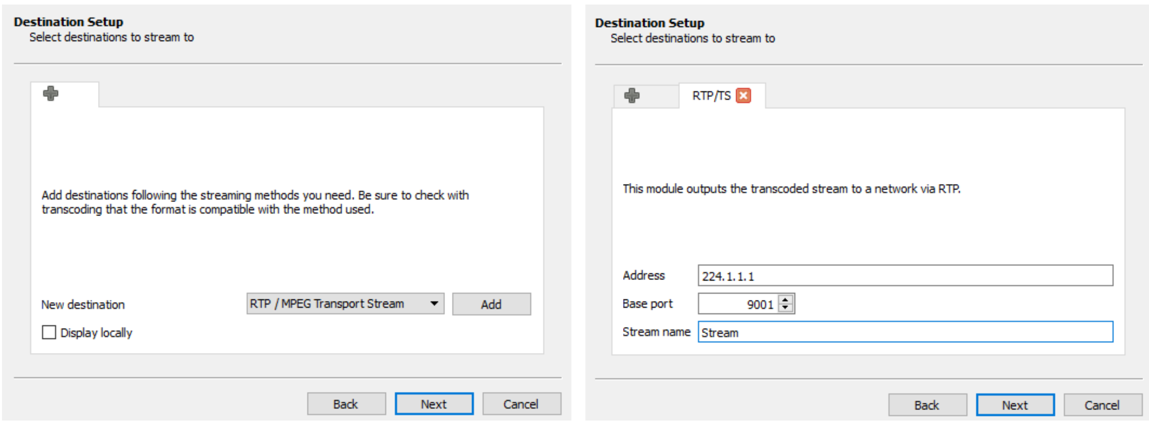

Figure 124: Set Video Broadcasts Step 5 and 6

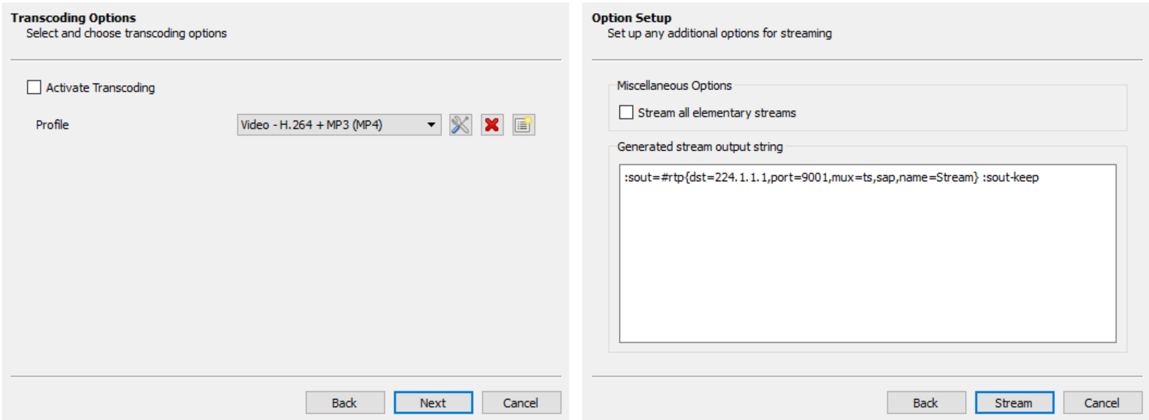

Figure 125: Set Video Broadcasts Step 7 and 8

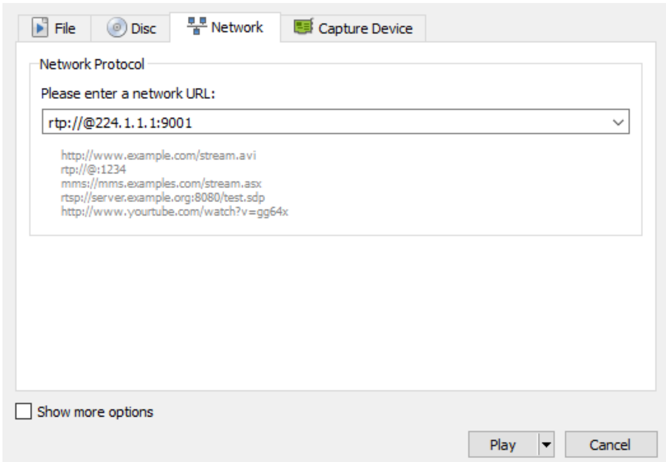

Figure 126: Step of receiving a video on the user side

C.2 Virtual box with IxChariot Set up

IxChariot software is installed on a Windows 7 virtual machine with the ixia password. The VirtualBox software was used to run the virtual machine. First, you need to create a new virtual machine. In the pictures below are the individual steps to run..

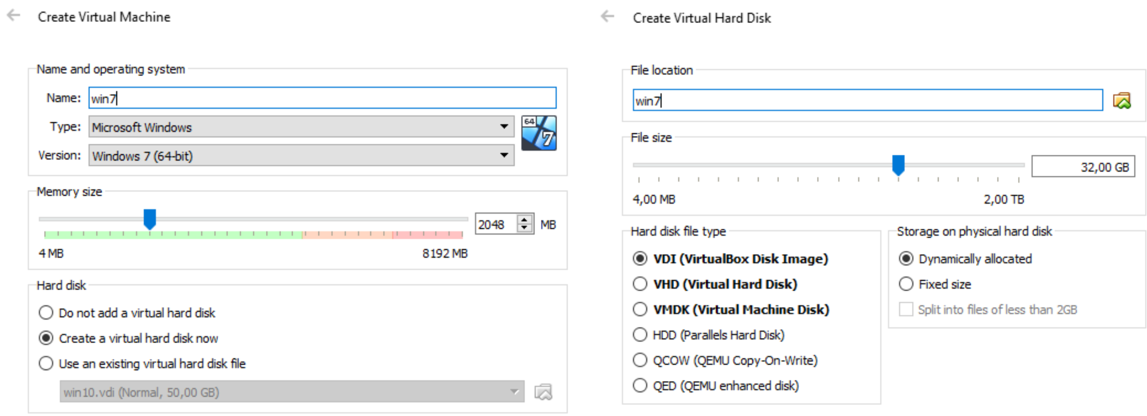

Figure 127: Start Virtual Machine - Step 1 and Step 2

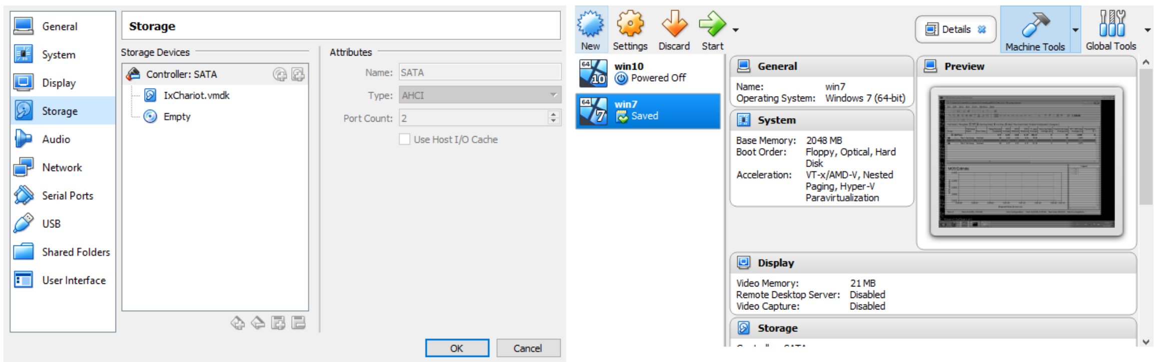

Figure 128: Start Virtual Machine - Step 3 and Step 4, left: boot virtual machine storage, right: select ready virtual machine

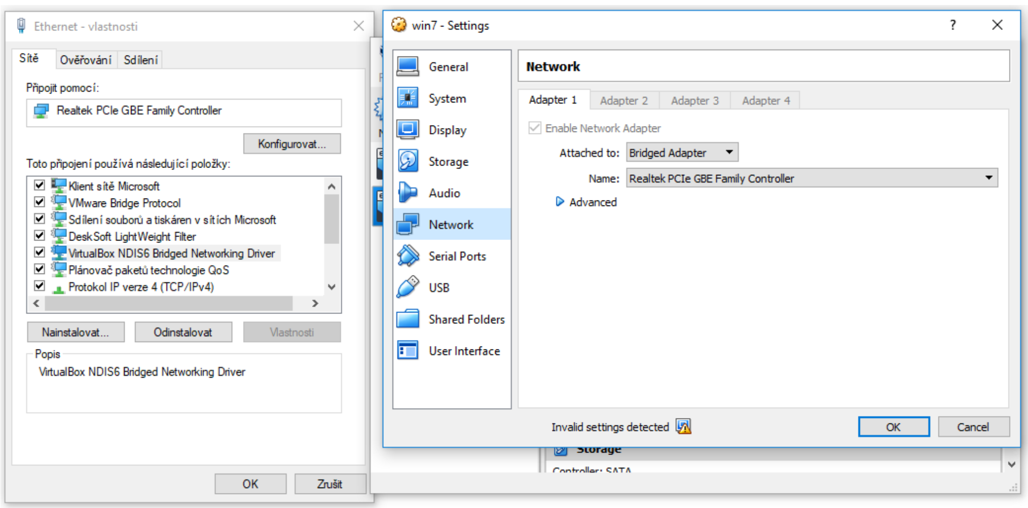

Figure 129: Start Virtual Machine - Step 5 and Step 6, left: settings bridge in Windows 10, right: In Virtual Machine set Network on Bridged

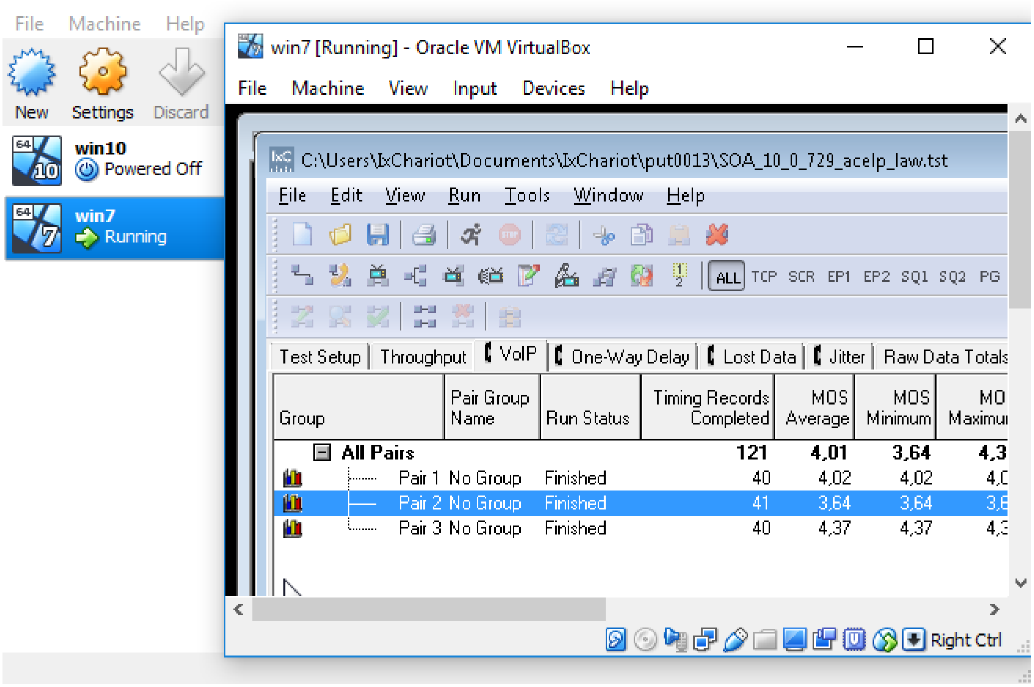

Figure 130: Running OS Windows 7 in VirtulBox password is ixia

C.3 Visual view on Web server

Web server view. The web browser after entering URL `10.1.4.10` displays the following statement shown in the figure 131.

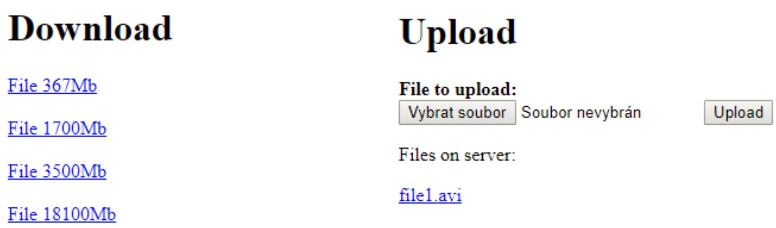

Figure 131: Web interface from Triple Play services