From Noise to Zapping

Quality of Service Issues Affecting Internet TV Distribution

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To Kerstin

Abstract

The Internet[1] and the development and deployment of new access network technologies has opened up for new information and communication applications. The mobile or cellular networks have evolved from a facility for voice call to today's IP only network. The evolution has also allowed for re-use of old and available infrastructure for technologies that increases the available capacity far beyond the original objective. The strive of having the Internet as the only 'carrier' for all applications, leads to change of transport channel for 'old' and well-known and well-established services. The customer thus expects the old behaviour - it is still the same service - but the changed channels makes the applications behave differently.

This thesis discusses performance parameters and monitoring with focus on Digital Subscriber Line (DSL) links and TV over Internet (IPTV) or Video on Demand (VoD). Studies of some typical disturbances that degrade the DSL channel and their impact on IP datagram transportation, and thus on IPTV Quality of Experience (QoE), are presented. Also, profiling of VoD users for pre-fetching and terminal caching is shown to be a possible path for increasing the QoE and lower the network utilisation.

[1]Internet is not *one* network, it is a network of individual networks with Internet Protocol (IP) as the common network protocol.

Preface

In today's changing world things turn quickly. Not only the technologies evolve but also the usage of them changes, sometimes beyond expectations. The Internet is a clear example of that. The multitude of applications and services that rely on a working network of networks could only have been foreseen by the Internet founders as a mere vision. Another example is the evolving of mobile telephony towards mobile computing. With the introduction of the iPhone a mobile telephone changed the user context from available most of the time for talks to always connected to social media and 'always on'. The demands on the applications' quality, whether delivered over fixed or mobile access networks, increases as the devices opens up for more refined service qualities.

Another sign of the pace of progress is the fact that when this work started optical fibre to the home was a surmountable vision. But the roll out especially in Swedish residential areas has really taken place during the last two three years. Even so, the interest in re-using copper based physical access networks for mobile fronthaul and backhaul is still valid because of the high cost for installing new infrastructure.

Along the way of my career I have been involved in building and managing campus networks and also to some extent wide area networks. This has placed my forte somewhere in the middle of the Open Systems Interconnection (OSI) reference model. Working with projects that are the basis for this thesis has broadened my scope to also include some of the mysteries of the physical layer as well as applications' quality issues. The theme for this thesis can be summarised in these key words: Quality of Service, Performance Monitoring and Management and Cross Layer Interaction targeting in the first place Internet TV distribution.

The work leading up to this thesis acknowledges the FP7[2] project COMBO[3],

[2]EU Research Funding 7th FrameWork Programme
[3]Convergence of Fixed and Mobile BrOadband access

the CELTIC+[4] projects R2D2[5] and NOTTS[6], Sweden's Innovation Agency VINNOVA and EIT Digital project NFMD[7], funding for laboratory equipment from Åforsk Foundation and the collaboration with Ericsson Research in Kista as well as Acreo Swedish ICT in Kista.

[4]EUREKA Cluster focusing on ICT and telecommunication
[5]Road to Media-Aware User-Dependant Self-Adaptive Networks
[6]Next Generation Over-the-top Multimedia Services
[7]Networks for Future Media Distribution

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Hjärup in September 2016

Acronyms and Abbreviations

ADSL Asymmetric Digital Subscriber Line

ADSL2 Asymmetric Digital Subscriber Line, version 2

ADSL2+ Asymmetric Digital Subscriber Line, version 2+

AL-FEC Application Layer Forward Error Correction

AM Amplitude Modulation

BBU BaseBand Unit

BFD Bidirectional Forwarding Detection

BS Base Station

CAPEX Capital Expenditures

CDF Cumulative Distribution Function

CODEC COder-DECoder

CPE Customer Premises Equipment

CPRI Common Public Radio Interface

CRC Cyclic Redundancy Check

CV Code Violation

dB Decibel

DELT Dual Ended Line Test

DMT Digital Multi-Tone

DiffServ Differentiated Services

DSL Digital Subscriber Line

DSLAM Digital Subscriber Line Access Multiplexer

DVD Digital Video Disc

EIN Electrical Impulse Noise

EMS Element Management System

EPC Evolved Packet Core

ES Errored Second

ETSI European Telecommunication Standards Institute

FEC Forward Error Correction

FEXT Far-End Crosstalk

FFT Fast Fourier Transform

FMC Fixed and Mobile Converged

FTTx Fibre Access Network

FTTH Fibre To The Home

GTP GPRS Tunnelling Protocol

HTML HyperText Markup Language

HTTP Hypertext Transfer Protocol

ICMP Internet Control Message Protocol

ICMPv6 Internet Control Message Protocol version 6

IDFT Inverse Discrete Fourier Transform

IETF Internet Engineering Task Force

IN Impulse Noise

INP Impulse Noise Protection

IntServ Integrated services

IP Internet Protocol

IPv4 Internet Protocol version 4

IPv6 Internet Protocol version 6

IPG Inter Packet Gap

IPPM IP Performance Metrics

IPTV TV over Internet

ISI Inter Symbol Interference

ITU International Telecommunication Union

ITU-T ITU Telecommunication Standardization Sector

LAN Local Area Network

LTE Long Term Evolution

MAVAR Modified Allan Variance

MCM Multi-Carrier Modulation

MDF Multiplexed Data Frame

MDI Media Delivery Index

MOS Mean Opinion Score

MPEG-2 TS Moving Pictures Expert Group, version 2 Transport Stream

MPLS MultiProtocol Layer Switching

MTIE Maximum Time Interval Error

MW Medium wave

NCSA National Center for Supercomputing Applications

NCP Network Control Protocol

NEXT Near-End Crosstalk

NIC Network Interface Card

NMT Nordic Mobile Telephony

NRM Network Resource Manager

NTP Network Time Protocol

OAM Operations, Administration, and Maintenance

OFDM Orthogonal Frequency-Division Multiplexing

OH Overhead

OPEX Operating Expenditures

OS Operating System

OSI Open Systems Interconnection

OTT Over The Top

OWAMP One-way Active Measurement Protocol

PAM Pulse Amplitude Modulation

PC Personal Computer

PDV Packet Delay Variation

PEIN Prolonged Electrical Impulse Noise

PMD Physical Media Dependent

PMS-TC Physical Media Specific Transport Conversion

PSTN Public Switched Telephone Network

PTM Packet Transfer Mode

PTM-TC Packet Transfer Mode Transport Conversion

QAM Quadrature Amplitude Modulation

QoE Quality of Experience

QoPh Quality of Physical Layer

QoS Quality of Service

R2D2 Road to Media-Aware User-Dependant Self-Adaptive Networks

RED Random Early Discard

REIN Repetitive Electrical Impulse Noise

RF Radio Frequency

RFI Radio Frequency Interference

RMON Remote Network Monitoring

RS Reed-Solomon

RSVP Resource Reservation Protocol

RU Radio Unit

SELT Single Ended Line Test

SES Severely Errored Second

STB Set Top Box

SDSL Symmetric Digital Subscriber Line

SHINE Single High Level Impulse Noise Events

SLA Service Level Agreement

SMON Remote Monitoring for Switched Network

SNMP Simple Network Management Protocol

SNR Signal to Noise Ratio

TCP Transport Control Protocol

TPS-TC Transport Protocol Specific Transmission Convergence

TWAMP Two-way Active Measurement Protocol

UDP User Datagram Protocol

URI Uniform Resource Identifier

UUCP Unix to Unix Copy

VDSL Very-high-bit-rate Digital Subscriber Line

VDSL2 Very-high-bit-rate Digital Subscriber Line version 2

VoD Video on Demand

VoIP Voice over IP

VQM Video Quality Metric

WWW World Wide Web

XML Extensible Markup Language

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Chapter 1

Introduction

1.1 Background

The development pace of new communication technologies is accelerating. Consider that the first patent for a working telephone in 1876 was followed by a nearly 100 yearlong evolution of the same technology before the first mobile phone systems aimed for the consumer market was introduced in 1981. The evolution of mobile or cellular telephone systems followed a pattern of a new and vastly more advanced system ever 10 years.

1.1.1 Internet Challenges

The Internet has two "birthdays". The first one is 29th of October 1969 when Charlie Kline sends the first message on the ARPANET. The second birthday is the 1st of January 1983 when the ARPANET abandoned the old protocol Network Control Protocol (NCP) and started to only rely on The Internet Suite (TCP/IP)[1]. Since then the Internet has seen a great evolution over a relatively short period of time. This is true especially regarding applications but also, needless to say, regarding network technologies.

The initial driver for Internet was for transferring of data between computers or computer sites. Already 1971 a mail application was added. In 1979 a simple form of social media was formed in Usenet called News where users could ex-

[1]Both protocols had lived side-by-side in ARPANET for a longer period.

change information in a mail like fashion over different networks like Unix to Unix Copy (UUCP) and of course the Internet. To publish, search and retrieve documents and data the Gopher protocol was introduced in 1991, Gopher servers and clients for the most used platforms were developed. What the 'Gopher technology' lacked was developed by Sir Tim Bernes-Lee during 1989 and 1990: The HyperText Markup Language (HTML), Uniform Resource Identifier (URI) and the Hypertext Transfer Protocol (HTTP). This is what was to be called the World Wide Web (WWW), which linked together information of any sort and in any electronic form stored anywhere. At first the web was only available inside CERN[2], but in 1991 it was opened for the global community. With the release of the first web browser, the NCSA[3] Mosaic in 1993, the stage was set for the information society to flourish. This also marked the beginning for developing and presenting services produced by public enterprises and organisations outside the academia aimed for the public audience.

Parallel to this, the term *multimedia*, formed already in 1966, started to be filled with content. The step to include multimedia as a vital part in the world wide web was of course a minor one, but affected the way telephony services, radio and TV were distributed. The global Internet as well as operators' Internet Protocol (IP) based intra-domain networks took over as infrastructure also for these services, and the term *triple play*[4] was created. Traditional services that were distributed over analogue networks, be they wired or wireless, moved over to the packet switched best-effort network infrastructure. For the purpose of reusing infrastructure already at hand, Digital Subscriber Line (DSL), among other technologies, were developed and rolled out. Internet had become a consumer product.

The Hour Glass model in Figure 1.1 represents the fact that today's global data communication is built entirely on all IP networks. But, like the original Open Systems Interconnection (OSI) reference model, the Hour Glass model has limitations. Both models are based on each layer being fully independent of other layers, communication between neighbouring layers is performed in the interfaces. Especially wireless links introduce new challenges regarding the need for cross layer interaction. For example, Transport Control Protocol (TCP) is based on a model were links are near to perfect and have deterministic behaviour; All packet loss in a TCP session is regarded as originating from congestion in active network nodes. A not so perfect link e.g. a wireless link with packet loss deteriorates TCP's performance. A solution is to either make TCP aware of performance on intermediate links in the end-to-end path, or to create link functions that cover, for example, packet loss at higher layers.

[2] European Organization for Nuclear Research

[3] National Center for Supercomputing Applications, University of Illinois at UrbanaŰChampaign

[4] Triple play means that telephony, IPTV and the traditional best-effort service Internet is delivered over the same network

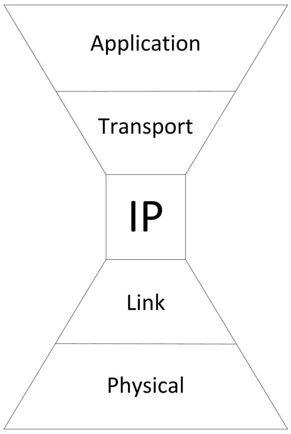


Figure 1.1: The Hour Glass reference model.

In itself, a transition from one transmission technology to another sounds just like a pure technical issue. But new technology introduces new types of errors and sometimes unexpected issues. Going into the consumer market poses other problems. The consumers form a user group of great diversity regarding technology awareness and understanding; a technology nerd handles issues in a different fashion than the average user. Also, consumers of a specific service do not care about the delivery mechanisms for this service; the service should be delivered with the same or better quality after the transition to the new infrastructure.

Opening up Internet for the customer market also have other implications. Throwing bandwidth on every problem is probably too expensive. The real Operating Expenditures (OPEX) and Capital Expenditures (CAPEX) costs are found when the network is utilised close to its maximum. But for to utilise close to maximum, over-subscription has to be deployed; subscribers do not use their available capacity constantly. Also, *user* is not the same as *subscriber*; many users can and will use the same one subscription, and these users could most probably gain from diverting usage profiles. Not only intra-subscription exertion, but also external events, new applications and services affect usage patterns. Thus, a subscriber's usage pattern might change dramatically.

All of this creates new demands on the service's support. As an example a distribution of OPEX per DSL subscribers shows that some subscribers are extremely costly, in fact to such degree that the overall revenue is severely decreased, see Figure 1.2. One road towards a solution is to try to identify errors and correct them automatically without user intervention by implementing some sort of network resource manager.

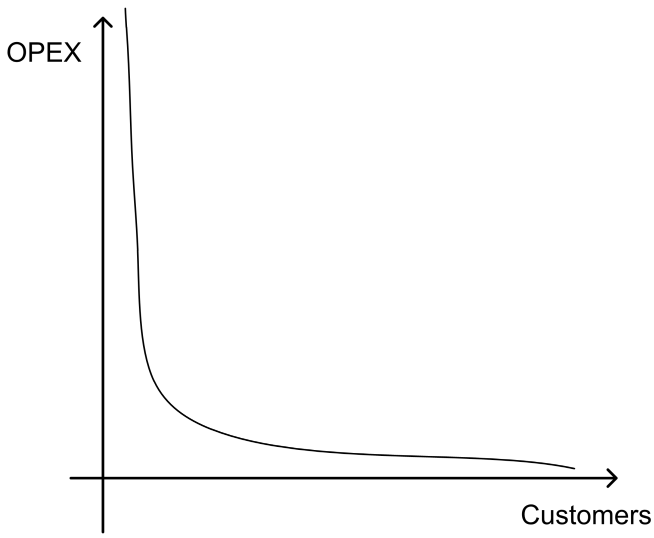


Figure 1.2: Distribution of OPEX per customer.

1.1.2 DSL and Mobile Challenges

The introduction of mobile phone systems started in the Nordic countries in 1981 with the introduction of Nordic Mobile Telephony (NMT). From there a global customer market expanded more or less exponential, with new mobile systems and applications. Today, the mobile stations are designed and used more as a multi-purpose client for Internet based applications than as a device for voice calls. The usage of the mobile networks become more like the usage of the fixed Internet; mobile stations are always on and always connected. The demand for more capacity in the mobile networks leads to an increasing number of cells, covering smaller and smaller areas. Today, two more or less parallel networks can be identified, the fixed best-effort network and the mobile back-haul network. Currently, efforts are made to combine these two networks to one, a converged network.

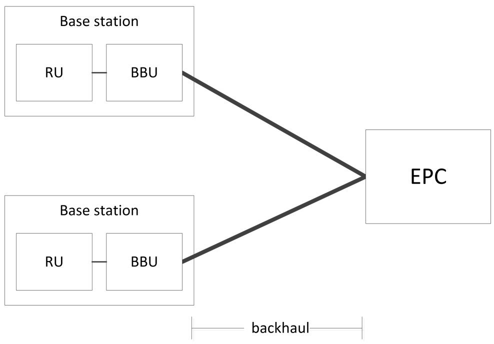
Different flavours of DSL are still the most common types of broadband access technologies globally with approximately 35% market share for central office type of deployments (ADSL2, ADSL2+, SDSL etc), approximately 25% market share for Fibre Access Network (FTTx) solutions with VDSL2 in Q4 2015. The market share for Fibre To The Home (FTTH) is approximately 20% [45]. FTTH is growing with 60% between the fourth quarter of 2014 and 2015 and FTTx with 15%. DSL solutions are decreasing by 19% in the same period. Until today, DSL technology has not been used widely for mobile backhaul purposes, but this is about to change as the development of mobile networks and specifically heterogeneous networks with small cells drives the need for flexible and low cost mobile backhaul links. Copper mobile backhaul is also enabled by the advancement in DSL since the latest features with vectoring and multi-pair bonding now can cope with the mobile backhaul bit-rate requirements [59]. The bit-rate will even continue its path to ultra-high access speed as new initiatives such as 4th generation broadband systems [46] and the G.Fast ITU-T standard G.9701 [42] deliver gigabit speed over the last copper drop.

The deployment of high capacity cellular technologies like Long Term Evolution (LTE) might negatively influence the interest of using DSL technology in access networks. On the other hand, the evolution of cellular networks with respect to more access points or base stations in a denser configuration calls for new backhaul solutions, and in this light DSL solutions are still of great interest.

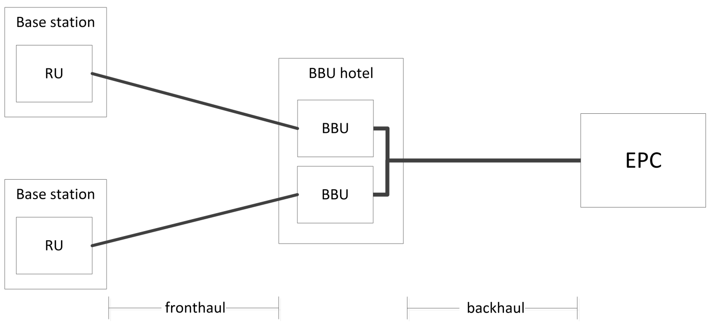
Figure 1.3 gives a brief description of the mobile backhaul and fronthaul. In Figure 1.3(a) the Radio Unit (RU) and the BaseBand Unit (BBU) are co-located in the Base Station (BS). The backhaul connects the base station with the Evolved Packet Core (EPC), the mobile core network. In Figure 1.3(b) only the RU is located at the base station, while the BBUs are located at a centralised so called BBU hotel. The baseband signal is sent from the BBU to the RU via the mobile fronthaul.

The re-use of existing copper infrastructure does however bring its challenges since the quality of the copper cables can vary significantly. In many locations the copper cables are more than 50 years old and might only have paper as insulation. Pure metallic faults such as corroded joints and wire cuts will directly affect the transmission channel and degrade, or completely kill, the performance. Poor balance will also make the links sensitive to external noise, which can significantly reduce the link stability if not mitigated correctly. The risk of transmission issues such as noise disturbance has increased as the development of DSL has steadily pushed the transmission towards higher frequencies. Modern triple play services, such as TV over Internet (IPTV), has also made the impact of bit errors on the Quality of Experience (QoE) larger compared with when copper access was used only for browsing and mailing; A lost packet will be directly visible as pixilation or freezing of the TV picture. For mobile applications the users expect the same QoE as for their wireline services and if several users are connected over the same backhaul link they all risk being affected by the transmission issues. It is then even more important to minimize downtime and keep a high stability of each link.

Noise and disturbances in a DSL environment can have many sources. Impulse noise from the power line is one example. More or less continuous radio signals can have a negative influence on a DSL link. Poor wiring of the Public Switched Telephone Network (PSTN) network is also known to negatively affect a DSL



(a) BBUs are located at the base stations.



(b) BBUs are centralised to a BBU hotel with fronthaul between the RU and the BBU hotel.

Figure 1.3: Two generations of a mobile access network.

link. Depending on the type, power and duration of the disturbance, the result can be a decrease in the link's capacity, just a short interruption of the packet flow or in worst case a total re-training[5] sequence.

Disturbances in the access network have direct impact on the perceived quality (QoE); A physical layer bit error could result in a lost packet on the network layer. The DSL technology suffers from limitations that have become evident as Internet based TV grows in popularity [81].

All this put high requirements on the performance management and monitoring solutions in order to secure high Quality of Service (QoS) and QoE while keeping OPEX as low as possible. Because of its importance, performance management for copper access is something that has been studied in several projects before, such as the FP6 large integrated project MUSE [25], and techniques and monitoring parameters are defined in several standardization bodies.

1.2 Related Work

A global problem description and motivation for this thesis can be concluded in the fact that real time application and content delivery requirements are adapted and implemented over a packet switched best-effort network with many and diverted access network technologies.

1.2.1 Quality of Experience and Quality of Service

Understanding the relationship between the perceived QoE and the QoS parameters of underlying layers is the basis for the development of new tools and managing systems in this area. The European Telecommunication Standards Institute (ETSI) guidelines [30] suggest that QoE should focus on user touchpoints for the whole lifecycle of a given service, including the selection and use process. Example data relating to QoE is provided for different service types. For video services two metrics are defined, which as expected relate to delay: end-to-end packet delay and lip sync (audio/video alignment).

Internet based real-time video streaming comes in two flavours: IPTV and Over The Top (OTT). IPTV systems are fully controlled by an operator, from video head-end to the user, since the operator is responsible for both the content and

[5] Training is the procedure done each time a DSL link is initialised or when bit swapping is no longer possible or feasible. It normally takes approximately 20 to 30 seconds, in during that time no packet transport is possible.

the network. OTT uses the public Internet and the content distributors are not controlled by the access network operator. OTT is therefore delivered as unicast and cannot be separated from other best-effort services like web browsing and file downloading. A suggested solution is a media aware, self-adaptive network, a topic studied in the Celtic project Road to Media-Aware User-Dependant Self-Adaptive Networks (R2D2) [37]. The full delivery chain, from content server to end user, has to be monitored on all levels by a network resource manager that can automatically take corrective measures in all nodes involved in the delivery.

The ITU Telecommunication Standardization Sector (ITU-T) IPTV requirements document [26] also defines QoE in terms of human perception and user centric behaviour. It is further noted in [28] that QoE may depend on context and user expectation. The possibility of using subjective Mean Opinion Score (MOS) data measured for different QoS scenarios is explored with the intention of supporting a model allowing:

1. Prediction of QoE based upon QoS results
2. Derivation of QoS parameters for a given QoE requirement

Many methods for estimating QoE from foremost the network or IP layer QoS parameters are proposed. Already in 2002 Khirman and Henriksen suggested a method for relating objective QoS parameters with the subjective QoE regarding Voice over IP (VoIP) [82]. In [83] Kim et al show that, among QoS parameters on layer 3, packet loss has a relative importance degree on IPTV of 41,7%. [26] deals with how delays within the network contribute to QoE, especially focusing on how users select channels in IPTV services. A key QoE implication is defined relating to channel zapping time[6]. Users are accustomed to the service and quality level of broadcast television. Therefore, the requirements on real-time video over IP are high. For IPTV, the mean time between visible errors must not be less than four hours [23] [24]. There must be no more than one Errored Second (ES)[7] in the bit stream per hour for standard definition TV and one ES per four hours for high definition TV [97].

Due to the efficient compression techniques used in IPTV, packet loss has severe effects. Already at a non-recovered packet loss of 0.1%, the viewers lost interest and the viewing time decreased with 50% [89]. Note that errors are not only found on the physical layer, e.g. bit errors, but are also introduced in network nodes on the network layer where packet discarding is a feature, e.g. Random Early Discard (RED).

[6] The zapping time is the time from when the user requests a change of view till the selected channel is actually played out

[7] An error second is a second during which at least one error, e.g. a code violation, has been recorded. This is described in more detail in Section 3.4.1.

The reference for video quality is subjective experiments, which represents the most accurate model for obtaining video quality ratings [107]. In a subjective experiment, a group of viewers are asked to watch a set of video clips and rate the image quality. Disturbances are introduced in a controlled way. The average rating of all viewers for a specific scenario (video plus disturbances) is also called the MOS. MOS (see for instance [29]) and Media Delivery Index (MDI) [20] are examples of QoE parametrisation methods. The cross layer interaction between Packet Delay Variation (PDV), sometimes denoted jitter, and packet loss and the perceived QoE by the user measured as MOS was also studied in the R2D2 project.

To ensure that TV applications, either IPTV or OTT video, meet the end users' QoE expectations, it is necessary to develop tools and methods to monitor and assess the QoE of the video bit stream in real-time. Thus, it would be preferable to see ways to predict perceived quality using just physical QoS characteristics [101].

1.2.2 DSL Access Link Performance

The user employment of Internet based services has changed radically over the last couple of years, which puts new demands on the existing access technologies. The increasing demands concern both more extensive use of capacity demanding services like streaming video, as well as new technologies where small cells starts to enter the home network. The next generation mobile access networks will call for exertion of currently available access networks, including copper based networks technologies, for mobile backhaul (or even fronthaul). In these environments one link will service many users that, though mobile, have the same quality demands on services, such as video, as if they where streamed over fixed access. These increasing demands on the access network, together with the upcoming Fixed and Mobile Converged (FMC) networks, will be applied on Very-high-bit-rate Digital Subscriber Line (VDSL)2 links and the newly developed ITU-T standard G.9701 (G.fast) [2] in combination with FTTx [106, 91, 64]. This is indeed a new challenge for the DSL technology.

Over-provisioning has always been a good way to reduce packet loss, delay and PDV in a network. However, IPTV over DSL links does not permit over-provisioning. A single bit error causes the loss of a full network layer datagram. Typical DSL bit-error rates of 10^{-7} translate to packet loss rates in the order of 10^{-3}, which approximately produce a visible error every few minutes. [54]

In [88], a refined method for solving the shortages in DSL that introduces packet loss is presented. The method relies on unicast retransmission of lost packets. As a support for a Retransmission Server, Peer-assisted Repair (PAR) is introduced. A lost packet can be retained from a neighbour Set Top Box (STB) as well as from a retransmission server. Forward Error Correction (FEC) packets are also used. This calls for updates of the STBs, but also adds demands on the DSL uplink. This requirement is modest according to the authors.

The necessity for DSL error control is discussed in [56]. Moving Pictures Expert Group, version 2 Transport Stream (MPEG-2 TS), the most common transport technique for video content, is designed for low packet loss and PDV, which is not the case in an IP network. Interleaving (in DSL) cannot be made too deep because of increased delay and buffer space. FEC on the application layer (AL-FEC) cannot be utilized too much without hitting delay thresholds; and the protection period is correlated to the well-functioning of channels switching and rewind or fast-forward operations. As a conclusion, a hybrid of AL-FEC and retransmission is suggested to overcome the techniques respective drawbacks, but such techniques are not thoroughly investigated. The paper [56] also discusses admission control, and challenges like Change Time, Network Management and Video Quality Monitoring.

In [57], physical-layer impairments and error-mitigation techniques for DSL environments are investigated. The objective is to evaluate FEC as an error-control for IPTV over DSL. The focus is on impulse noise which is a non-stationary stochastic type of noise that is induced due to electromagnetic interference from domestic sources and external events.

To be able to realise new technologies and services e.g. mobile backhauling over DSL access networks, a deeper understanding and relation of measured QoS parameters on different layers in the OSI reference model is essential.

1.2.3 Monitoring of IP Performance Parameters

The network layer is the lowest layer where it is possible for any end user to actually monitor performance over the full path. Lower layers are local, and performance parameters are only available for the operator in question. This implicates that any infrastructure between the two end points has to be seen as a black box.

ping and *traceroute* are examples of monitoring tools available in any operating system. These tools operate *in-band*, meaning the monitoring takes place utilising the same path and nodes that is used for the actual data transfer. *Out-of-band* management uses a special infrastructure, parallel to the data path, attended for the sole purpose of monitoring and management.

Ahlgren et al uses a train of packets, transmitted as close together as possible [48]. A train is defined as a number of packets sent as closely as possible. Each packet is time-stamped at the sender side and at the receiver side. Ahlgren estimates the bandwidth at the path's bottleneck, actually the worst bottleneck in the full path, by dividing the spacing time at the receiver with the packet size; all packets in a train are of the same size. Ahlgren states that the longer the train, the better is the accuracy of the estimate. The train length, as well as the packet size, is a delimiter in our experiments. If the test packets or the trains are too long, the *user data delivery* will be badly influenced.

Bregni [62] uses a similar method for estimating the jitter. They do not use trains, but time-stamps packets in a similar way as Ahlgren. By comparing the difference of both transmitting and receiving packet time-stamps the jitter can be estimated. Bregni uses a Least Square estimate of the initial time offset of the transmitter and receiver clocks, as well as the clocks fractional frequency offset. The measured jitter data was analysed using Modified Allan Variance (MAVAR) and Maximum Time Interval Error (MTIE). Bregni uses the Real-time UDP Data Emitter/Collector. [86]

Wang [105] warns for the long-range or self-similar dependence characteristics in network traffic; Scale-invariant burstiness will be induced, which in turn will introduce changes in delay due to varying utilization of queues. This leads to an underestimation of delay if poissonian sampling is used. The question is if this factor will interfere with our proposed experiments.

Ishibashi [79] also discusses the underestimation of the user data delay that active probing can suffer from. To make the estimation more accurate a combination of active and passive probing is suggested. This is done by calculating the number of packets arriving between two successive active probing packets and adjusts the active probing measurement accordingly.

Ciavattone [65] discusses poissonian and periodical probing/sampling. Poissonian probing detects network events, while the periodic probing is used to detect characteristics seen by an IPTV stream.

1.2.4 Cross Layer

Solutions to many of the issues discussed here call for going outside of the layered reference model; layers that are not close neighbours need to interface and exchange information as indicated in Figure 1.4.

Much of the previous research in the cross-layer related area has been focused

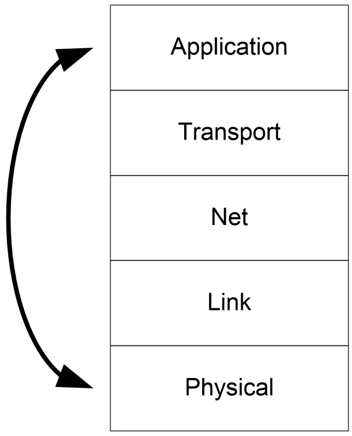


Figure 1.4: Cross Layer means interfacing between functional layers of the OSI reference model that are not direct neighbours.

on the understanding of the relations between the network layer up to the application layer. This is perfectly relevant considering the OSI reference model; the impact on higher layer delivery quality from the network layer should be independent of lower layer technology. Many of the studies, e.g. [85], are directed towards video distribution whereas Kim *et al* in [83] state that packet loss is the QoS parameter on the network layer that has the highest relative importance degree (41.7%) regarding IPTV.

The relation between QoS parameters on lower layers, e.g. impact on layer 3 from impairments on layer 1, is not so deeply studied. On layer 1, all parameters that have an impact, direct or indirect, on upper layer QoS parameters must be defined as QoS parameters. For example, a change in Signal to Noise Ratio (SNR) in DSL affects packet delay variation and packet loss on layer 3 [52][61]. Another example is [60] where performance metrics on the physical level (QoPh) are compared with the objective QoE parameter VQM[8]. Bogović et al discuss in [58] the relation between QoE and DSL physical layer QoS parameters. Orosz et al. [95] investigated the correlation between video QoS and QoE. By studying these parameters for a specific DSL link, triple play QoE can be estimated.

In [76] Goran et al have investigated the impact of physical layer disorders on both QoS and QoE in an Asymmetric Digital Subscriber Line, version 2+ (ADSL2+) system. Different disturbance types have different impact on QoS and QoE dependent on Asymmetric Digital Subscriber Line (ADSL) link setup. The number of error counters such as ES and Severely Errored Second (SES)

[8]Video Quality Metric (VQM) is a method for predicting a user's perceived experience from QoS parameters [109].

are directly correlated to the quality of an IPTV stream. Škaljo *et al.* also estimate the impact of impairments on the physical link have on IPTV quality [99]. They, like Goran et al, use an ADSL2+ system for their experiments. Five physical layer parameters are used, among them ES. They find that not all Code Violations (CVs)[9] cause decreased QoE, which probably is due to that the disturbance hits layer 3 packets carrying other services. [72] proposes a model for capturing the cross-layer dynamics of CRC and Far-End Crosstalk (FEXT) and the impact on packet loss.

In [66] Souza *et al* described how non-stationary noise impacts a DSL system. Experiments were performed with a DSL connection over a wireline simulator, where a noise generator injected spikes of noise. It was determined that the packet loss rate, packet loss count, bandwidth and transfer delay are not suitable for a detailed analysis of impulse noise impact. Begen investigated physical-layer impairments and error-mitigation techniques for DSL environments in [57]. The objective was to evaluate FEC as an error-control for IPTV over DSL. The necessity for DSL error control was further discussed in [56] by Begen *et al.*

1.2.5 Prefetching and Caching

A provider of a service or product should strive to keep the customer satisfied. In the context of IPTV and Video on Demand (VoD) this includes providing as low latency as possible between user action and service delivery. Reducing the *zapping time*, i.e. the time from channel selection to start of playout, is one example, quick reaction to fast forward or rewind another. A solution is to predict the user's next action and prefetch larger or smaller program parts accordingly. The challenge is to predict the correct content, but also to prefetch the right amount of this content so that the zapping time is as low as possible while the continued playout will be performed without glitches. To download data that is never used is of course a waste of resources. Knowing the user is essential for an effective utilisation of the infrastructural resources. This is a motivation for user profiling.

Earlier studies discussing zapping in different types of TV networks exist. Cha *et al.* have analysed user behaviour in an IPTV network [63]. Three user modes are defined and analysed, *surfing*, which is the equivalent to what herein is called *zapping*, *viewing* and *away*. Surfing is defined as the *channel hold time* being between one second and one minute. Once an interesting program is found the user goes into viewing mode, which has channel hold times above one minute. Finally, away is when the channel hold time exceeds 60 minutes. The used dataset did not register user turning of the TV or Set Top box, so instead it

[9] A CV occurs when the Cyclic Redundancy Check (CRC) decoder indicates an error.

is proposed that the user has turned off the TV or the Set Top box if the channel hold time is greater than one hour.[10] Approximately 95% of all channel hold times are either in the surfing mode or in the viewing mode. Plotting the frequency of how long a user stays with one channel before moving to the next show that channel switching is done as early as after four seconds. But over 60% of all channel hold times are less than 10 seconds and they are positively related to the popularity of channels or programs. Also more than 60% of channels changes follow a sequential order.

In [75] Gopalakrishnan *et al.* are modelling the behaviour of a single user - the Couch Potato - viewing session with events like Start, Pause, Play, Fast Forward, and Rewind. From state Play 63% of all state changes go into Fast Forward, an indication of zapping behaviour.

Ali-Edin *et al.* discusses the *impatient user* behaviour in [49]. In their study over 90% of all sessions last less than one hour, and 20% less than 30 seconds. A comparison with Yu *et al.* [110] shows the same behaviour, except for views lasting longer than 10 minutes. The impact on prefetching and caching is not studied.

Program popularity's impact on caching is the main focus of Abrahamsson *et al* [47], but it is noted that while most of the users request only a few videos per month some users have number of requests per day in the order of hundreds. These requests are often very short, less than 5 minutes, and the user tend to request several different programs.

The analysis by Du *et al.* in [68] is focused on pre-fetching in a VoD service but has also looked into the viewing time per request. 20% of all requests are shorter than two minutes, which can be defined as zapping. The impact from zapping on caching and/or pre-fetching strategies has also been studied by Du *et al.* [68] and Zhang [113]. In these studies no categorisation of users were done.

One note from [53] is appropriate in this context. Repeats of YouTube streams are self-generating (a clip is more prone to be played if it has already been seen many times), long-lasting (users tend to return to a clip even after several days) and semi-regular (clips tend to be repeated with some regularity).

[10] Very few programs, if any, in the dataset have a duration of more than 60 minutes.

1.3 Outline

The chapters of this thesis are organised as follows:

- Chapter 2 gives a brief introduction to DSL systems.
- In Chapter 3 performance monitoring aspects and parameters on different levels of the OSI reference model and their relations are discussed. Also a review of applicable standards is presented.
- Chapter 4 takes the theoretical discussion into practise. The test lab and the equipment is described and some packet generators/analysers are evaluated.
- The impact of broadcast radio signals on IP packet transportation over a DSL link is analysed in Chapter 5.
- Impulse noise is probably the major disturbance that affects DSL links. How IP packet loss is affected by impulse noise is evaluated in Chapter 6.
- One method to keep the user satisfied is to allow for the network itself to adapt to changes in performance, and to allow the services delivered adapt to the network's current quality of service. Chapter 7 presents contributions to a proof-of-concept project for such a functionality.
- A study on the application layer ends the path through the OSI reference model, from bottom to top. In this case traffic data from a VoD network is analysed. Profiling users is important for prefetching and caching policies in VoD networks. A special user group is browsers or zappers, users who consumes minor parts of programs when searching for something interesting, is studied in Chapter 8.
- The thesis ends with Conclusions and Future work.

1.4 Deliverables and Publications

The thesis is based on the following publications and project contributions, except for Chapters 2 and 4, which are based on unpublished internal thesis project milestones.

- Chapter 3: M17: Survey of monitoring parameters & methods. Project milestone, COMBO - Convergence of fixed and Mobile BrOadband access/aggregation networks, 2013. Section 1.2.2.

- Chapter 5: Jens A Andersson, Maria Kihl, Stefan Höst, and Daniel Cederholm. Impact of DSL link impairments on higher layer QoS parameters. In *Proceedings of SNCNW 2012.* 8th Swedish National Computer Networking Workshop, SNCNW 2012, 2012. [52]

- Chapter 6: Jens Andersson, Stefan Höst, Daniel Cederholm, Maria Kihl, et al. Analytic model for cross-layer dependencies in Very-high-bit-rate Digital Subscriber Line version 2 (VDSL2) access networks. In *Software, Telecommunications and Computer Networks (SoftCOM), 2014 22nd International Conference on, pages 269-273.* IEEE, 2014. [50]

- Chapter 7: Deliverables D422 and D424, Celtic Plus project Road to Media-Aware User-Dependant Self-Adaptive Networks (R2D2). [37]

- Chapter 8: Jens A Andersson, Maria Kihl, Åke Arvidsson, Manxing Du, Huimin Zhang, Stefan Höst, Christina Lagerstedt. User profiling for Prefetching or Caching in a Catch-Up TV Network. In *2016 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB)IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB).* [51]

Chapter 2

A Brief Introduction to DSL Systems

For the continued discussion, a brief introduction of the internals of Digital Subscriber Line (DSL) systems is called for.

2.1 From SNR to Transmission Rate

The DSL system is based on versions of Multi-Carrier Modulation (MCM)[1]. MCM means that the available signal bandwidth is divided into a number of evenly separated sub-carriers or tones, see Figure 2.1. In the DSL case, the tone separation is typically $\Delta F = 4.3125$ kHz[2]. Each tone is then modulated separately according to some modulation scheme. VDSL2 uses Quadrature Amplitude Modulation (QAM), and the constellation, and thus the number of bits per tone, normally up to 15, is dependent on the Signal to Noise Ratio (SNR) for that tone. This is referred to as *bit loading*. The bit loading can change dynamically if the SNR margin is decreased for a tone. This process called is called *bit swapping*.

A group of bits from a data stream are modulating the tones for a certain

[1]Digital Multi-Tone (DMT) or Orthogonal Frequency-Division Multiplexing (OFDM)

[2]Asymmetric Digital Subscriber Line (ADSL), Asymmetric Digital Subscriber Line, version 2+ (ADSL2+), Very-high-bit-rate Digital Subscriber Line (VDSL) and Very-high-bit-rate Digital Subscriber Line version 2 (VDSL2) all have tone spacing of 4.3125 kHz, except for the 30 MHz band-plan in VDSL2 which has 8.625 kHz tone spacing.

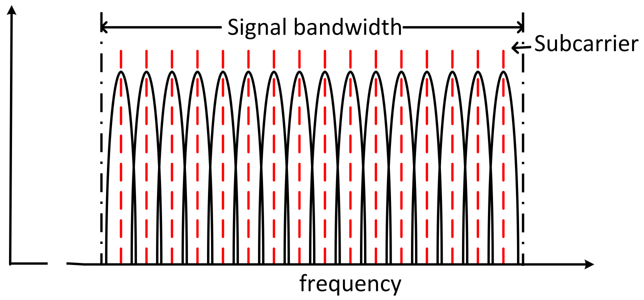


Figure 2.1: A schematic view of Multi-Carrier Modulation (MCM). The dashed lines indicate the centre frequency for each sub-carrier.

time, and then the modulation changes with the next group of bits. Thus, the transmitted signal is stable for a specified time, and this is referred to as a *frame*. The duration of a frame is dependent on the tone spacing, and is typically $250\mu s$.[3]

At the initialisation of a DSL connection, the network performance parameters such as transmission rate are adapted to the channel behaviour; Channel transmission rate depends on the bit loading, which in turns reflect the channel performance parameter SNR margin per tone. After a completed initialisation and in the active state, it is assumed that the channel performance is varying slowly, following e.g. the SNR level changes over the day. The adaptation to changes in the channel is therefore limited to a SNR margin, which typically is set to 6-9 Decibel (dB).

Figure 2.2 illustrates the bit loading process as being a function of the SNR margin and the maximum number of bits per tone. The top curve is the measured SNR. The SNR margin is then subtracted from the available SNR per tone. The remaining level is used for dimensioning the modulation level with a maximum of 15 bits per tone. As there is a bits per tone maximum in a QAM constellation, the actual SNR margin is higher than the configured minimum in the lower frequency band.

The Digital Subscriber Line Access Multiplexer (DSLAM) continuously controls the SNR margin for each tone. Bit swapping is used when a tone's SNR margin is decreased; the constellation for that tone has then to be changed and the number of bits allocated to that tone has to be decreased. If possible, bits are swapped to other tones where the SNR margin allows for more bits. The opposite does not happen when the SNR for one tone is increased; bits can only be swapped back to a tone if the SNR margin for other tones are reduced.

[3] A more detailed discussion is found in Section 6.2.

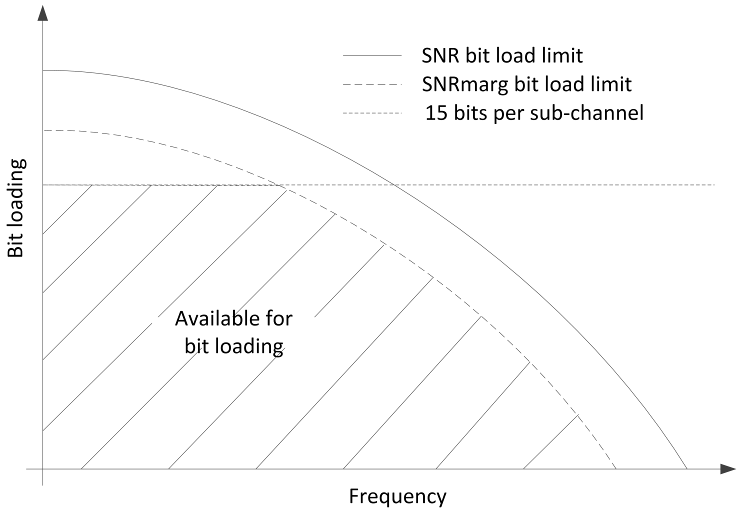


Figure 2.2: Bit loading as a function of the channel's SNR, SNR margin and a constellation limit of 15 bits per tone.

Even though the channel does not change in the same manner as a typical radio channel, these precautions are sometimes not enough. The dominating external disturbance in a DSL system is due to inductive crosstalk from other users, via electrical couplings in the cable bundle. Next to this, impulse noise typically generated in or close to the home environment where the Customer Premises Equipment (CPE) is a typical entry point, is a severe impairment for copper based access. A connection can be severely affected if a neighbour turns on their CPE and a new signal in the binder appears. Furthermore, there are radio station signals induced in the cables with varying strength over the day. Also, broken power adaptors can inject a square wave resulting in a comb-like spectra, just to mention a few typical disturbances.

Near-End Crosstalk (NEXT) and Far-End Crosstalk (FEXT), see FIGURE 2.3, need special attention in a DSL system. NEXT is the crosstalk from one transmitting line to a receiving line, where the transmitter and the affected receiver are at the same end of the binder. FEXT, on the other hand, is the impact that a transmitted signal has on the far end receivers for other lines in the binder. To get around the problem of NEXT, VDSL tones are grouped for up-link and down-link communication; The grouping is shown in Figure 2.4. In the VDSL2 standardisation [33] there is support for *vectoring*, which is a technique for mitigating the FEXT between users in the same cable binder. Other disturbances than FEXT will thus be more important to deal with.

In the central office equipment for the DSL connection, i.e. the DSLAM, typ-

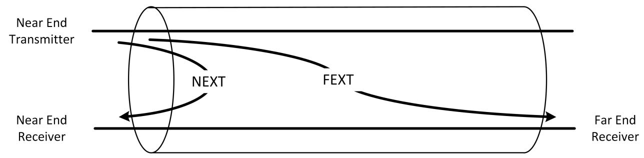


Figure 2.3: Illustration of NEXT and FEXT in a binder.

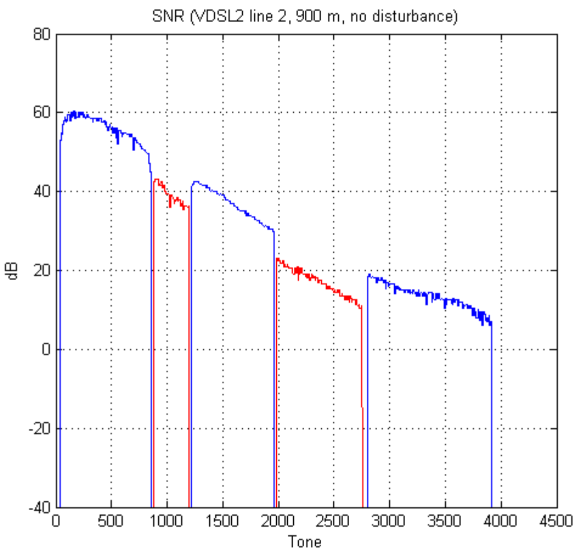


Figure 2.4: SNR per tone illustrating VDSL2's grouping of tones for the down-link (blue) and the up-link (red).

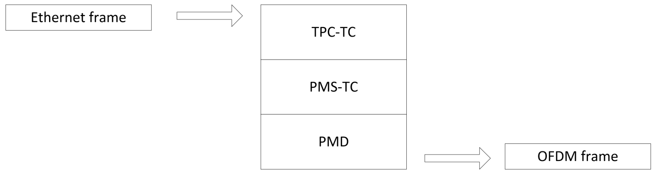


Figure 2.5: The functional blocks of a VDSL2 DSLAM. See sections 8 - 10 in [33] for details.

ical physical layer parameters can be read via SNMP. Among those the error conditions are indicated by the number of bit swapping occasions, Errored Seconds (ESs), Severely Errored Seconds (SESs) and Code Violations (CVs) during a time period.

Similarly, considering the path from the content server to the end user, obvious Quality of Service (QoS) parameters on the network layer are packet loss, latency, packet delay variations (PDV or jitter), inter packet arrival time and packet rate.

2.1.1 From Ethernet to VDSL2 OFDM Frames

For the understanding of the relation between the physical layer and the interface between the link and the network layers in DSL systems, deeper knowledge of the packet handling that takes place in a DSLAM is called for. The description will use a VDSL2 system with a single latency path and an Ethernet uplink as an example.

The transition from incoming Ethernet frames to bits loaded to sub-carriers of an Orthogonal Frequency-Division Multiplexing (OFDM) symbol can be briefly presented as in Figure 2.5. In the Transport Protocol Specific Transmission Convergence (TPS-TC) block an incoming Ethernet frame is re-framed into a Packet Transfer Mode (PTM) frame and multiplexed with idle bits before 64/65B encoding. The byte stream from the TPS-TC block is multiplexed with overhead, i.e. control data, and framed into OFDMs in the Packet Transfer Mode Transport Conversion (PTM-TC) block. Lastly, the Physical Media Dependent (PMD) block the data frames are encoded and modulated (Inverse Discrete Fourier Transform (IDFT)) into OFDM symbols.

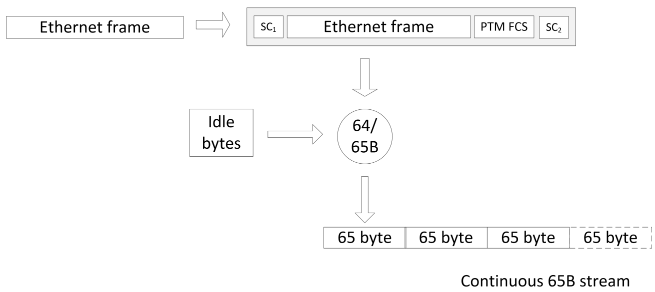


Figure 2.6: An Ethernet is framed into a PTM frame before 64/65B coding.

2.1.2 The TPS-TC Block

As an Ethernet frame enters a VDSL2 system it is re-framed into PTM frames, see Figure 2.6. Two or four Cyclic Redundancy Check (CRC) bytes are added dependent of Forward Error Correction (FEC) is used or not.[74] The 64/65B encoder adds one synchronisation byte to each 64 byte block of data. Additionally, two bytes are added to each PTM frame, one S-byte in the beginning and one C-byte at the end. The 64/65B encoding is also the place where the data stream is filled up with idle bytes to the configured data rate.

2.1.3 The PMS-TC Block

The 64/65B encoder outputs a continuous byte stream into one so called bearer channel as a purely binary stream, see Figure 2.7. A multiplexor combines overhead octets with bearer channel octets into Multiplexed Data Frames (MDFs). A number of MDFs makes one so called Overhead (OH) subframe. A number OH subframes are combined to one OH frame. One byte CRC is calculated over and added to each OH frame. Finally, a number of OH frames constitute one OH super-frame. [33]

The multiplexed byte stream is now input to a *scrambler*. The effect of the scrambler is that the first bit from the original Ethernet frame actually will contribute to all transferred bits. After the scrambler follows a Reed-Solomon (RS) encoder. A number of MDFs are carried in each RS codeword, so one OH sub-frame is carried in several RS codewords. After the RS encoder follows an optional interleaver for Impulse Noise Protection (INP).

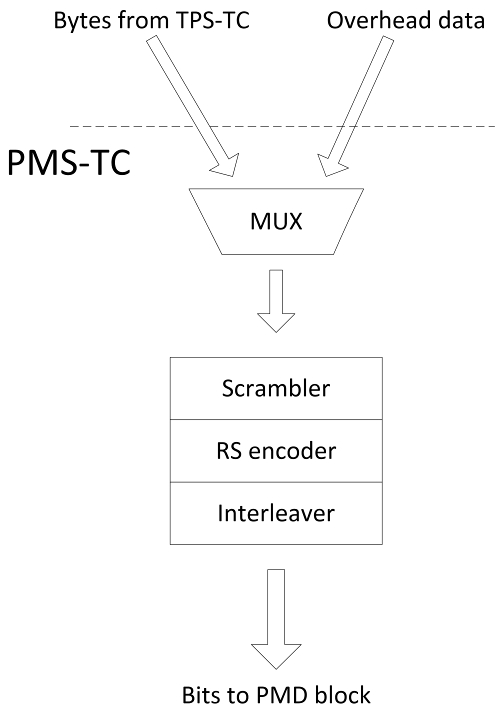


Figure 2.7: Overview of the PMS-TC functional block. The figure is simplified and shows only the condition used during the experiments. For the general case and details see section 9.1 in [33]

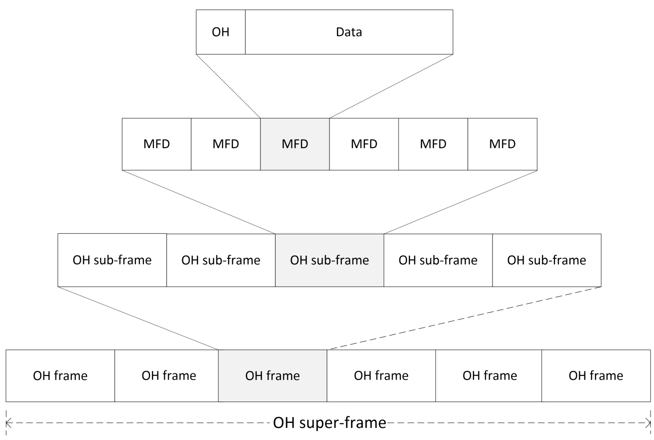


Figure 2.8: Framing of data and overhead. For details see section 9.5.2 in [33]

2.1.4 The PMD Block

In the PMD block, the OH super-frames are converted to OFDM frames. The bit stream is blocked according to the bit loading for the individual sub-carriers and modulated to signals. Here also a cyclic extension is added to prevent Inter Symbol Interference (ISI) due to the impulse response on the channel.

Chapter 3

Performance, Parameters and Monitoring

This thesis is built on work mainly concerning performance issues in Digital Subscriber Line (DSL) systems and their impact on the transport of Internet Protocol (IP) datagrams. Thus, the focus of this chapter is on performance parameters and monitoring thereof for these technologies. For comparison, Ethernet is also discussed in a minor section of the chapter.

Historically, the Internet has provided best-effort as the only service level[69]. Every service quality issue could be fixed by 'throwing bandwidth on the problem'. Today's users of Internet based services have higher and rightful demands when the applications are demanding high data rates and immediate response. These demands could be summarised as Quality of Service (QoS) and Quality of Experience (QoE) requirements. The definitions of QoS and QoE are rather vague and thus also their use and interpretation.

QoS parameters are available on all reference model layers. Thus, monitoring should be possible on all layers and along the full path from source to destination. But a network is very complex by its nature. Monitoring parameters in this environment by deploying measurement equipment on strategic positions is in most instances not feasible. Instead, monitoring has to be performed in-band and thus directly interfering with the traffic to be monitored. Acquiring parameter data from intermediate nodes in a path is only doable in domains under total control of the measuring instance.

Ethernet, as defined in IEEE 802.3 [5], is an example of layered packet switching in a pure form; A frame is transmitted bit by bit on the physical layer, one bit after the other. Normally one IP datagram is encapsulated in one Ethernet frame. DSL, on the other hand, is based on a combination of frequency and time division multiplex, where Multi-Carrier Modulation (MCM) frames are transmitted with regular time intervals. Bits forming an IP datagram are distributed over one or more MCM frames, and not necessarily in the order they are found in the datagram. An error in one Ethernet frame normally affects one IP datagram, while an error in one MCM frame can affect from zero to a couple of datagrams.

3.1 QoS, QoE, and Their Relations

QoS was first defined for telephony by International Telecommunication Union (ITU) in 1994 stating requirements for parameters like service response time, signal-to-noise ratio, cross-talk, and echo [8]. The concept QoS refer to a group of traffic characteristic parameters in the sense that these parameters reflect the quality with which traffic i.e. packet flows are distributed in the network. However, it can also refer to traffic engineering methods with the goal of fulfilling some level of quality for the traffic flow regarding these parameters. The current QoS can be estimated by monitoring, but it is also possible to define a level of QoS in for instance a Service Level Agreement (SLA).

The concept QoS is normally defining objective performance parameters and monitoring thereof at the lower layers, the physical layer to the network layer or even the transport layer, of the OSI reference model. QoE is very closely connected to the application layer and the user's subjective perception of the quality of which a service is presented with. QoE is purely end-to-end, meaning that the networks between source and destination can be seen as a black box; The resulting QoE is a function of the QoS measured along the path from source to destination, as shown in Figure 3.1.

3.1.1 QoS and QoE for IPTV and VoIP

TV over Internet (IPTV) and Voice over IP (VoIP) are examples of real time applications whose underlying layer QoS requirements differ significantly. A network that delivers IPTV with good QoE can deliver VoIP with low QoE. IPTV is a simplex application while VoIP is duplex. Not only is packet rate demands for VoIP orders of magnitude lower than for IPTV, but low latency is critical for VoIP while packet loss is the critical component for IPTV. Latency variations can, in the IPTV case, be compensated for by the use of *jitter buffers*.

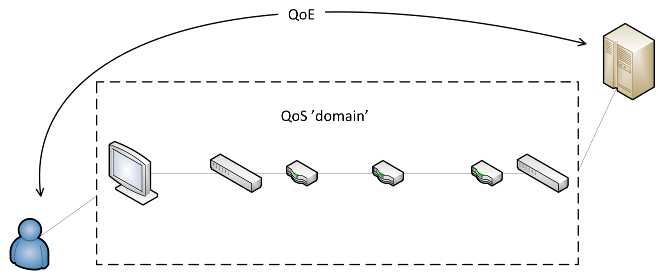


Figure 3.1: The scope of QoS and QoE.

In VoIP, latency is a well-known critical QoS parameter. Jitter buffers introduce latency in the order of hundreds of milliseconds to be effective and therefore cannot be used in this application. Packet loss is critical for the IPTV QoE, but it is acceptable in VoIP to a higher degree.

It is reasonable to believe that these relations between lower layer QoS and QoE are valid also in a mobile environment. Since the physical and link layers differ significantly between wired links and radio based links for mobile terminals, also the relation between QoS of the network layer and the two lower layers differ. In the case of mobile backhauling, where both types of applications are deployed by many users over a shared link, the two application groups dissimilar requirements is a challenge: Provide low packet loss and low latency at the same time.

3.1.2 Traffic Engineering QoS

The acronym QoS can address not only traffic parameters but also traffic engineering methods. QoS traffic engineering methods can be used for securing that the delivered service fulfils the required quality. There are several traffic engineering methods with the common goal of establishing a certain traffic quality level. In packet switched networks, techniques like leaky bucket and token bucket protect links and network nodes from overloading and the thereof following packet loss or delay. Tagging frames or packets according to different traffic classes enable nodes to prioritise individual packets/frames or flows of packets [11][17].

IP is by definition best-effort, meaning that there is no QoS. Integrated services (IntServ) [9] together with Resource Reservation Protocol (RSVP) [10] and Differentiated Services (DiffServ) [13] are two techniques to introduce QoS in this layer. IntServ reserves resources for a flow in the end-to-end path, while DiffServ tags packets on ingress and the nodes act on the tagging. MultiProtocol Layer

Switching (MPLS) [17] introduces prioritised switching on the network layer.

3.2 Passive and Active Monitoring

Passive and active monitoring techniques are discussed in the FP7 project COMBO[1] milestone M17 [41]. This is an important issue when discussing monitoring, especially regarding monitoring of networks. Many monitoring activities have to be done in an intrusive way, due to the fact that full control of the path end-to-end is difficult to accomplish.

In the context of network performance monitoring, passive could be defined as monitoring with no visible impact to or reaction from the monitored system. Also, passive monitoring could imply that the action is continuously ongoing; it is not activated for a special monitoring occasion. Active monitoring is then the opposite; monitoring triggered as response to a particular event or action.

Passive and active monitoring can also describe the amount of intrusion into the monitored system. Sending probe packets or streams on to the monitored network is one example of intrusion, the lookup of the whole or parts of a datagram another.

RFC 6632 [34] defines active monitoring as being monitoring of injected test traffic, while passive monitoring is only defined as being able to monitor without test traffic. It could be discussed how well-formed this definition is. A case where traffic is totally disrupted for the sake of monitoring would, according to RFC 6632, be classified as passive since no traffic is injected. [77] defines observations without disturbing the monitored traffic to be passive, while active is when probe packets are injected into the network.

The discussion can be summarised as a number of definitions: *Passive–Non-intrusive* monitoring has no effect on monitored elements. *Passive–Payload Intrusive* monitoring has no effect on monitored elements but investigates or visualises packet contents (payload and header). *Active–Link Intrusive* monitoring affects link performance. *Active–Service Intrusive* is monitoring of a selected service which affects that service alone. Finally *Active–Payload Intrusive* monitoring means manipulation of or addition to packet payload or header.

[1] COMBO - Convergence of fixed and Mobile Broadband access/aggregation networks

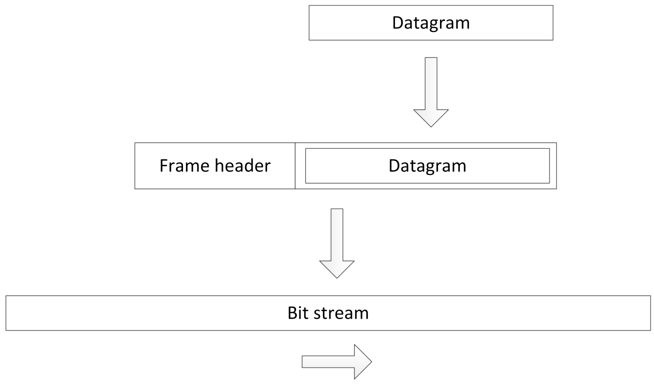


Figure 3.2: The relation between datagram, frame and bit stream. One datagram is encapsulated in one link layer frame and then transmitted as one bit stream.

3.3 Performance Parameters and Monitoring: Ethernet

In Ethernet based networks, one network datagram is normally encapsulated in one link layer frame, see Figure 3.2. A bit error on the link layer corresponds to one errored network datagram. Also frame delay variations on the link layer is transferred to Packet Delay Variations (PDVs) on the network layer.

IEEE 802.3 [5] describes various QoS parameters and monitoring methods for the physical layer. For 100BASE-X a Link Monitor is specified, which states if the link status is OK or not. 1000BASE-T is more delicate; Auto negotiation and master and slave clock synchronisation introduces a more defined level of QoS monitoring. 10GBASE-T physical layer status can be checked because of a defined Physical Control method. Bit error rate is a physical layer QoS parameter that is common for all versions.

Important Ethernet QoS parameters on the link layer are availability, frame error or loss rate, frame delay, and frame delay variations. Even though Remote Network Monitoring (RMON) Information Base [16][21] is more aimed at higher layers, it is applicable to Ethernet monitoring. Internet Engineering Task Force (IETF) also defines in RFC 2819 [16] an list for QoS parameters.

QoS parameters can be monitored by invoking intermediate nodes. Tools used for IP performance monitoring, see Section 3.5.2, reflects Ethernet QoS status.

3.4 Performance Parameters and Monitoring: DSL Systems

DSL transport of packet data is much more complicated than for Ethernet, see Chapter 2. It also breaks the close to linear relation between the physical layer and upper layers that exist in for example Ethernet based networks.

3.4.1 DSL QoS Parameters and Monitoring

DSL standards define a vast number of QoS parameters, most of them related to the physical layer. For example, the number of Code Violations (CVs) and Maximum Attainable Data Rate tracks line condition changes during showtime. Errored Second (ES) and Severely Errored Second (SES) counts the number of seconds with one or more CV (ES) or 18 or more CVs (SES). If physical retransmission is activated, the Forward Error Correction counter counts the number of corrected code words.

Testing of individual lines can be performed with or without termination in the far end. The first test method is called Dual Ended Line Test (DELT) and the latter Single Ended Line Test (SELT). The Digital Subscriber Line Access Multiplexer (DSLAM) system used in our lab incorporates possibilities to perform both types. To perform a DELT, a Customer Premises Equipment (CPE) has to be connected to the far end of the line. This is because in DELT, the line is tested from both ends, while in SELT all measurements are performed from the DSLAM end.

DSL is focused on residential applications. The lines behave individually from performance point of view. Thus, DSL management systems are geared toward single line monitoring and control.

3.5 IP Performance Monitoring and Parameters

Internet Protocol version 4 (IPv4)[78] is by far the most used Internet protocol of today. But Internet Protocol version 6 (IPv6)[12] is gaining momentum. IPv4 has support for QoS traffic management but this is rarely used. IPv6 has stronger inbuilt QoS traffic engineering support, e.g. flow identification and prioritisation, and it could thus be discussed if the two versions have to be treated separately from a performance monitor perspective. Controlling IPv6 router advertisement and neighbour discovery might be a means of controlling paths and thus indirectly influencing QoS performance.

Given an end-to-end path consisting of several networks with diverting link layer technologies, the network layer is the first layer where real end-to-end measurements can be accomplished. This layer is also the lowest layer that is fully available for performance measurements for all involved parties, that is content providers, all connectivity providers and end users. Therefore the first approximation is to only consider monitoring of nodes working on the network layer. Also, events or incidents on the link layer most likely have an influence on the network layer and thus affect QoS parameters on this layer.

In a network where network layer routing is focused on establishing a single working bi-directional path between any two end-points, it should be straight forward to identify and measure performance per flow. When the routing process in such a network has converged, it is also fairly stable. Thus, it is a simple and possible task to identify the nodes involved in the flow transportation, and from them acquire the performance metrics. With the deployment of hot potato routing[2], see Figure 3.3, or load sharing or load balancing over equal cost paths, the situation becomes somewhat cumbersome. Traffic flows are no longer symmetric, but still in some sense deterministic.

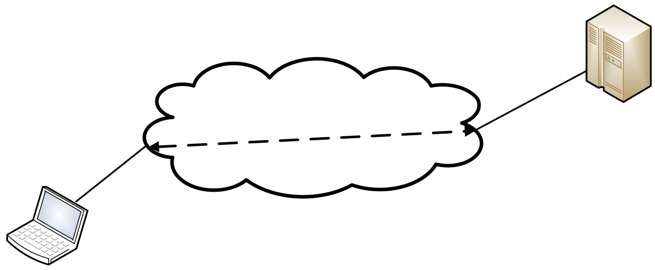
Another noticeable aspect is the trend towards all-IP networks; IP is used not only on strict layer 3, i.e. the network layer, but also what could be denoted layer 2.5, for tunnelling of user data (e.g. the GPRS Tunnelling Protocol (GTP) in mobile core networks), tunnelling of signalling and lower lay, Interestingly, the value of IP measurements, specifically TCP testing, is under debate. Bagnulo et al defines some issues that have an impact on the reliability of the measurements: the lack of standardisation regarding measurement platforms, methods and metrics, different TCP implementations, the definition of which TCP parameter(s) that influences the users QoE. [55]

Due to the lack of proper flow identification in the IPv4 header it is difficult to measure flow specific parameters other than with active intrusive methods without relying on information from higher layers' headers.

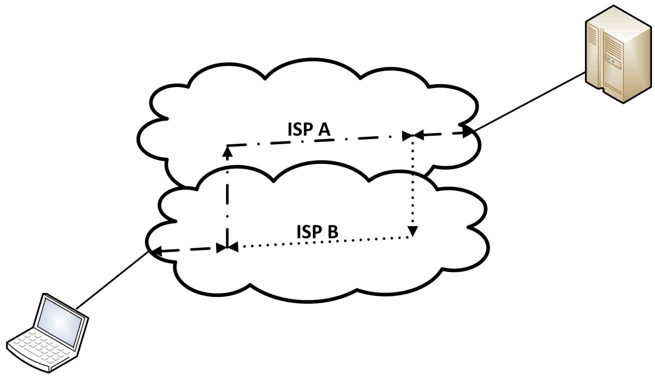
3.5.1 Network Layer QoS Parameters

Packet loss and *delay* or *latency* are found among the obvious QoS parameters in packet switching networks. Packet Delay Variation (PDV), also named *jitter*, inter-arrival time and packet rate could also be included among obvious

[2]Route traffic out of own network as soon as possible.



(a) Symmetric packet flow. Packets in a bi-directional flow follow the same path, independent of direction



(b) Asymmetric packet flow. Packets in a bi-directional flow take different paths dependent on direction. This is called Hot Potato Routing

Figure 3.3: Illustration of symmetric versus asymmetric packet flow.

parameters. Other QoS parameters address the level of redundancy in the end-to-end path. Existence of active equal cost paths, switch-over time and routing protocols used can be included in this group.

There exist other, less obvious, QoS parameters like bit rate, burstiness and Maximum Transmission Unit (MTU). The characteristics of cross traffic could also be included here.

Considering strictly IP, all parameters that depend on time-stamping are rolled out, due to the lack of a header field for time stamping.

It should be noted that, for foremost real time media applications, PDV above some threshold as well as out-of-sequence packets could be interpreted as packet losses; Packets arriving too late for the real time play out are useless.

Special interest groups and fora have defined QoS metrics, both generic and of special interest for the specific technologies. The IP Performance Metrics (IPPM) working group within IETF has defined performance measurements parameters. In the Internet Draft "An Overview of Operations, Administration, and Maintenance (OAM) Mechanism" [90] the authors present an overview of OAM mechanisms that are defined by the IETF.

Broadband Forum defines attributes and metrics for performance measurements as well as a test architecture in [38]. ITU Telecommunication Standardization Sector (ITU-T) has defined metrics and methods for packet delivery and end-to-end performance.

3.5.2 Measuring the Parameters

Most of the parameters discussed in Section 3.5.1 can be extracted from the intermediate nodes. Only a few of them are measurable with passive non-intrusive methods without direct access to intermediate or end nodes. For most parameters, active intrusive methods have to be deployed.

The generic parameters are in a sense implicitly defined for the full path between the hosts involved. But they could of course also be defined per link or per flow. The straight-forward way to measure them is by applying methods described in 3.5.2 for the full path end-to-end. Since interfaces on intermediate network layer nodes are obligated to act as network layer hosts, these methods could also be applied for measuring links between nodes, given that you have access to the nodes or agents in the nodes involved.

Measuring QoS parameters for all flows between two hosts would require IP header inspection, using the IP address pair and the protocol field. Transport level header inspection is needed to identify a specific flow between the two hosts. The IPv6 header contains a flow label field, which might be enough for singular flow identification.

Tools

Common and well-known tools for monitoring paths end-to-end are *ping* and *traceroute*. Ping is an application that utilises ICMP [4][19] messages echo request and echo reply for checking end-to-end connectivity, but also returns information about packet loss and round-trip time. By manipulating the time-to-live field in the IP header, the tool traceroute can force nodes on the path to the remote host to respond with Internet Control Message Protocol (ICMP) Time Exceeded messages, thus reporting the path that packets take from the local host to the remote host. Ping and traceroute can be used for IPv4 and IPv6, because Internet Control Message Protocol version 6 (ICMPv6) contains all three messages echo request, echo reply and time exceeded.

The Simple Network Management Protocol (SNMP)[6][7] is a well-known and widely spread protocol for monitoring and management of both network nodes and content services. Remote Network Monitoring (RMON) [18] extends SNMP's functionality. While SNMP relies on agents in network nodes (agents that only can handle node-related parameters) and deploys device-based monitoring, RMON allows special nodes to monitor flows and then provides flow-based monitoring. RMON also opens up for monitoring on layers above typically the physical and link layers, including the transport layer and the application layer. RMON uses SNMP for data collection. Remote Monitoring for Switched Network (SMON) is defined in [39] and defines an extension of RMON for switched networks.

Cisco's NetFlow [100] uses measuring agents built into network nodes. These agents have the ability to identify flows using ingress interface, IP address pairs, Transport Control Protocol (TCP)/User Datagram Protocol (UDP) port pairs, IP protocol and IP Type of Service. This information is reported to a NetFlow Collector from where data can be fetched for analysis. The time of flow start and stop, number of packets and bytes in a flow, and more can be presented.

sFlow [98] relies on sampling of packets for monitoring high speed switched networks. Not all packets are sampled and not all data in the sampled packet is stored, but the sample rate is enough for reliable data and accuracy. Interface and system counters are also sampled periodically and sent to the data collector.

A more recent approach that can replace RFC 2679 [14] and RFC 2680 [15] is the RFC 4656 One-way Active Measurement Protocol (OWAMP) [22]. OWAMP describes methods for obtaining unidirectional characteristics by actively injecting network layer datagrams to characterize the network performance such as one-way delay and one-way packet loss. Since the measurement is one-way a session-receiver has to be deployed in the far end of the path. Also a fetch client is required to gather data.

For bi-directional monitoring OWAMP could be used in both directions. But for monitoring round trip time and other two-way metrics, Two-way Active Measurement Protocol (TWAMP) [27] could be deployed. TWAMP is an extension of OWAMP and is thus also based upon active data injection. The difference compared to OWAMP is that instead of the session-receiver it utilizes a so called session-reflector. In a typical measurement scenario, the session-sender injects packets into the network destined to the session-reflector which then sends the packet back to the sender as soon as possible. The packet is time stamped by both the nodes for both departure and arrival, and each packet is thus time stamped four times for a successful measurement. With this procedure, it is possible to measure metrics such as round-trip time, PDV and packet loss for the path between the sender and the reflector.

TWAMP as described in RFC 5357 is not capable of measuring metrics related to network capacity in both directions, since such measurements typically requires functionality for setting the packet-train rate. Using the standard TWAMP, it is only possible to measure capacity in the forward direction. A feature for capacity measurements in both directions is added in RFC 6802 "Ericsson Two-Way Active Measurement Protocol Value-Added Octets" [35], where the reflector is extended with a buffer that can store the packets and then send them back with a new configurable rate.

A challenge with using TWAMP is where to place the senders and reflectors to get a good characterization of the network. Since TWAMP is an active measurement method that inject packets into the network, it is crucial to consider scalability so that the measurements themselves are not congesting the network. Another risk if many measurement paths overlap is that the measurements may interfere with each other and reduce the reliability of the estimated metrics.

Bidirectional Forwarding Detection (BFD), RFC 5880[31], is a protocol aimed at detecting faults in bidirectional paths between two systems or hosts. The protocol relies on hello-like packets sent between two systems. If these packets are not received, the path is declared down. RFC 5881 [32] defines the special considerations that have to be applied to BFD for the use in IPv4 or IPv6 environment.

Chapter 4

The Laboratory, and Experimental Practicalities

In Chapter 3, a general discussion of performance parameters and monitoring of these parameters was presented. In this chapter, more practical considerations concerning monitoring performance parameters in a lab environment are presented. Our lab setup and how to create and induce controlled disturbances into the Digital Subscriber Line (DSL) system is described. Issues of monitoring packet transmission performance are discussed. Finally, some traffic generator/analysis packets are studied.

Our work has been oriented towards cross-layer performance issues, in this case how the transport of Internet Protocol (IP) datagrams over DSL systems is affected by impairments on the physical DSL line. In order to study this, a lab facility was established in collaboration with Ericsson Research, Kista.

The overall goal for the lab was to perform measurements on the IP level over a DSL link. One aim was to find out if it is feasible, and in the case to what degree it is feasible, to estimate the dynamic characteristics of an access link by active or passive monitoring - or a combination of both - on the IP layer. Another aim was to study how disturbances on the physical DSLlink impact the IP Quality of Service (QoS) parameters.

Obviously, it is important to create an experiment environment that is not disturbed by any external source. It is also of great importance to control the impact of network equipment in the path or link under test. Since normal Personal Computers (PCs) were used for generating and capturing IP traffic, special

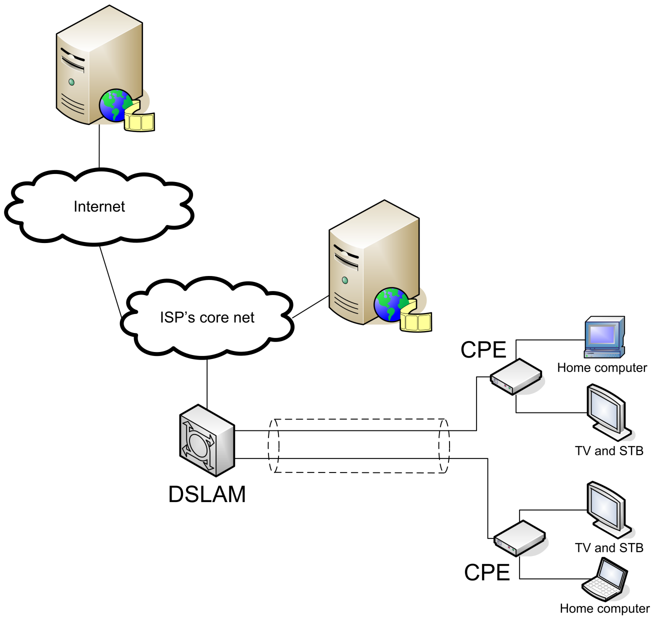


Figure 4.1: A schematic of the testbed layout

focus had to be given to the impact of used network interface cards NIC and operating system's kernels to time domain measurements.

A problem, that Ahlgren recognizes [48], is how to handle re-ordered packets in a data stream. Ahlgren suggests that the best method is to use the received packet order in all calculations. In our experiments, reordering was not a concern, due to the simple network setup.

4.1 The Testbed

The testbed, see Figure 4.1, was built around a DSL central office system with Digital Subscriber Line Access Multiplexers (DSLAMs) for Asymmetric Digital Subscriber Line, version 2+ (ADSL2+) and Very-high-bit-rate Digital Subscriber Line version 2 (VDSL2). Physical cable connections stretching over several kilometres were deployed using multiple pair landline copper cables on cable drums. A number of Customer Premises Equipments (CPEs)[1] to which a customer Local Area Network (LAN) can be connected, terminated the DSL connections. In the central office, termination point IP datagram sources of

[1] Often called *modem*

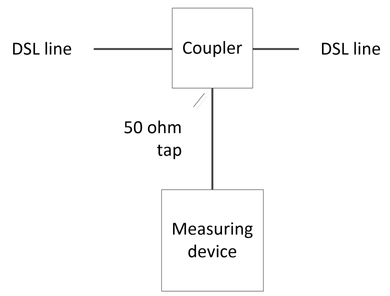


Figure 4.2: A three-way coupler.

different kinds were deployed. Controlled electrical disturbances were induced in the DSL link. How services on any level in the Open Systems Interconnection (OSI) reference model were affected by the generated impairments was analysed.

Three way passive couplers, connected in the copper drop connection as seen in Figure 4.2, were used for inducing disturbances or for measuring the physical signals on the drop line. The two through ports were adapted to the drop lines impedance 100 Ω, adding minimal dampening, while the third connector, a 50 Ω tap, was used for connecting laboratory instruments.

The DSLAM system in the testbed hosted a management and monitoring system giving access to a vast amount of both long term and short term parameters and monitoring functions. These were accessible from MATLAB® scripts issuing Simple Network Management Protocol (SNMP) calls.

4.2 Inducing Disturbances

Two types of disturbances have been of special interest for this work: Broadcast radio signals and bursts of impulse noise. [2]

4.2.1 Modelling Disturbances from Radio Stations

Radio Frequency Interference (RFI) originates from a wide variety of sources, not only Amplitude Modulation (AM) and amateur radio stations but also badly shielded electrical devices and switched power supplies with worn out capacitors.

[2] Chapter 1 discusses sources of disturbance that affect DSL systems.

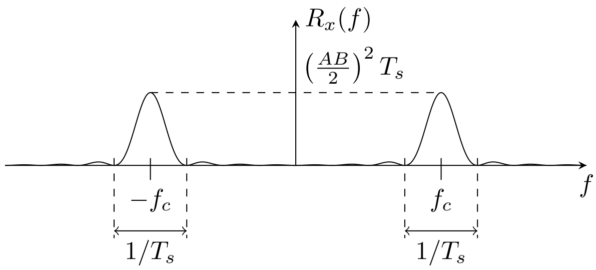


Figure 4.3: Power spectral density of RF model.

The external interference will induce a signal in a badly balanced twisted pair that will decrease Signal to Noise Ratio (SNR) for one or a couple of neighbouring DSL tones.

The frequency spectrum of interest for Asymmetric Digital Subscriber Line (ADSL) and Asymmetric Digital Subscriber Line, version 2 (ADSL2) systems is 0-2.2 MHz and for VDSL2 0-30 MHz. Switched power supplies, which is a known impairment factor, have a switch frequency of typically 400 kHz, and AM broadcast stations are typically found in the spectrum 1-1.5 MHz. Both these sources of disturbance coincide with the DSL frequency spectrum.

One of the disturbance sources that were analysed was a simulated broadcast radio signal. The disturbance used was a Pulse Amplitude Modulation (PAM) sinus wave. This type of signal has similar power spectrum as that of a broadband radio station. In the equation

$$x(t) = \sum_{k=0}^{\infty} A_k \cdot g(t - kT_s) \cdot Bcos(2\pi f_c t) \tag{4.1}$$

$A_k \cdot g(t - kT_s)$ constitutes a bipolar bit stream with the amplitude shifting between $-A$ and $+A$ randomly. f_c is the centre frequency or the carrier frequency, as can be seen in Figure 4.3 where the power spectrum density of $x(t)$ is shown. The width of the centre lobe is determined by the bit rate of the square wave. The signal power at f_c is a function of A^2, B^2 and the bit time T_s. Thus, it was possible to control where in the DSL spectrum the disturbance (f_c) was placed, how wide the centre lobe was (the bit rate of the bit stream) and the power of the centre lobe (the amplitude of the bit stream and/or the carrier signal).

In the lab, this type of signals was generated by letting a random bit stream control the amplitude, $-A$ or $+A$ in (4.1), of the baseband signal. This signal was then used to amplitude modulate a carrier signal in a Vector Signal Generator.

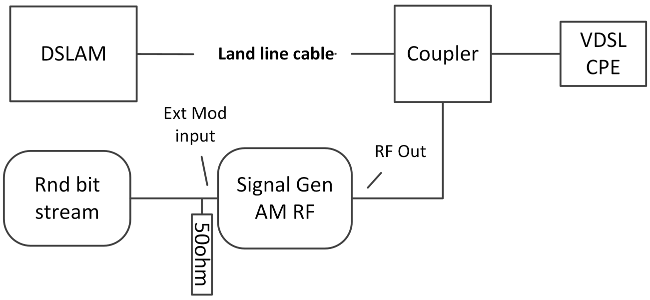


Figure 4.4: Schematic view of the lab setup during narrow band disturbance experiments.

The resulting narrow band signal, $x(t)$ in (4.1), was eventually applied to the DSL line under test according to Figure 4.4. Figure 4.5 shows the resulting SNR and bit loading pattern per ADSL2+ tones of an interfering signal with f_c set to 1 Mhz and the average bit rate set to 10 kbps.

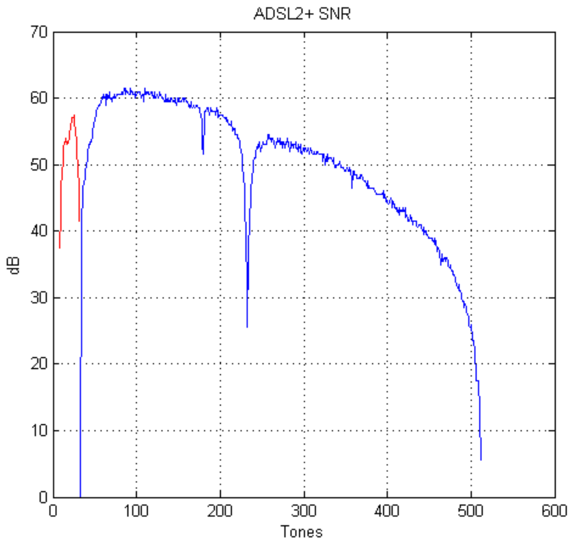
4.2.2 Modelling Impulse Disturbances

Impulse Noise (IN) are short burst or impulses of disturbances. The average SNR is not significantly affected by a single Electrical Impulse Noise (EIN) burst, because the lengths of the bursts in the time domain are short. But due to the bursts' high energy and broad frequency spectrum, a DSL link's stability is decreased. The typical entry point for EIN into a DSL system is via the CPE.

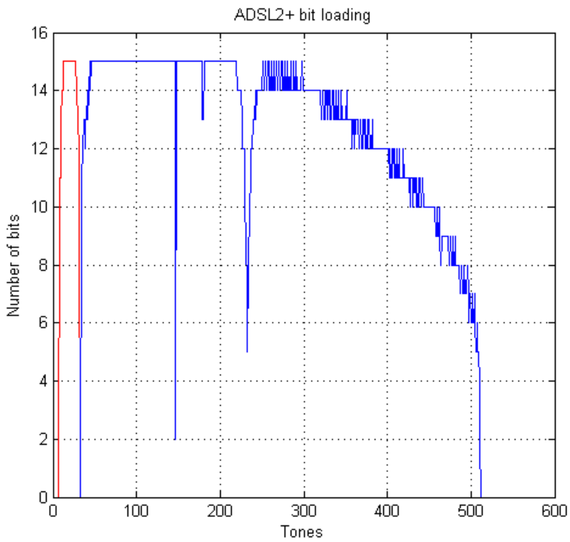
Repetitive Electrical Impulse Noise (REIN) is noise bursts with a fixed inter-arrival time of typically twice the period of that of the main power distribution. The burst length is rather short, in the order of $100\mu s$, which corresponds to one or two DSL symbols. Due to the repetitiveness this type of noise can generate a lot of bit errors, but the system's Impulse Noise Protection (INP) can alleviate the situation.

Prolonged Electrical Impulse Noise (PEIN) and Single High Level Impulse Noise Events (SHINE) also have a negative impact on the DSL link. It is very hard to protect the DSL transmission against SHINE due to the extent burst duration.

To model impulse disturbances, a noise generator, generating close to white noise in the frequency range 10 kHz to 80 MHz, was used. The noise generator had a fast Radio Frequency (RF) switch, used as a gate, connected to its output. The gate was in turn controlled by a monostable multi-vibrator that was triggered by a micro-controller. The testbed setup is shown in Figure 4.6.



(a) SNR per tone.



(b) Bit loading per tone. The dip at tone 147 is the synch tone.

Figure 4.5: SNR and bit loading per tone for an ADSL2+ line with an interfering signal at $f_c = 1Mhz$ and the average bitrate $= 10kbps$. Blue is the down-link, red the up-link.

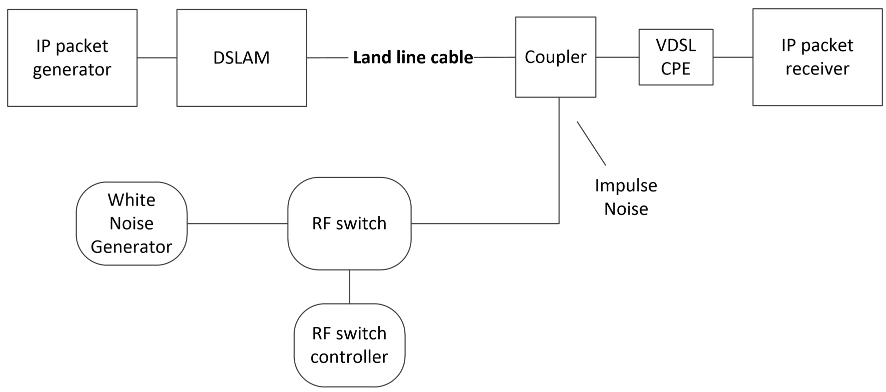


Figure 4.6: Testbed used for impulse disturbance experiments

Impulse disturbances were injected in the physical link using a passive coupler. Pulses of impulse disturbance of chosen length and frequency could thus be generated and injected at a chosen position on the cable. The position of the passive coupler in relation to the energy in the noise pulse is of importance due to the dampening of the signal along the link. In the downstream link, a signal is much more attenuated near the CPE than in the central office; The same disturbance pulse has more impact close to the CPE than close to the DSLAM.

4.3 IP Performance Issues

One objective was, as said in the introduction to this chapter, to examine the relationship between a DSL link and IP traffic carried by that link. The measuring of typical network layer parameters, such as packet loss, delay and Packet Delay Variation (PDV), and the time synchronisation of these measurements with events in the lower layers are of interest.

IP performance between end hosts are normally of most interest. Detection of packet loss only need IP datagrams to be numbered in sequence. Latency, or derivations thereof, call for time-stamping of the IP datagrams. In two-way measurements one host acts as both transmitter and receiver and the other just like a reflector, and there is no problem to establish the required time synchronisation between the two entities; They are relying on the same system clock. One-way measurements require elaborate time synchronisation between the transmitter and the receiver.

ADSL and Very-high-bit-rate Digital Subscriber Line (VDSL) links are by defin-

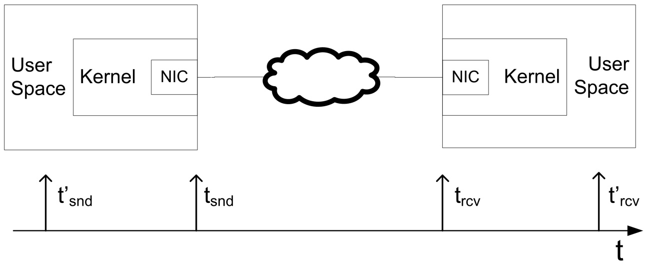


Figure 4.7: The time domain. Time stamping was done at time t'_{snd} and t'_{rcv}, while the requested times were t_{snd} and t_{rcv}.

ition asymmetric, the downlink has higher bandwidth than the uplink. Therefore, network layer one-way measurements are preferred. This in turn stresses the problem of time synchronisation. It is essential for one-way delay and PDV measurements that the time stamping of packets at both sender and receiver side correlate well to absolute time. PDV, on the other hand, can be estimated by monitoring Inter Packet Gap (IPG) at the receiver side. In the experiments performed, only time differences between events in one tool were measured, thus no need for time synchronization.

4.3.1 Time-stamping

When using common PCs as measuring devices, special care must be taken regarding where, in the path from measurement application to sending of the link layer frame by the Network Interface Card (NIC), the time-stamping is physically performed. NICs adapted for time-stamping exist, but are expensive. The Endance DAG®card is one example of such an interface [67]. If common NICs are used, time-stamping can take place in the Operating System (OS) kernel or in the application; The former is preferred, since it is more accurate. An alternative is to execute a time-stamping application in kernel space, instead of the normal user space. An application that can use an OS kernel's real time support is favourable.

Time-stamping in user space does not give correct results for experiments aimed at measuring temporal characteristics in the network path. Figure 4.7 describes this. The actual time wanted at the sender side is the time when a packet is sent out by the NIC, t_{snd}, but the time stamping is done at time t'_{snd}. Likewise, at the receiver side the actual time wanted is t_{rcv}, but the time stamping is done at time t'_{rcv}.

An experiment was conducted to examine how the time-stamping in the user space, see Figure 4.7, affected PDV measurement. A continues 3 Mbps stream of equal sized (1400 bytes per datagram) and equal time spaced User Datagram Protocol (UDP) datagrams was sent from one computer to another over an Ethernet link. The IPG at the receiver side was registered and plotted, see Figure 4.8(a). As can be seen, the variance in $t'_{rcv} - t_{rcv}$ is high, influenced by the kernel and other applications.

4.3.2 Time Synchronisation

Due to the cross layer environment of the experiments, several monitoring devices working on the layers of interest, were used. Disturbances were added on top of the DSL signalling level, and the performance on DSL symbol level were studied. On the IP level, monitoring traffic was generated, in general with the addition of time-stamping of individual datagrams. This made events and performance monitoring on all levels synchronised in time.

The DSL monitoring system is based on cumulation of number of events per time unit. It is not possible to time stamp e.g. a single event like the reception of a DSL symbol with a Code Violation (CV), i.e. a Cyclic Redundancy Check (CRC) error. Instead, the number of CVs are accumulated and the cumulated sum could be read out from the system as fast as the monitoring system allowed, approximately each third second in the used system.

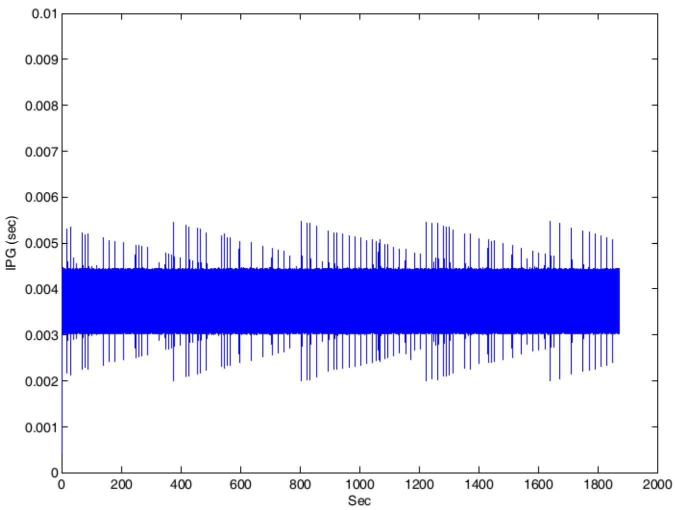
Time-stamping of IP datagrams in a standard personal computer depends on many different factors, as discussed in Section 4.3.1 in this chapter. Empirical findings showed that the drift of a lab PC's system clocks was too great if the experiment last for a longer time, in these studies 1 to 2 hours.

A study of time stamping accuracy in virtual environments was presented at ICACT 2011. [92]

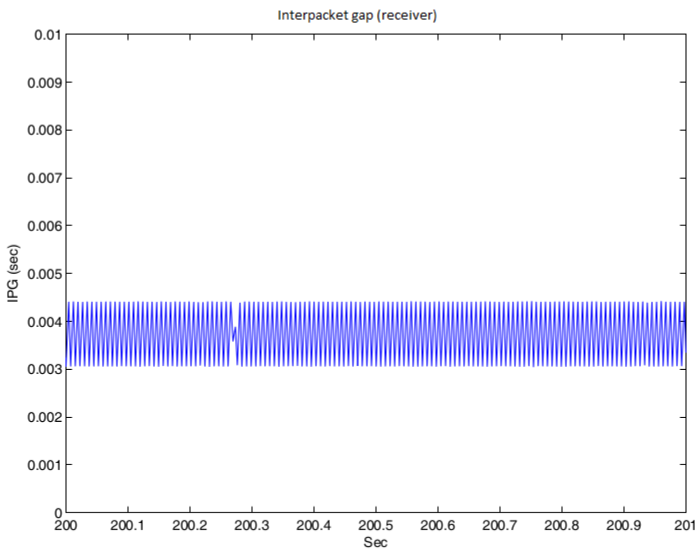
4.3.3 Time Domain Background Noise

Background noise in measurements can in general be tolerable if the SNR is high. The background noise, when measuring IP datagram latency and PDV, is located in the time domain. The magnitude of the noise is the added packet delay introduced by the application, the OS including the NIC driver and the NIC itself. The measured signal is, in this case, a function of the shift in time introduced by events on the DSL link.

The background noise and the oscillations, as seen in Figure 4.8(a) and Figure



(a) IPG at the receiver side of an Ethernet link.



(b) Zooming in on IPG of an Ethernet link.

Figure 4.8: IPG at the receiver side.

4.8, can emerge anywhere in the path from the sending application to the receiving application. The OS kernel scheduler does not maintain a deterministic and equal time spacing between creating and time stamping one packet and the next, nor does the kernel secure that the packets are transmitted out onto the link immediately from the application without further delay.

An interpretation of the PDV oscillations in Figure 4.8(b) suggests that packets are joined together two and two. But the first (or the last) packet in each group has kept the original time space to the first (or last) packet in the next group. The proposed source of this artefact is the scheduling mechanism in the used OS.

4.4 Packet Generators and Analysers

There are several packet generators/analysers available. For the selection of a viable IP traffic generator and analyser package, some requirements were listed. The packet generator must be able to generate IP packets carrying UDP and TCP segments. Packets must be time-stamped and have sequence numbers. Packets must be generated either periodically and deterministic or with varying inter packet separation according to some given distribution. It must also be possible to generate trains of packets. Inter packet separation in this case is deterministic, but it should be possible to set the start of the train according to a given distribution.

A selection of packet generators/analysers are listed below. It should be noted that the listed tools do not meet all requirements.

Wireshark and tcpdump

Wireshark [108] can time-stamp incoming or sniffed packets in real time and store them for offline evaluation, but it has to be considered that the library libpcap [3], that Wireshark as well as tcpdump [80] uses, executes in user space.

D-ITG

D-ITG (Distributed Internet Traffic Generator) [1] is a platform capable to produce traffic at packet level accurately replicating appropriate stochastic processes for both inter departure time and packet size: Various distributions (i.e. exponential, uniform, cauchy, normal, pareto) can be used. D-ITG supports both IPv4 and IPv6 traffic generation and it is capable to generate traffic at network, transport, and application layer.

RUDE/CRUDE

RUDE/CRUDE [86] is a packet generator and analyser like D-ITG, but it can only generate UDP packets. The traffic generation process, time-stamping and sequence numbering can be controlled. According to [62] this application has good precision in the time stamping compared to other generators/analysers executing in user space. Both the RUDE and the CRUDE applications offer the possibility to set the real-time priority. But the applications are from 2002, and since then the Linux scheduler has undergone major changes.

Ostinato

Ostinato [96] is a fairly recent packet generator/analyser. It is a client/server application where several servers, the application that actually performs the generations and measurements, can be controlled remotely by a client. Both server and client executes in user space and therefore could be suspected to perform not differently from D-ITG and RUDE/CRUDE. Another negative behaviour is that the server needs a running X window system to execute.

KUTE

KUTE [112] is a kernel based traffic generator and receiver. KUTE generates UDP traffic. Zander *et al.* have compared KUTE with RUDE/CRUDE and shown that the standard deviation for KUTE is much lesser than for RUDE/CRUDE [111].

NTools

NTools, provided by N. Vegh [104], belongs to the group of applications that make use of the Linux kernel's real time functionality. NTools includes a packet generator and a packet receiver. Packets are time-stamped when sent and when received. Several sending profiles are available. NTools gives a higher accuracy in time stamping compared to RUDE/CRUDE. This is most probably due to the added RT functionality of the kernel.

Chapter 5

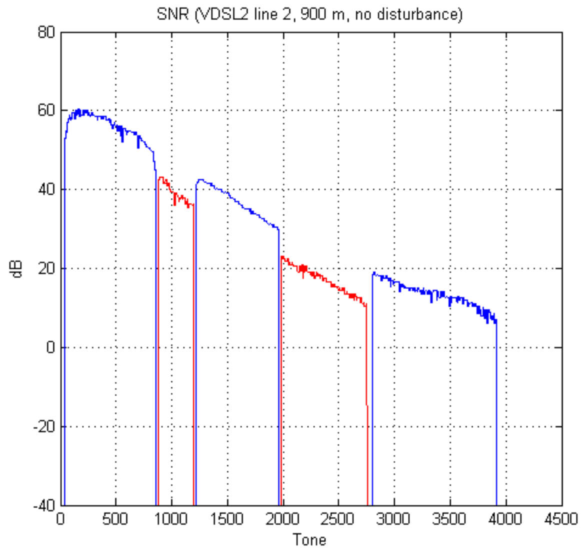
Broadcast Radio Disturbance's Impact on DSL and IP

In this chapter the impact of narrow band disturbances on Digital Subscriber Line (DSL) bit swapping and Internet Protocol (IP) Quality of Service (QoS) is discussed. Radio stations operating on the Medium wave (MW) band is a typical source of this type of interference. Another noticeable source of interference is faulty power adapters [94]. Results discussed in this chapter were presented at SNCNW 2012 [52].

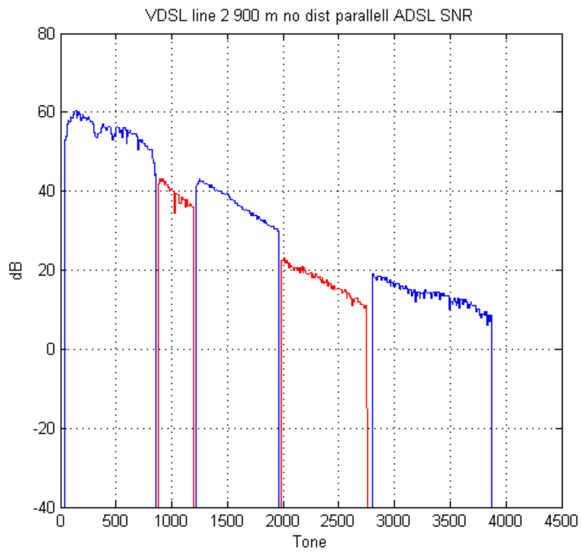
5.1 Narrow Band Interference on DSL

Disturbances and noise in a DSL environment can have many sources. More or less continuous radio signals can have a negative influence on a DSL link. Poor wiring of the Public Switched Telephone Network (PSTN) network is also known to affect a DSL link negatively. Impulse noise from e.g. the power line is another example. Depending on the power and the duration of the disturbance, the result can be a decrease in capacity, a short interruption of the packet stream, or in worst case a total retraining sequence.

Bit swapping is done when a tone's SNR margin is decreased. No restoring of bits is performed, even if the SNR margin is increased. Thus, a short disturbance



(a) VDSL2 DELT with no disturbance.



(b) VDSL2 DELT with ADSL2+ in parallel pair.

Figure 5.1: Reference measurements of Signal to Noise Ratio (SNR).

impulse might influence the link's capacity for a long time.

Error conditions in DSL systems are indicated by the number of bit swapping occasions, Errored Second (ES) counted and the number of Cyclic Redundancy Check (CRC) errors during a time period. Of course there are more error indicators, but these are sufficient for the current discussion. An ES is a one second time frame during which at least one error - CRC error, Loss of Signal or Severely Errored Frame - in the transmission has occurred.

5.2 Disturbance from Radio Stations

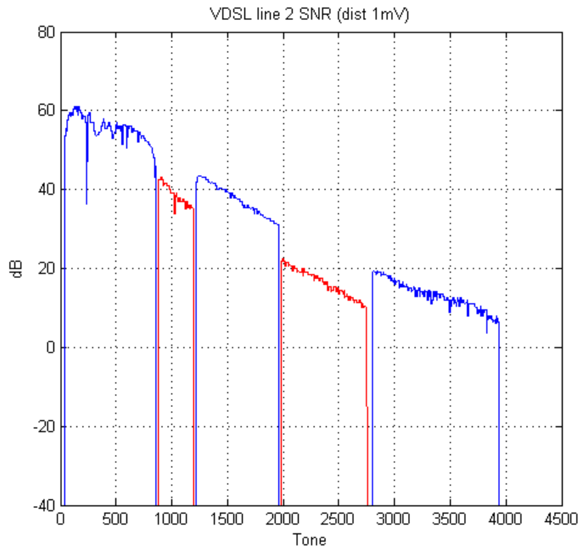
Earlier studies performed by the research group include trying to identify error conditions by evaluating patterns in DSL lines' status, so called *fingerprinting* [103]. But this approach is only available for a DSL system operator. For example, third party content providers have no other option than IP QoS measurements.

To get an understanding of the impact of a Amplitude Modulation (AM) radio station like disturbance on a packet stream over VDSL2 links, a User Datagram Protocol (UDP) packet sender and receiver was connected to the Digital Subscriber Line Access Multiplexer (DSLAM) and the Customer Premises Equipment (CPE) respectively. For the experiments, the lab configuration and the disturbing signal discussed in Section 4.2.1 was used, see Figure 4.4.

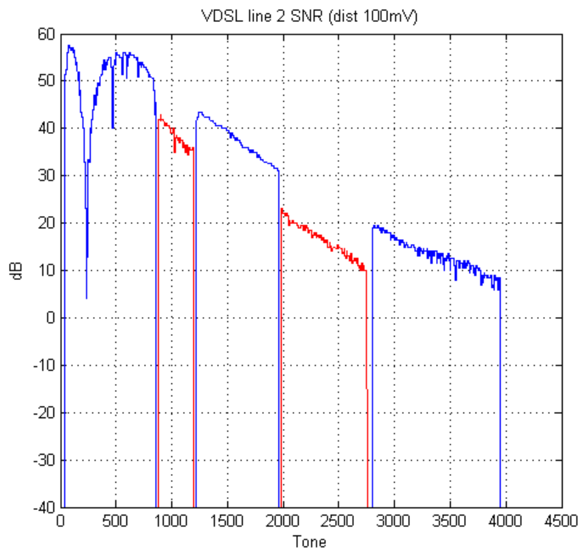
First, two reference VDSL2 DELT measurements were performed, one without any disturbances and any interfering DSL links in the binder and one with an ADSL2+ link running in a parallel pair in the binder. Figure 5.1 shows the SNR per tone for the two cases. As can be seen, the ADSL2+ link has no, or very limited, impact on the SNR per tone for the VDSL2 link.

Emulated AM radio disturbances were added to the VDSL2 line. The output level of the Radio Frequency (RF) disturbance signal was increased from 0.1 mV to 100 mV in steps according to Table 5.2. During this experiment an ADSL2+ link was active over a parallel pair in the binder. The resulting per tone SNRs for the test cases RF signal levels equal 1 mV and 100 mV are shown in Figure 5.2. In the Figure 5.2(a) the emulated radio signal is visible, and in Figure 5.2(b) also the first overtone can be seen. The bit loading per tone will of course reflect the decreased SNR by moving away bits from the affected tones. This is shown in Figure 5.3.

The impact of these disturbances on the IP level was then tested. A stream of UDP packets was sent over the VDSL2 link using the NTools applications



(a) VDSL2 DELT with a 1mV disturbance.



(b) VDSL2 DELT with a 100mV disturbance.

Figure 5.2: SNR with applied AM radio disturbance.

time	*RF level*
0 min	0 mV
2 min	0.1 mV
4 min	0.25 mV
6 min	0.5 mv
8 min	1 mV
10 min	10 mV
12 min	25 mV
14 min	50 mV
16 min	100 mV

Table 5.1: RF level settings during the test

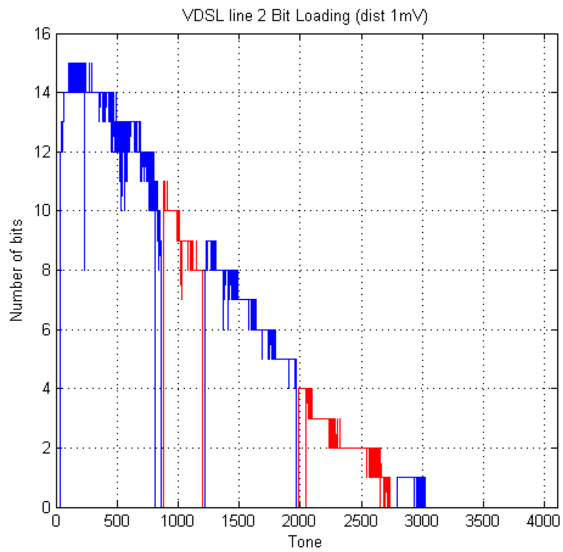
[104]. All packets carried 1400 bytes of data and were sent with deterministic intervals, yielding an IP layer bit rate of approximately 3 Mbps. Each received packet was time stamped and logged on arrival. All packets were also tagged with a sequence number by the sender, making it possible to detect packet loss.

During the experiment, the radio signal with a centre frequency of 1 MHz was induced on the link. The disturbance signal level was increased each other minute, starting at 0 mV, according to Table 5.2.

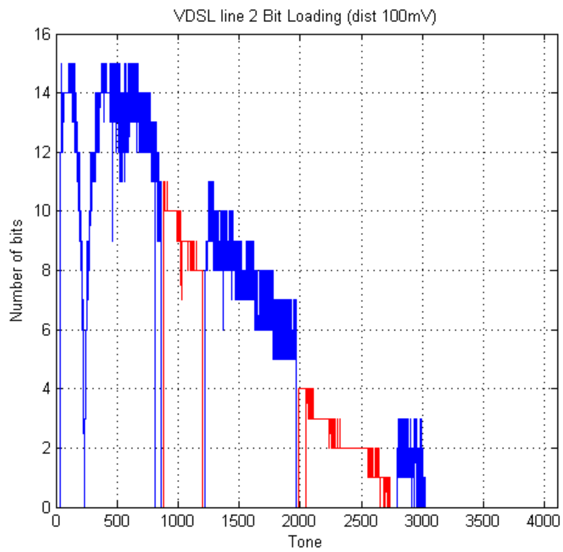
The resulting Inter Packet Gap (IPG) and packet loss were plotted, see Figure 5.5. As can be seen, both parameters are influenced negatively by the applied disturbance. Also seen are incidences of up to 30 consecutive packets lost. Of special interest is packet loss at the increase in disturbance RF level from 1 mV to 10 mV. This leads to severe packet loss and thus also IPGs in the order of 100 ms per occurrence. Note that following the increase of the RF signal level to 10 mV, the effective IPG is a combination of several instances of high IPG. Disturbances of this type will have a negative impact on e.g. streamed TV over Internet (IPTV) Quality of Experience (QoE).

5.2.1 Disturbing the Up-link

With the same lab setup, i.e. where the disturbance is applied closed to the CPE, the carrier frequency was increased so it coincided with the first up-link tone group. The resulting SNR and bit loading are shown in Figure 5.4. As can be seen, the up-link tone group is not affected but the downlink is. This is expected, since the disturbance is applied close to the CPE.

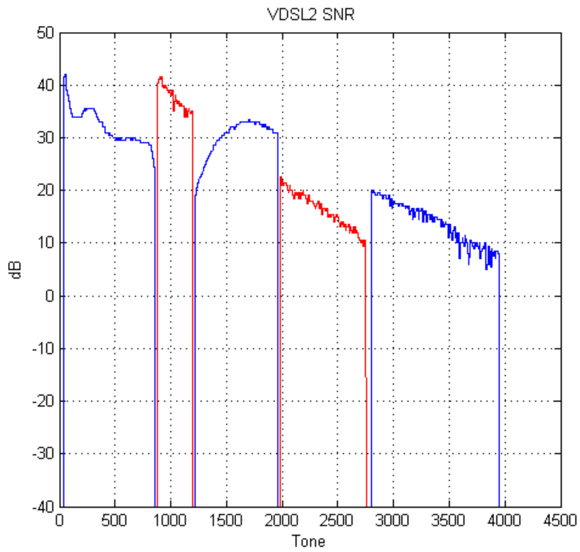


(a) VDSL2 bit loading with a 1mV disturbance.

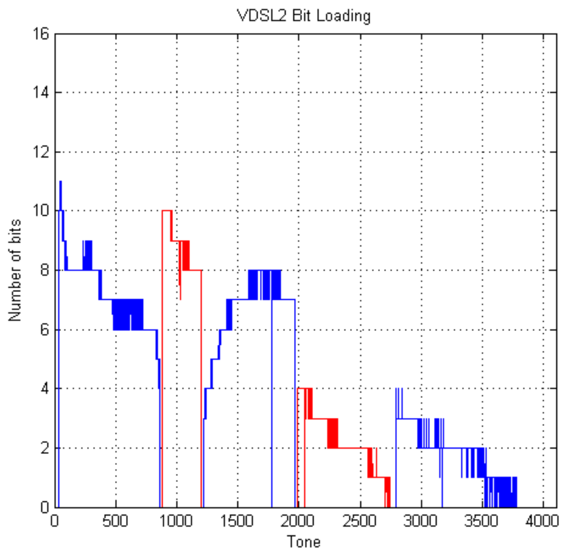


(b) VDSL2 bit loading with a 100 mV disturbance.

Figure 5.3: Bit loading with applied AM radio disturbance.



(a) SNR



(b) Bit loading

Figure 5.4: SNR and bit loading for VDSL2 with disturbance in first up-link tone group.

5.2.2 ADSL2+ Compared with VDSL2

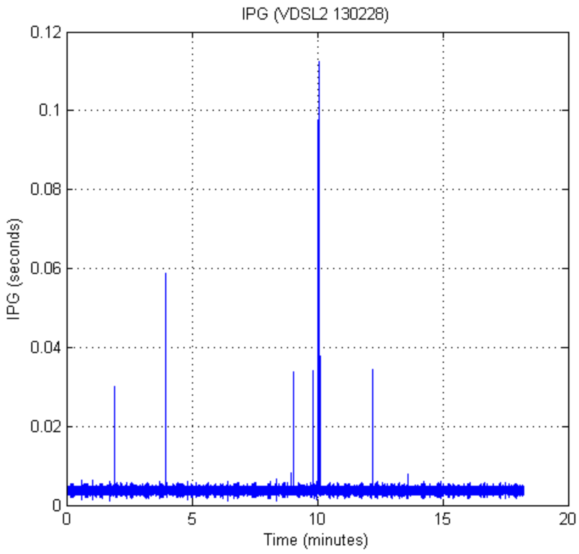
For comparison, the same experiment was repeated once more, but this time on an ADSL2+ link. The corresponding IPG and packet loss plots are found in Figure 5.6. It can be seen that the disturbance's impact on ADSL2+ is much more severe than for VDSL2. For instance, the shift from RF level 1 mV to 10 mV at time=10 min causes IPG in the order of 10 seconds when over 200 packets are lost. When the disturbance RF level is increased from 25 mV to 50 mV, the link is forced to retrain.

From these simple experiments, it can be concluded that ADSL2+ is more vulnerable than VDSL2 to radio signal disturbances. The impact on the QoE of such a disturbance can be foreseen to be more severe on an ADSL2+ access link than on a VDSL2. Unrecoverable packet losses will always lead to visible distortion, but Packet Delay Variation (PDV) below approximately 200 ms will be handled by a normal jitter buffer. Notable is that the applied disturbance did not force a retrain of the link.

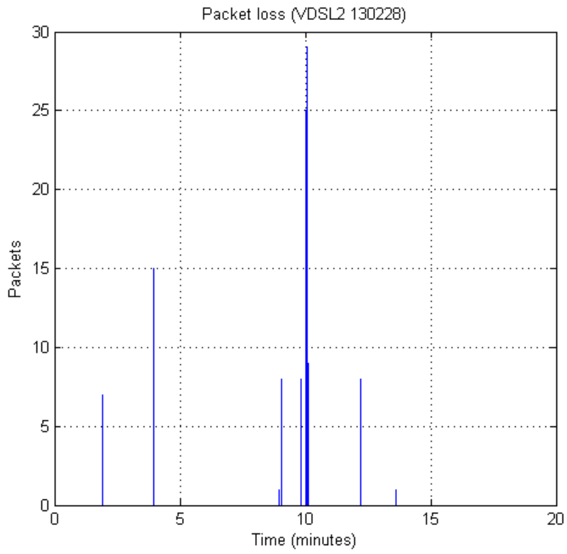
5.3 Comparison of Two CPEs

In the experiments described in the previous section, CPEs from two different manufacturers were used. It was thus of interest to compare the performance of the two CPEs using the same type of DSL technology. Therefore, an experiment whereby the used CPEs' performance on an ADSL2+ link was tested. The simulated radio station disturbance was applied to the ADSL2+ according to Table 5.2. The IPG and packet loss as a function of time for the two CPEs are plotted in Figure 5.7. The plots show that the lesser resilience to disturbances actually follow the DSL technology, not the individual CPE. The DSLAM interface and the cabling between the DSLAM and the CPE were the same in both experiments.

The proposal that VDSL2 should be more resilient to the simulated radio station might be a result of the disturbance hitting relatively fewer tones when applied to VDSL2 than to ADSL2+; The VDSL2 band plan used have the same tone spacing as ADSL2+, but the number of available tones were approximately four times higher for the VDSL2 case than for ADSL2+.

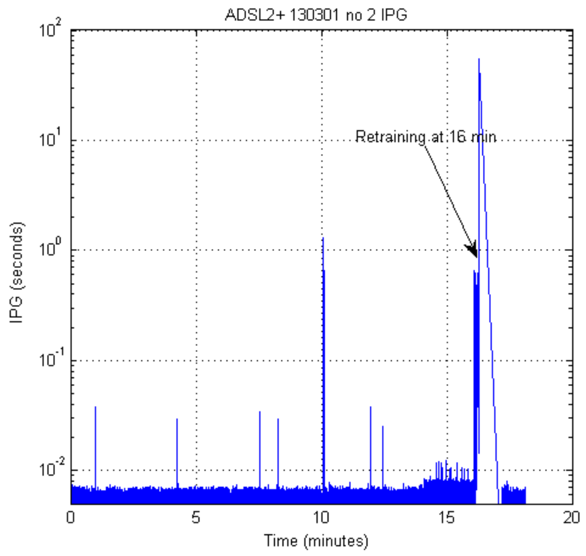


(a) IPG

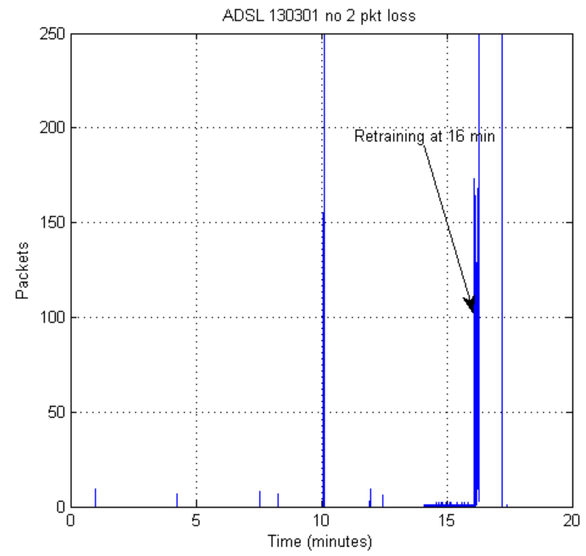


(b) Packet loss

Figure 5.5: IPG and packet loss from the disturbed VDSL2 line.



(a) IPG

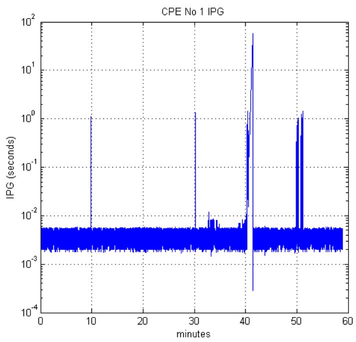


(b) Packet loss

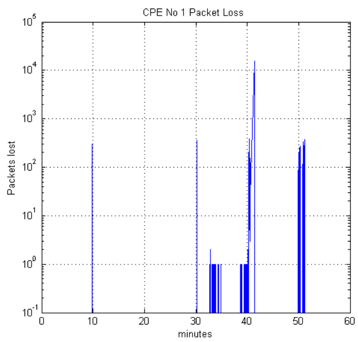
Figure 5.6: For comparison, IPG and packet loss from the disturbed ADSL2+ line.

Test section	*Time*	*Disturbance*
1:	0-10min	No disturbance
2:	10-20min	Disturbance at $f_c = 2.0MHz$
3:	20-30min	No disturbance
4:	30-40min	Disturbance at $f_c = 6.5MHz$
5:	40-50min	Disturbance at $f_c = 13MHz$
6:	50-60min	Disturbance at varying f_c

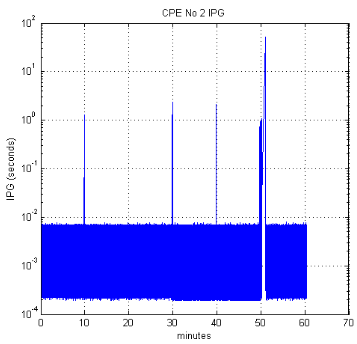
Table 5.2: Test scheme for ADSL2+ with simulated radio station disturbance added.



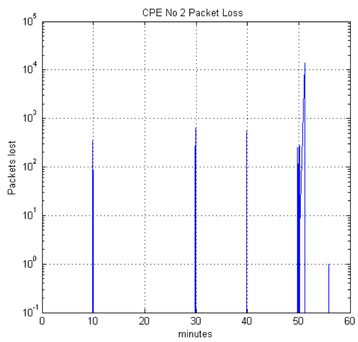
(a) IPG for ADSL2+ CPE no 1.



(b) Packet loss for ADSL2+ CPE no 1.



(c) IPG for ADSL2+ CPE no 2.



(d) Packet loss for ADSL2+ CPE no 2.

Figure 5.7: IPG and packet loss for two ADSL2+ CPEs.

Chapter 6

Impulse Disturbance's Impact on IP over DSL

To be able to realise new technologies and services e.g. mobile backhauling over DSL access networks, a deeper understanding and relation between measured Quality of Service (QoS) parameters on different layers in the OSI reference model is essential. Electrical Impulse Noise (EIN) is one of the most severe types of external interference to Digital Subscriber Line (DSL) systems. This chapter studies the relation between impulse disturbances, link utilisation and packet loss. The work was presented at SoftCom2014 [50].

6.1 Impulse Disturbances, and a Hypothesis

Impulse disturbances often appear in bursts of very short pulses. The spectrum is more or less white and covers more than the frequency spectrum of interest. Each white noise pulse is normally very short in time, but the pulses appear as bursts with durations in the same order as the duration of one Very-high-bit-rate Digital Subscriber Line version 2 (VDSL2) system's Orthogonal Frequency-Division Multiplexing (OFDM) frame. A burst of such noise will hit all tones in a frame, thus effectively averting bit swapping.

It is known from e.g. [76] [58] [95] [99], that there is a relation between specific physical layer parameters and Quality of Experience (QoE) for the end user, but not how this relation is manifested. From studying the internal functions of a VDSL2 system in the VDSL2 standard G.9332 [33] two hypotheses were formulated:

- Packet loss on the network layer is dependent not only on noise bursts levels and duration, but also of the user data utilisation of the link
- There exists a utilisation threshold over which every noise burst introduces a network layer packet loss.

To validate these hypotheses, first a theoretical derivation of the packet loss probability conditioned on an impulse was performed, followed by experiments using a VDSL2 system, where impulse noise was injected into the transmission cables. The theoretical derivation in Section 6.2 is for a basic scenario without Forward Error Correction (FEC) and physical layer retransmit. The objective was to establish a starting point for further elaborations aimed at the, in practice, more realistic and by far more complex case, if possible. The experiment described in Section 6.3 was performed with the same assumptions. In reality, an operator will of course deploy the coding and error correction functionality called for, but the current investigation gives a primary understanding of the relation between impairments on the physical layer and the network layer for this disturbance case.

6.2 Estimation of Packet Loss Probability

An Ethernet frame transmitted over a VDSL2 system is converted to a group of OFDM frames, which forms the signals for the transmission. The process is described in Chapter 2. By deriving this process, it should be possible to find the probability of a single white band impulse disturbance hitting one Internet Protocol version 4 (IPv4) datagram.

The packet stream in the derivation is based on IPv4 datagrams with payload 1400 bytes, equally distributed over time to achieve a certain data rate. These IPv4 datagrams correspond to Ethernet frames of length $L_e = 1438$ bytes. In the derivation the configured data rate and the considered service data rate were denoted by R_C and R_S respectively.

In an OFDM system, the tone spacing determines the symbol rate, since this is also the OFDM frame rate on the physical layer. With a tone spacing of 4.3125 kHz, the OFDM frame length equals $\frac{1}{4.3125} = 232\mu s$. Adding the cyclic extension extends the OFDM frame to $250\mu s$. Hence, in VDSL2 the transmission rate of the OFDM frames is $F_S = 4$ kHz. Then each of the OFDM frames carry $L_O = \frac{R_C}{8F_S}$ bytes.

When re-framing Ethernet frames into Packet Transfer Mode (PTM) frames, two or four Cyclic Redundancy Check (CRC) bytes, N_{FCS}, are appended to the PTM frame. Additionally, two overhead bytes, SC, are added, one in the beginning and one at the end of each PTM frame. The PTM frames are then split in blocks of 64 bytes. Each of these blocks are framed by one byte, meaning the number of bytes corresponding to an Ethernet frame of L_e bytes is

$$L_E = L_e + N_{FCS} + SC + \left\lceil \frac{L_e + N_{FCS} + SC}{64} \right\rceil$$

where the last term corresponds to Physical Media Specific Transport Conversion (PMS-TC) framing overhead. In our case, an adjusted number of bytes per transmitted Ethernet frame is $L_E \approx 1474$.[1]

The minimum number of OFDM frames output by the Physical Media Dependent (PMD) block containing any of the bytes from the original Ethernet frame is

$$N_{\min} = \left\lceil \frac{L_E}{L_O} \right\rceil$$

Alternatively, an Ethernet frame can occupy $N_{\min} + 1$ OFDM frames and the probabilities for these two events are

$$P_{EO}(N_{\min}) = \frac{N_{\min} L_O - L_E}{L_O}$$
$$P_{EO}(N_{\min} + 1) = 1 - P_{EO}(N_{\min})$$

Hence, expected number of OFDM frames occupied by an Ethernet frame is

$$E[N] = \frac{L_E + L_O}{L_O}.$$

Similarly, it is important to measure the gap between two consecutive Ethernet frames in terms of number of OFDM frames. Since a sequence of evenly spaced Ethernet frames is considered, it is possible to derive the number of bytes in one period, i.e. from the start of an Ethernet frame to the start of the next, as $L_P = \frac{R_C}{R_S} L_E$, and the total number of bytes in a gap is $L_G = L_P - L_E$. The maximum number of OFDM frames in a gap that does not have any content from the Ethernet frames is

$$G_{\max} = \left\lfloor \frac{L_G}{L_O} \right\rfloor$$

and alternatively the gap can contain $G_{\max} - 1$ idle OFDM frames. The corresponding probabilities are

$$P_{GO}(G_{\max}) = \frac{L_G - G_{\max} L_O}{L_O}$$
$$P_{GO}(G_{\max} - 1) = 1 - P_{GO}(G_{\max})$$

[1] The process is described in Section 2.1.1.

The expected value of a gap is, as long as $L_G \geq L_O$, $E[G] = \frac{L_G - L_O}{L_O}$.

Finally, to derive the probability for packet loss from an impulse disturbance, the span of the impulse must also be translated into OFDM frames. In [36, 40] measurements on impulse noise in the home network have been carried out. There is a wide spread in the time duration of the impulses, but an average value of about $100\mu s$ turns out to be reasonable for tests. Since the cyclic extension of the OFDM frame will be discarded in the receiver, it is not a problem if an impulse hits this part. Therefore, the effective burst duration will be $\tilde{T}_B = T_B - T_{CE}$, where T_B is the burst duration and T_{CE} the duration of the cyclic extension. The minimum number of OFDM frames affected by an impulse of time T_B is

$$B_{\min} = \lceil \tilde{T}_B F_S \rceil$$

It should be noted that the expression for $\tilde{T}_B$ can be negative, which means that the burst time is shorter than the cyclic extension. This will give that $B_{\min}$ equals zero. The probabilities for the number of OFDM frames affected by an impulse is

$$\begin{aligned} P_{BO}(B_{\min}) &= B_{\min} - \tilde{T}_B F_S \\ P_{BO}(B_{\min}+1) &= 1 - P_{BO}(B_{\min}) \end{aligned}$$

The corresponding expected value is $E[B] = 1 + \tilde{T}_B F_S$.

Now, as everything is synchronised with the OFDM frames, the probability for packet loss due to a burst can be derived. Under the condition that an Ethernet frame corresponds to N OFDM frames and the corresponding gap to G OFDM frames, and that the burst corresponds to B OFDM frames, the packet loss probability is

$$P(PL|N,G,B) = \frac{N+B-1}{N+G}, \quad L_G \geq L_O$$

That gives the unconditional probability as[2]

$$\begin{aligned} P(PL) = \sum_{i,j,k=0}^{1} & P_{EO}(N_{\min}) P_{GO}(G_{\max}) P_{BO}(B_{\min}) \\ & \cdot P(PL|N_{\min}+i, G_{\max}-j, B_{\min}+k) \end{aligned} \tag{6.1}$$

Thus, the probability of a loss of a non-fragmented IP packet given an impulse disturbance is equal to the probability that an Ethernet frame carrying an IP packet is lost. This probability, in turn, is dependent on the number of OFDM frames that carries the lost Ethernet frame, the number of OFDM frames that

[2]It is here assumed that the Ethernet frame length and the gap length are independent. This is not entirely true but it simplifies the calculations considerably and the error is negligible.

corresponds to the gap between two Ethernet frames, the number of OFDM frames the impulse disturbance hits, and the probability that the burst hits at least one of the OFDM frames that is carrying the lost Ethernet frame.

6.3 Experiment and Measurements

In the experiments, Ethernet frames with Internet Protocol (IP) payload of 1400 bytes were transmitted over a VDSL2 link, configured for maximum bit rate of 60 Mbps. The packets were transmitted equally spaced with various total bit rate for the different test runs. For each test run 250 impulse disturbances were injected, spaced by 20 seconds, giving a total experiment duration of 5,000 seconds. The duration of each impulse disturbance was $T_B = 100\mu s$. A total cable length of 950 m was used, and the impulses were injected near the Customer Premises Equipment (CPE). Every three seconds the accumulated number of Errored Seconds (ESs) and Code Violations (CVs) were read. The number of packets lost during one second was sampled periodically. Each reading was time stamped with Network Time Protocol (NTP) synchronized clocks. The raw data was then scanned finding CVs and packet loss events within plus/minus 3 seconds from the time an impulse was injected. The test results are given in Table 6.1.

To comply with the restrictions of the analytical derivation above, the experiment was performed without active Impulse Noise Protection (INP) and neither physical layer retransmission nor Trellis coding was activated. Thus, no shift in latency is expected; only packet loss should be seen.

6.4 Results and Discussion

Table 6.1 shows that a CV occurs for almost all impulse disturbances, independent of the service rate. An impulse disturbance will always hit an OFDM frame, which will generate a CV, independent of the OFDM frame carrying user data or idle bits. Thus, probability of packet loss is a function of the service rate. When the service rate is low compared to the configured rate, there will be many OFDM frames carrying only idle bits. If an impulse disturbance hits such a frame, a CV will be the result but it will not affect user data.

In the experiment leading to Table 6.1, the energy of an impulse disturbance is much higher than the received signal in the downstream direction. Even when the impulse overlaps only a small part of the OFDM frame in time, the impulse

Table 6.1: Result from test runs. Column 2 shows the number of CVs given an impulse disturbance. Column 3 shows the number of packet loss events given a CV given an impulse disturbance.

Rate (Mbps)	**# CV**	**# Pkt Loss**	**Pkt Loss/CV**
1	249	27	0.1084
3	242	39	0.1612
7	250	68	0.2720
11	243	95	0.3909
15	249	120	0.4819
20	248	151	0.6089
25	249	182	0.7309
30	248	202	0.8145
35	250	215	0.8600
40	244	230	0.9426
45	246	241	0.9797
50	250	250	1.0000
55	247	247	1.0000
58	247	247	1.0000

energy will be evenly distributed over the frame in the frequency domain after the Fast Fourier Transform (FFT) in the receiver. If the resulting noise level is stronger than the Signal to Noise Ratio (SNR) margin, this should give an extensive error over the complete frame. Hence, there will in our case essentially always be a fault in at least one OFDM frame. However, this is dependent of the energy of the impulse noise. Figure 6.1 shows the average number of ESs and CVs as a function of the power level of the impulse noise. There is a quite distinct threshold where the energy of the noise will fill up the SNR margin in the VDSL2 bit loading.

In the laboratory set-up described above for a VDSL2 connection using the 17a band plan, the measurements in Table 6.1 are plotted in Figure 6.2 together with the 95% confidence intervals. In the figure, the estimated probability of packet loss for the same settings is also plotted. It is seen here that the curves coincide reasonably well. In the derivations it is assumed that all Ethernet frames are handled by the system without changing the inter packet gap, i.e. the distribution of arrivals to the TPS-TC and PMS-TC blocks is deterministic. In reality this is not necessarily true; the measured data suggest that some functionality in the VDSL2 system changes the inter packet gap slightly[3].

The break point in service data rate, for which a burst always will affect the

[3] This is discussed in Section 4.3.3.

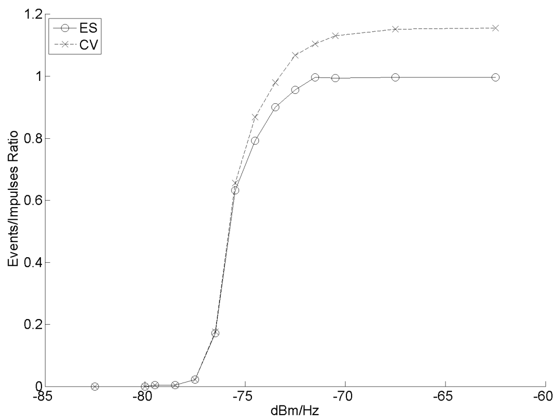


Figure 6.1: The number of ESs and CVs as function of power level of 100 μs long impulse disturbances.

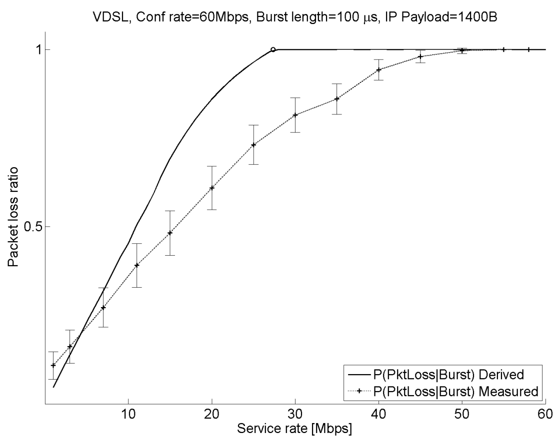


Figure 6.2: Estimation and measurements for packet loss probability from a burst.

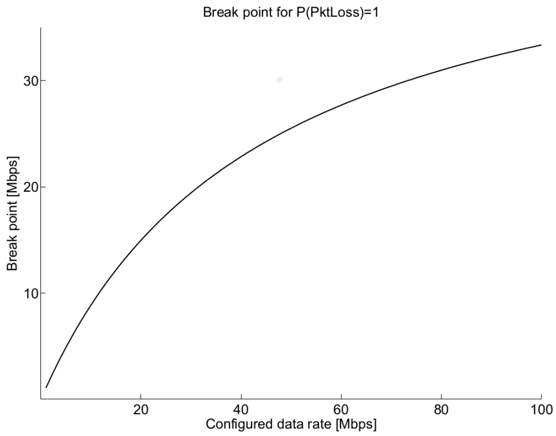


Figure 6.3: Estimation of the break point for which a burst will always give a packet loss in the IP stream.

service in form of packet loss, is an interesting parameter for the system. For the theoretically estimated plot, this can be derived by viewing the case when the gap length is shorter than the gap, giving the following bound for when $P(PL) = 1$,

$$R_S > \frac{L_E}{L_O\lceil \frac{L_B}{L_O} \rceil + L_E - L_{CE}} R_C \tag{6.2}$$

For $R_C = 60$ Mbps it gives the service rate $R_S > 27.4$ Mbps, which is plotted in Figure 6.2 as a circular mark. In Figure 6.3 this break point service data rate is plotted as a function of the configured data rate. As can be seen, the break point is roughly half of the configured data rate. This might be the reason for the assumption that one CV will invoke one packet loss. This miss-understanding probably comes from the deployment of TV over Internet (IPTV) over Asymmetric Digital Subscriber Line, version 2+ (ADSL2+), where this assumption is true in many cases, but it does not apply to VDSL2.

The effect of noise impulses on packet loss of an IP stream does not depend only on how the noise impulse affects OFDM frames but also to what extent OFDM frames carrying user data are hit. Thus, there is no direct mapping between ES or CV on the physical layer and QoE for the service. Because of how the Digital Subscriber Line Access Multiplexer (DSLAM) inserts idle bytes to fill up the OFDM frames and to what extent other network layer packets carrying other services are present, packet loss depends on how much the packet stream of interest utilises the available link capacity. The effect of an impulse is therefore also a function of the link's configured maximum data rate and the regarded service data rate. The service data rate when each burst will have an impact on each network layer packet is significantly lower than the channel's configured maximum data rate when no Impulse Noise (IN) protection is deployed. The presented probability estimation model can easily be adopted to other OFDM based access systems, such as G.fast, Wi-Fi and Long Term Evolution (LTE).

Chapter 7

The Adaptive Network

In the introductory chapter, the need for keeping the customer satisfied without intervening with customer support was discussed. It would be beneficial if the network itself could identify and act on changes in the network performance that affects the user’s perceived quality of the service involved. It is even more beneficial if this adaptation could be deployed *before* the user has started to react on the degraded Quality of Service (QoS). The Celtic Plus project Road to Media-Aware User-Dependant Self-Adaptive Networks (R2D2) [37] was investigating methods for deploying a self adaptive network.

What is presented in this chapter can be seen as proof-of-concept, rather than new science.

7.1 The R2D2 Project

The R2D2 project aimed at finding a way to create self-adaptive networks that are user-dependant and media-aware. The objective was to adapt both services delivered over a triple-play network as well as the network itself to current usage and demands so that the end user is delivered a service quality that can be the best or at least acceptable in all situations. One example is to let the network demand a change of codex for a service in the case that the network is experiencing over-utilisation. Our contribution in this project was to interface a typical Digital Subscriber Line Access Multiplexer (DSLAM) management application to the Network Resource Manager (NRM) to let e.g. a changed bit loading event that reduces the Digital Subscriber Line (DSL) link bit rate trigger a change of codex for a video streaming service.

7.2 The NRM and EMSs

A main target for R2D2 was to create a centralised function for performance management in an operator's heterogeneous access network. This function was called NRM. The objective of the NRM was to react to performance changes anywhere in the operator's distribution chain from source to customer. The reaction could be to change physical or link layer parameters, i.e. increase link capacity or re-route traffic, or to adapt the service transported over the network to the new conditions, i.e. change the COder-DECoder (CODEC) for a video stream to one that better match the changed situation.

The NRM had to be able to communicate and manage a wast number of different manageable nodes. Examples of nodes are routers and switches, but also management systems for example DSLAMs. For this sake, a general function, called Element Management System (EMS), was introduced. An EMS is an agent, implemented in a managed node, that act as a shim between the NRM's generalised control and surveillance functions and the individual nodes. Our contribution to the R2D2 project was to create and implement an EMS for a DSLAM, so that individual customer lines could be managed.

7.3 DSL Link Management Issues

DSL service levels are based on an agreement between the end customer and the operator, called Service Level Agreement (SLA). The operator offers different bit rate profiles, for example 8 Mbps in the downlink and 0.5 Mbps in the uplink. Normally only a very limited set of profiles are available for the customer. The profiles only offer maximum bit rates; the actual bit rate is established first as the customer deploys the Customer Premises Equipment (CPE) to the line.

The DSL service is provisioned over the user's or customer's telephone line, also called local loop. Limitations per subscriber are related to the quality of the subscriber's local loop and the aggregation of user traffic in the aggregation network.

There are a limited number of methods to adapt the line settings, if the conditions for a specific customer line changes. Most of them involve re-training of the CPE and DSLAM interface. A re-training effectively disrupts the customer's

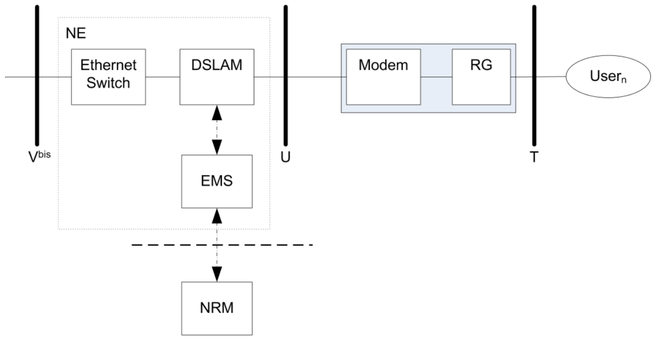


Figure 7.1: The DSLAM testbed with its EMS.

Internet connection for at least half a minute. During this time, no information can flow in either direction. The role of the NRM will therefore be to check status parameters in the DSLAM and compare them to the current SLA and the requirements for the content delivered. If the status parameters do not fulfill the level of QoS, the proposed first measure is to adapt the content service level to the current conditions. If that is not possible, a re-training might be solicited.

The parameters that the NRM can request from the DSLAM EMS are of snapshot type. The NRM can use some of these values to trigger alarms to other systems if limits are breached or levels reached. The NRM can also send alarm levels and limits for some parameters to the EMS, thus making it possible for the EMS to send appropriate alarms to the NRM. Other parameters are of more interest for a human operator, and as such could be displayed on a management screen.

7.4 The DSLAM EMS

The DSLAM EMS, Figure 7.1, was an integrated part of the DSL testbed included in the R2D2 common testbed. An out-of-the-box DSLAM was used for the project; No extensions were added to the DSLAM itself.

The EMS was implemented in MATLAB®. Figure 7.2 shows the flow chart of the EMS main loop. Data between the NRM and the EMSes is coded as Extensible Markup Language (XML) messages. Thus, the MATLAB® library xml_io_tools [102] was used to convert between MATLAB® structures and XML.

7.4.1 The EMS in Detail

The DSLAM delivers all data to the EMS on request, i.e. data pull. In turn, the EMS receives *get* and *set* requests from the NRM, and delivers responses to the requests back to the NRM. The EMS also monitors some of the DSLAM's QoS parameters and if these parameters exceed alarm levels, set by the NRM, an alarm message is sent to the NRM, i.e. data push.

The received XML message is transferred to a *get* or a *set* function, depending on the type of XML message that is received. The *get* function reads requested parameters from the DSLAM with Simple Network Management Protocol (SNMP), and creates another XML message containing the answer, which is sent to the NRM.

The *set* function works similarly. The parameters, i.e. alarm levels, that are contained in the received XML message are not transferred to the DSLAM since they are aimed for the EMS. A response message including the result of the *set* commands is created an returned to the NRM. One *set* parameter that is actually transferred to the DSLAM, is the re-train command.

Alarm levels can be set for current bit rate, number of Errored Seconds (ESs), number of Cyclic Redundancy Check (CRC) errors, and number of line initialisations. Only the current bit rate is an absolute alarm level; The other alarm levels are reflecting the number of incidents that happens after one alarm has been triggered. For example, if the ES alarm level is set to three, an alarm message will be sent to the NRM on every third ES incident. If the alarm level for any parameter is set to zero, no alarms will be sent. The absolute value of a monitored parameter can be read by the NRM using a *get* XML message.

7.4.2 Three Different Bit Rate Parameters

The three different bit rate parameters need a comment. The *attainable* bit rate is a calculated maximum, based on measured Signal to Noise Ratio (SNR) and could be said to be the theoretical upper limit for that specific line in current environment. The *configured* maximum bit rate is set by the ISP and must be lower than the attainable bit rate. This is the maximum capacity that a customer can attain, and should be set according to the contracted service level for that customer. The *current* bit rate is what the line actually delivers to the customer, and is less or equal to the configured bit rate. Thus:

$$AttainableBitrate > ConfiguredBitRate > Currentbitrate \tag{7.1}$$

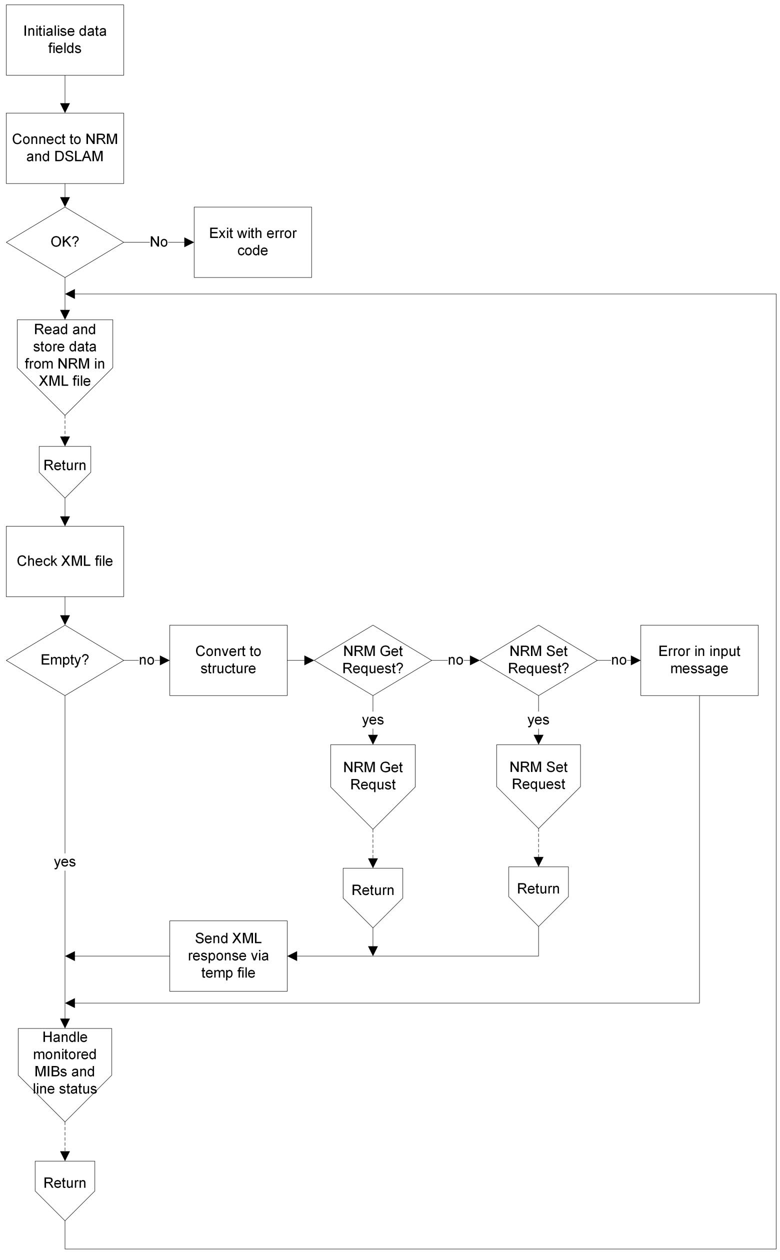


Figure 7.2: The DSLAM EMS main loop.

7.5 Proof of Concept

At the R2D2 final review in Madrid, the DSLAM with its EMS was implemented as part of the projects combined test bed. One of test cases was a disturbance degrading the bit rate on the DSL link while a video stream was sent on the link. The disturbance triggered a bit-swapping event and the resulting bit rate was too low to carry the video stream without impacting the visible QoS. Because the new bit rate was lower than the alarm level, the bit-swapping event raised an alarm to the NRM. The NRM as a consequence changed the coding of the video stream at the delivery server to one with lower output bit rate and thus restored the service.

Chapter 8

Profiling Users in a VoD Network

The studies that were the basis for the earlier chapters of this thesis discuss the effect disturbances on the lower layers of a link has on the traffic it transports. Even if this chapter does not directly involve disturbances, it is well motivated. This application level study concludes our cross layer project.

One way to take measures against disturbances is to allow for pre-fetching and terminal or personal caching. In this way, it is possible to cover up for at least some imperfections in the data transport for the good cause of keeping the customers satisfied.

In this chapter, accounts and channels in a Video on Demand (VoD) network associated with strong or weak surfing or browsing respectively are identified. The distributions of program hold times for the different types of behaviour are studied, and the potential of different pre-fetching and/or caching strategies for different user groups is discussed with respect to surfing or browsing. Finally, results from a request prediction model and a caching simulation for the different types of behaviour are presented. The study has been published at the international symposium BSBM 2016, Nara, Japan.

8.1 Setting the Stage

Nowadays, other than air broadcasting, cable networks and physical media like Digital Video Disc (DVD) or Blueray Disc, Internet has become a popular medium for distributing multimedia content like TV shows, movies, and user generated videos. The massive amount of multimedia traffic has imposed a significant burden on the Internet. Consumer internet video traffic, e.g. TV over Internet (IPTV) and VoD, which is the application studied in this chapter, was 64% of all Internet traffic in 2014 [44]. Consequently, users sometimes have to endure long access delay for filling up the playout buffer before the content is displayed. In [93], the result shows that user's tolerance of waiting time for downloading web pages is about 2 seconds. Results in [70] suggest that the more familiar users are with a web site, the more sensitive they are to delays.

According to [43], the Internet traffic is more and more centric to data centres, a development that will continue increasing in the future due to the increasing use of cloud computing. From around year 2008, peer to peer traffic no longer is the major contributor to Internet utilisation. Multimedia traffic is also by nature data centre focused and thus also has an impact on the Internet infrastructure. Although web caching is widely used as a solution to lessen the web traffic congestion and improve the network performance, the benefit of caches is limited. To further reduce the user perceived latency , pre-fetching has become a popular technique. The objective of the pre-fetching system is to pro-actively load certain content to the cache even before users request it.

When browsing or *zapping*, that is quickly switching from one TV channel to another, a user initiates many program views but consumes only a small part. Reducing the zapping time in IPTV services is essential for the user's perceived Quality of Experience (QoE) [87]. Reducing unnecessary downloads of unseen parts of programs is also of importance not only for fixed access networks but also in mobile networks [84][71].

VoD services have been studied regarding both pre-fetching and caching, but using the same strategy for all subscribers. Users groups' variant behaviour, especially regarding zapping, is expected to influence the network's pre-fetching and caching strategies. Very little has previously been studied regarding pre-fetching and caching strategies adaption to user profiles in VoD networks. This study tries to narrow this research gap by studying users as well as channels in this context. Two user groups, *zappers* and *loyals* are identified and their impact on pre-fetching and caching in a so called Catch-Up TV network is analysed. Furthermore, differences between zapping prone and loyal prone channels are presented. Zappers changes quickly from program to program searching for the wanted view while loyals consumes the first selection to the end before requesting another view.

Earlier studies have analysed zapping in broadcast or IPTV networks. Cha *et al.* have analysed user behaviour in an IPTV network [63]. Three user modes were defined and analysed, *surfing*, which is the equivalent to what herein is called zapping, *viewing*, and *away*. In [75] Gopalakrishnan *et al.* models the behaviour of a single user - the Couch Potato - viewing session with events like Start, Pause, Play, Fast Forward, and Rewind. From state Play 63% of all state changes go into Fast Forward, an indication of zapping behaviour. Ali-Edin *et al.* discusses the "impatient user" behaviour in [49]. In their study over 90% of all sessions last less than one hour, and 20% less than 30 seconds. A comparison with Yu *et al.* [110] shows the same behaviour, except for views lasting longer than 10 minutes. The impact on pre-fetching and caching is not studied.

Zapping in a broadcast or IPTV network means changing of channels. In a catch-up TV network or a play service, zapping has another modus, users tend to follow series and switches between episodes rather than channels. The analysis by Du *et al.* in [68] is focused on pre-fetching in a VoD service but has also looked into the viewing time per request. 20% of all requests are shorter than two minutes, which can be defined as zapping. The impact from zapping on caching and/or pre-fetching strategies has also been studied by Du *et al.* [68] and Zhang [113]. In these studies no categorisation of users were done.

The studied operator's service is to offer channels' subscribers with the option of watching programs in retrospect. Users in the studied network subscribe to some, but not necessarily all, of the available channels. Their selection of views is for series and episodes rather than channels.

8.1.1 The Dataset

The dataset used for this study comes from a major European IPTV operator. It contains one month of request logs of their Catch-Up TV service from June 1 to June 30, 2014. Different from traditional TV-on-demand service, catch-up TV provides a shorter content access window, in this case seven to fourteen days, for post broadcast viewing of the channels which are subscribed by users. The dataset contains over 17 million requests from about 570,000 accounts viewing more than 80 TV channels.

Table 8.1 shows an example of a request log. Account is the unique identifier of a specific user. PlayTimeHour records the time when the content was played back. City and District indicates user's geographical location. The details of a specific video include Title, SeasonNumber, EpisodeNumber, StationID (unique

Table 8.1: Example of dataset record.

Account	<user id>
PlayTimeHour	2014-06-07 08:43:52.287000000
City	<City name>
District	<District name>
Title	<Program name> - Ep. 11
IsPC	False
SeasonNumber	2
EpisodeNumber	11
EpgSeriesID	28849
EpgPID	635827
StartTime	2014-06-06 19:30:00
EndTime	2014-06-06 19:50:00
StationID	<Channel id>
CachedThemeCode	MSEPGC-Others

identifier of a channel), Video category (CachedThemeCode) and the length of the video (StartTime and EndTime). EpgPID is the unique ID for each video within each channel and EpgSeriesID is the unique ID for each program (series).

8.1.2 Definitions

Browsing the literature for well-known and accepted definitions or notions of parameters and characteristics related to this study shows a lack of consensus. In this paper the following definitions will be used:

- A *request* is a user selecting a specific program to view.
- A request initialises a *program view*, and the duration of a view is often referred to as *view time*. In this context, the term *program hold time* will mostly be used.
- Several consecutive program views back-to-back form a *session*, and the duration of a session is referred to as *session time* or duration.
- The pause or gap between two sessions is referred to as an *intermission*.
- Often the word *user* is used. In the IPTV operator dataset, only individual *accounts* are identified, and of course many users can use the same account. Thus the notion of a user actually refers to an account.

- Cha *et al* defines zapping time as the time from a channel change request till the new channel is shown on the screen [63]. The users' actions of frequently switching between programs is thus in this study called *zapping*.
- A user that is prone to zapping is defined being a *zapper* while a *loyal* is a user that is the opposite.

8.2 Identifying and Discussing Zappers and Loyals

The aim of this study is to try to find differences in two user groups' impact on pre-fetching and caching. One group consists of extreme zappers in the sense that these users have a high relative rate of short program hold times per viewing session. The extreme loyals on the other hand have a low relative rate of short program hold times.

8.2.1 Defining Program Hold and Zapping Ratio

As stated earlier, the dataset consists of logged and time stamped request events. To reduce ”noise”, only logged data from what can be denoted as active accounts were included in the study. An active account is defined by belonging to the 30% of all individual accounts that contribute to 75% of all requests, see Figure 8.2.

The analysed dataset lacks logging of when a user stops viewing a program or even starts viewing it; Only the time when the user requested a specific program is recorded. The program hold time needed for the zapping ratio thus has to be estimated, equation (8.1).

$$p_i = \min\{(r_{i+1} - r_i), (e_i - s_i)\}, i = \{1, \ldots, (l-1)\} \tag{8.1}$$

p_i is the program hold time for request i within one session, r_i is the time request i is made and l is the number of requests in the viewing session. s_i and e_i is the start and end hour of when the requested program i was originally broadcast.

The start of a session is defined by the first of one or several consecutive requests. The end of a session is defined as when an estimated program hold time is greater or equal to the requested program's duration.

$$(tp_{n+1} - tp_n) > (te_n - ts_n) \tag{8.2}$$

In (8.1), p_i for $i = l$, that is the last request in a session, has to be an educated guess, if not omitted. Omitting the last request in each session has effect on the number of requests per session, and the sessions will be shorter and more frequent since the majority of the last requests has to have an estimated duration of a full program view. In our case the study is aimed at browsing and thus this choice has little or no effect.

In Figure 8.1, the frequency of program hold times during the measurement period and the viewed program's full duration are plotted in a log-log diagram. The graph resembles one found in [63] where the frequency of channel hold time is plotted. The main difference is that in [63] TV channels are studied. TV channels have more or less infinite duration. Programs on the other hand are finite, which of course has an impact on the program hold time; the latter cannot be longer than the former. The two minor peaks between 10 minutes and 1 hour and the greater gradient of the second part of the plot are explained by this fact.

From the peak at 40 seconds and up to approximately 15 minutes the graph in Figure 8.1 follows a power-law distribution. The referenced plot in [63] has the peak at channel hold time 4 seconds, while in Figure 8.1 the peak is found at program hold time 40 seconds. One minute, as suggested by Cha *et al* and others, as an upper delimiter for zapping cannot be used here. Instead 300 seconds is proposed as more appropriate for this study.

A *zapping ratio* was calculated according to

$$R_a = \frac{Z_a}{N_a} \tag{8.3}$$

where Z_a is number of requests shorter than 300 seconds, herein called zapping request, and N_a is the total number of requests for an active account a.

Figure 8.3 shows the zapping ratio CDF for all users in the active users list. These users were then divided into the two user groups, zappers and loyals, according to the zapping ratio. The top 10% accounts with highest zapping ratio were designated as zappers (17385 accounts) and the 10% accounts with lowest zapping ratio as loyals (19899 accounts). The expressed gradients in Figure 8.3 point out that per each full program view the user makes one, two or three "zapping" requests.

Users are normally loyal to a few channels, see Figure 8.4, in this case limited by the sets of subscribed channels. From Figure 8.5 it can also be seen that there is a difference in the zapping ratio distribution between two of the channels in

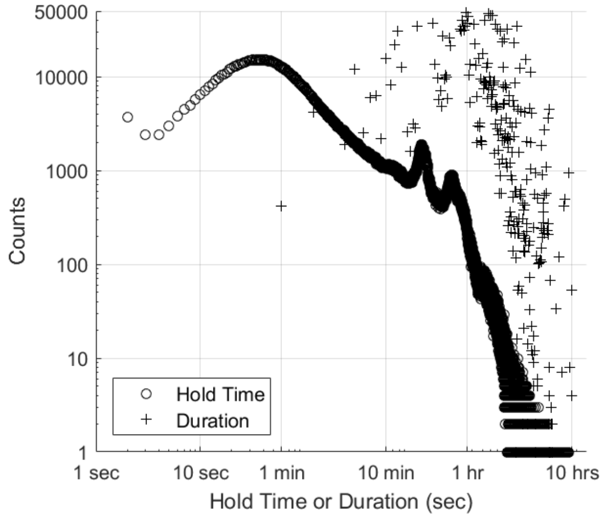


Figure 8.1: Occurrences of program hold times and requested program's duration

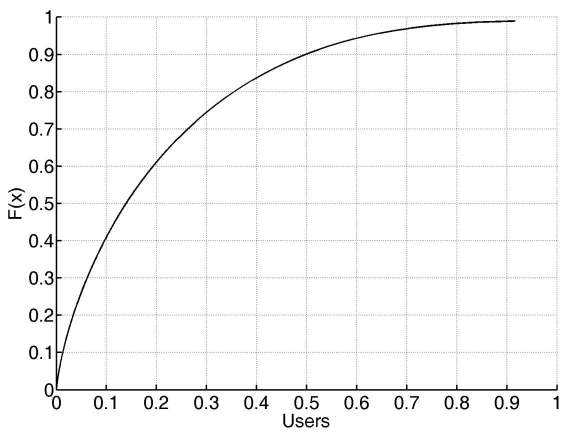


Figure 8.2: CDF of user's contribution to the total of number of request. The users are sorted descending regarding number of request over the full period.

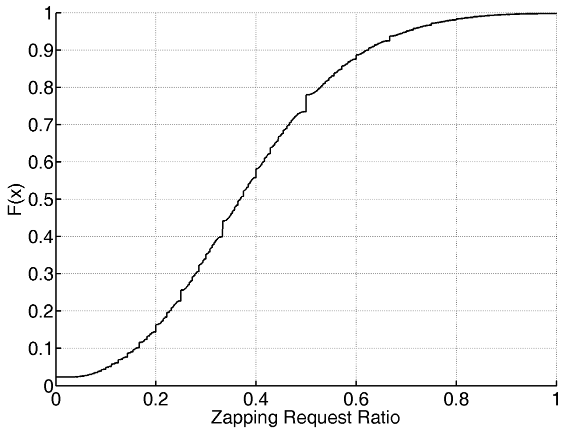


Figure 8.3: CDF of zapping ratio for all active users. The last request, the one that might have shown the program the user actually wanted to see, is not included.

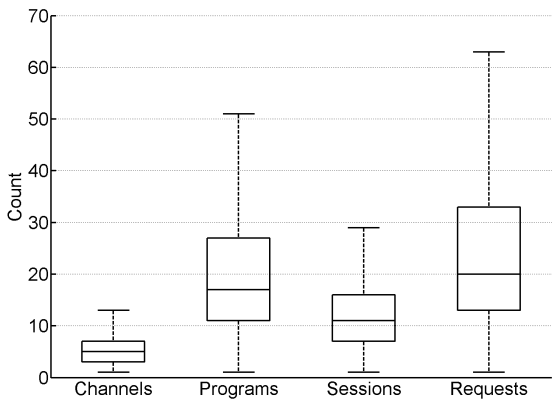


Figure 8.4: Number of channels, programs, sessions and requests per user. Outliers are removed.

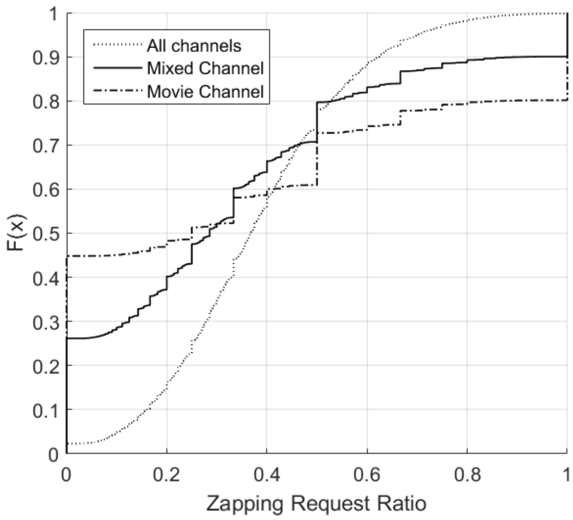


Figure 8.5: CDF of zapping ratio for two different channels.

the network, one movie channel and one with mixed content. All requests for the specific channel was evaluated per account. For the movie channel 45% of all accounts have no zapping request, while for the mixed channel the same behaviour is found in approximately 25% accounts. Therefore, it is of interest to compare also request patterns for individual channels. The channels were sorted according to zapping prone account frequency. The top 10% channels with the highest number of zapping prone accounts (9 channels) and the 10% of the channels with lowest number of zapping prone accounts (7 channels) were grouped.

8.2.2 A Quick Word on Intermissions

An intermission directly following a session break as defined in (8.2) can be included in the current session. The idea of omitting shorter intermissions between sessions is taken from web browsing evaluations. Measurable web browsing events are the requests, but a web browsing session contains also reading or studying the content of the request. Thus, pauses between two web browsing requests below a certain threshold value must be omitted and included in a browsing session. Göker and He conclude that intermissions less than 15 minutes can be omitted and included in web browsing sessions when evaluating web search logs. [73]

Table 8.2: Average and standard deviations over all sessions for zappers and loyals.

	Zapping Accounts	*Loyal Accounts*
Session duration (s)	2434.99 ± 1732.46	3171.64 ± 2463.81
Requests/session	1.92 ± 2.09	1.44 ± 0.92
Episodes/session	1.69 ± 1.48	1.37 ± 0.82
Episodes/Requests	0.95 ± 0.14	0.97 ± 0.11

Table 8.3: Average and standard deviations over all sessions for zapping and loyal prone channels.

	"Zapping" Channels	*"Loyal" Channels*
Session duration (s)	1935.19 ± 1216.61	5703.35 ± 3554.37
Requests/session	1.79 ± 1.62	1.37 ± 1.00
Episodes/session	1.61 ± 1.19	1.23 ± 0.64
Episodes/Requests	0.95 ± 0.13	0.95 ± 0.16

In this study, the duration of the last request in each session has to be an estimation. Adding a guessed intermission duration introduces even more uncertainty in the data and intermissions are therefore omitted.

8.3 Analysis

As can be seen in Table 8.2 and Table 8.3, zappers are more prone to watch several different streams per session than loyals. Zappers are more impatient and have shorter shorter sessions. The average of requests per session is nearly the same for the two user groups, but the standard deviation for zappers is double that for loyals. The difference in session length is pronounced when comparing zapping prone and loyal prone channels.

A note on self-fulfilling: Zappers have been defined to have a high ratio of requests with program hold times ≤ 300 seconds. It is obvious that this definition by intention introduces a bias regarding the number of shorter program hold times for the the two user groups, see Figure 8.6 at 300 seconds. It should also be noted that removing "noise" by excluding non-active accounts (see section 8.2) might impact on the number of loyals.

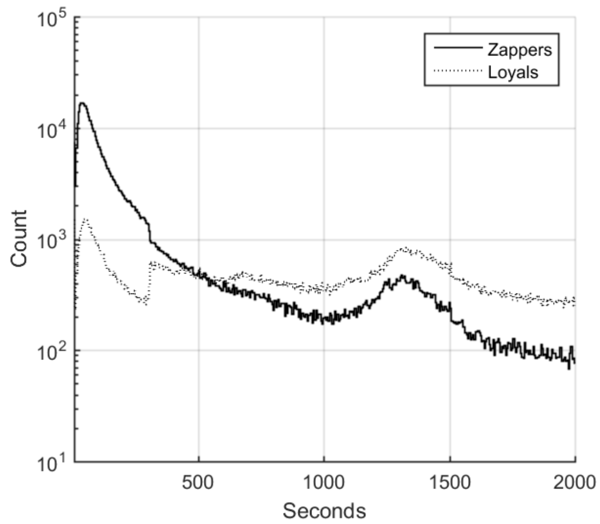


Figure 8.6: PDF for Program Hold Times. Note the logarithmic scale on the Y axis.

Table 8.4: Program hold time cumulative distribution from the study by Yu *et al.*[110]. Note that Yu *et al.* uses the notion *session* for what herein is called program hold time.

Program hold time	*5 min*	*10 min*	*25 min*	*50 min*
%	37.44	52.55	75.25	94.23

8.3.1 Program Duration and Hold Time

In Figure 8.7 the requested program's hold time is plotted together with the program's durations. Program hold times less than 15 minutes are more frequent than program durations less than 15 minutes. It can also be concluded that zapping events in the context of this study have longer duration than corresponding events in [63]. Another finding from Figure 8.7 is that 30% of all requests are for programs that are not viewed in their entirety.

Comparing the cumulative distribution of the program hold time with data from Yu *et al* [110] similarities can be found, see Table 8.4 and Table 8.5. In Table 8.5 the first row contains data from all requests while in the second row data from requests with program hold time greater than 60 minutes are excluded.

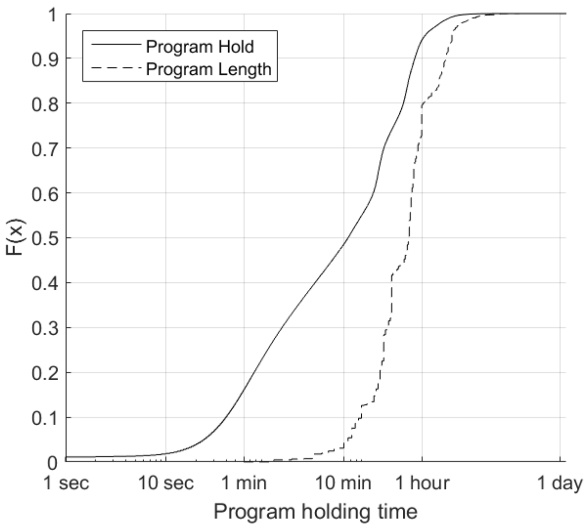


Figure 8.7: CDF of program hold time and requested program's duration over all requests for active users.

Table 8.5: Program hold time cumulative distribution

Program hold time	*5 min*	*10 min*	*25 min*	*50 min*
All session (%)	39.6	48.6	70.0	89.3
Sessions with duration 0-60min (%)	42.1	51.6	74.4	95.0

Table 8.6: Mean and median of *per session* user statistics for one extreme user account with all accounts. The last request per session is not included.

	Account	*All accounts*
Program hold time (mean)	191.4 sec	1370.8 sec
Program hold time (median)	96.5 sec	1042 sec
Number of programs (mean)	25.7	1.8
Number of programs (median)	26	1
Number of requests (mean)	109.3	2.1
Number of requests (median)	90	1

8.3.2 Extreme Account Example

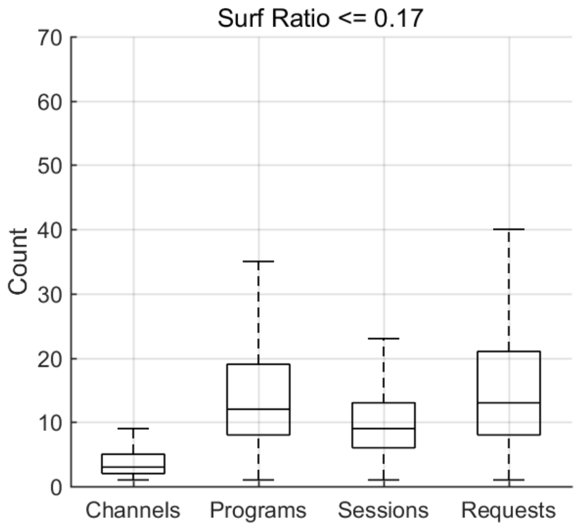
One user account is of special interest. This account has 25 sessions during the study's 23 days and a total of 2733 requests. The maximum number of requests for one session is 310! Statistics for this account were compared with all active accounts in the dataset in Table 8.6. The data conforms to findings by Abrahamsson *et al* in [47]. Most users have few requests per session and thus also few programs per session and the program hold time is long, while some accounts have sessions with many request, many programs and short program hold times.

8.3.3 Programs, Channels, Requests, and Sessions

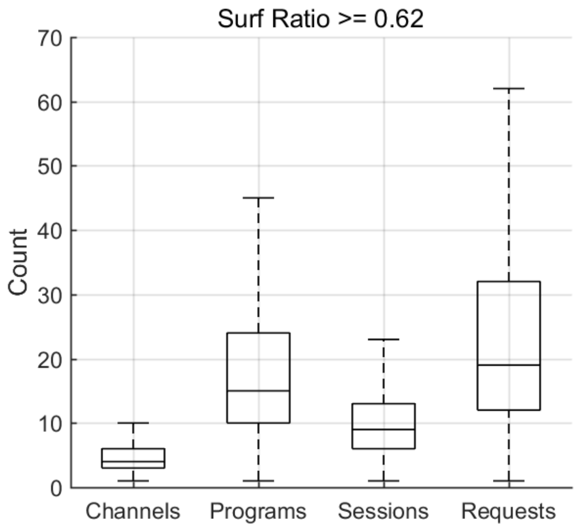
Studying the two user groups zappers and loyals render some interesting differences. Figure 8.8, which is plotted without outliers[1], show that the median number of requested programs among zappers is close to double that of the loyals; the spread is higher as are the extreme values. The same can be said about the number of programs per account, while the number of sessions and the number of requested channels per account are fairly equal in the two user groups.

Another way of presenting this data, at least regarding number of programs and requests per session in the two user groups, is found in the Cumulative Distribution Function (CDF) plots in Figure 8.9. The differences shown in Figure 8.8 can be identified, but on the other hand there is very little difference, if any seen from these figures, in the behaviour between the two user groups and also in comparison with all accounts.

[1] The outliers regarding both number of programs and number of requests are much more extreme among the zappers than the loyals.

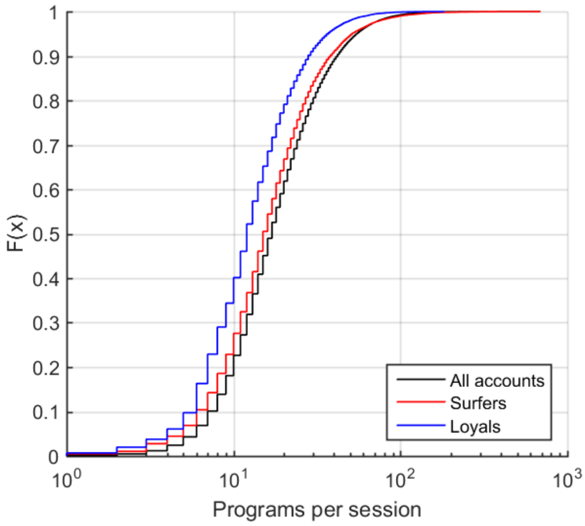


(a) Zapper accounts

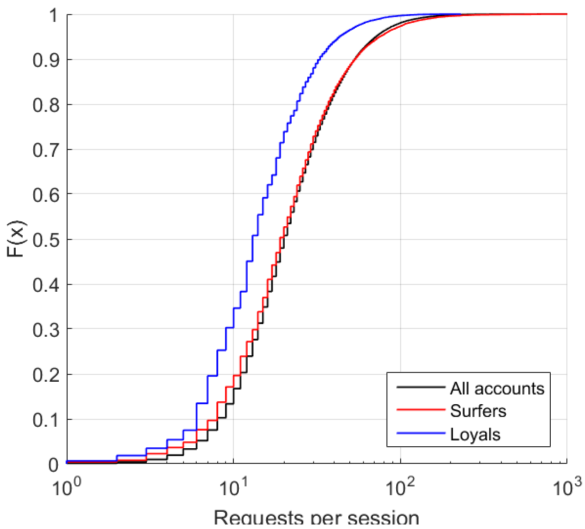


(b) Loyal accounts

Figure 8.8: Session data for zapper and loyal accounts according to program hold time and number requests. Outliers are removed.



(a) Programs per session



(b) Requests per session

Figure 8.9: CDF of programs and requests per session.

The "impatient user", defined by [49], exists in the sense that many requests consist of viewing only a part of a program; It is impossible to state from the dataset where in the program the view actually starts. From Figure 8.3 it can be conclude that for 50% of all active accounts, each third or more request is a zapping request. As can be seen in Figure 8.8(b), there are more zapping requests than individual programs in the sessions. This indicates that the same program is requested repeatedly in a session. Why this behaviour is seen is not studied here, but in some extreme cases it is even possible to imagine some sort of automata behind the account.

The program hold time in the studied network are longer than the corresponding channel hold time in legacy IPTV networks, as presented by [63]. This implies that requests that should be considered being zapping are longer in the studied network than in for example [63]. Compared to [49] and [110] where 20% of all requests last less than 30 seconds, only 5% of all requests in this study have a program hold time less than 10 minutes. An analysis of a Swedish dataset in [68] indicates that 20% of all requests have a duration less than 2 minutes.

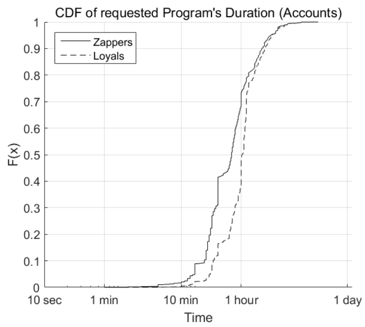
As in the study by Abrahamsson [47] most IPTV users request few programs, but some users have a high program demand. In the IPTV operator case 80% of the users request three programs or less over a selected day while approximately 2% of the users request more than 10 programs.

The program hold times in Figures 8.10(b) and 8.10(d) show differences between zapper and loyal accounts as well as for zapper prone channels and loyal prone channels. Some of this is due to self-filling. Even so, the findings in Figure 8.10 indicates that there is a minor difference in the chosen program's duration between zappers and loyals. The zappers are selecting a slightly different content than the loyals and tend to be requesting programs from other channels than loyals.

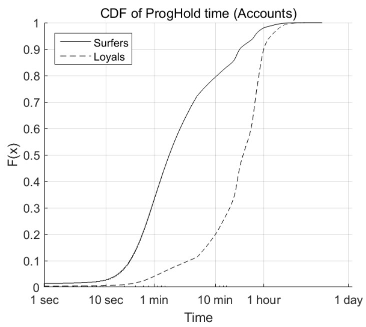
8.4 Pre-fetching and Caching

Pre-fetching shortens the time from requesting a program to start of play-out. Caching is beneficial in cases where content is re-requested either by another user (network caching) or by the same user (personal caching). It is thus motivated to study efficiency differences for the two user groups, zappers and loyals.

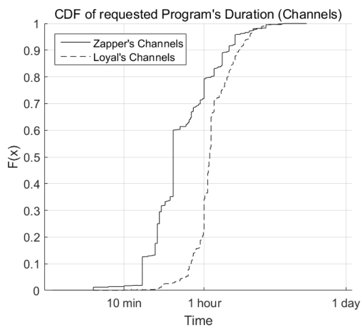
In [113] Zhang analyses the next events for one specific channel. By repeating these measurements for zappers and loyals separately over all channels in the network it is possible to make an analysis regarding different gains of prediction of episodes of the same series.



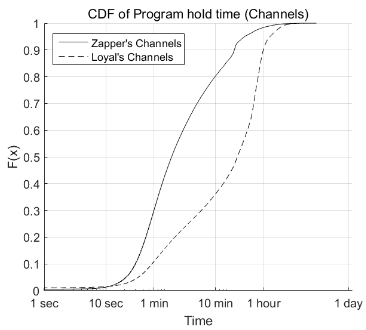
(a) CDF of program durations, accounts.



(b) CDF of program hold times, accounts.



(c) CDF of program durations, channels.



(d) CDF of program hold times, channels.

Figure 8.10: CDFs of program durations and program hold times per accounts and per channels.

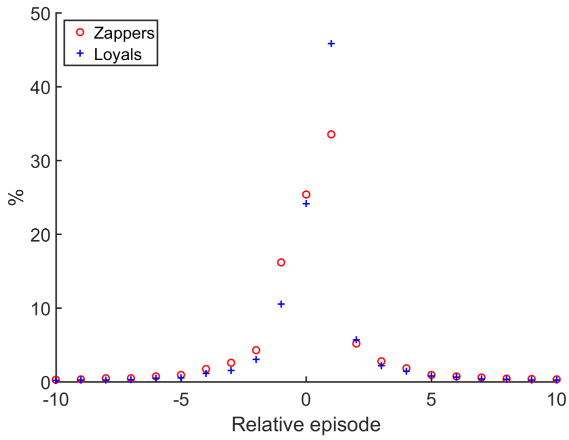


Figure 8.11: Next view event for zappers and loyals.

Figure 8.11 reflects the probability of a user watching episode X of a series as next event requests episode $X + i$ where $i \in \{-10, +10\}$. $i = 0$ corresponds to the cases where the user request the same episode again. Note that it is not possible to see from the recorded data where in an episode the user starts viewing. Events corresponding to $X + 0$ could very well be cases where the user has paused the viewing for a longer period.

Loyals have a higher tendency than zappers to go from episode X to episode $X + 1$ as shown by Figure 8.11. Zappers on the other hand requests previous episodes more likely than loyals. This could indicate that loyals are more pro-active by making more founded viewing choices, while zappers could to a higher extent be said to be more re-active and just pick a reasonable episode and change if the choice was wrong.

A notable remark is that in the current study more than 53% of the next viewing event for zappers is for another series than the current view and 45% for loyals; Zappers are less loyal to series.

Zappers gain slightly more from personal caching than loyals, Figure 8.12. The mean hit rate for zappers is 0.25 ± 0.16 and 0.15 ± 0.15 for loyals. This is in accordance with the findings in Table 8.2; Zappers have more request per episodes than loyals.

This study only takes zapping into account for a discussion on impact on pre-fetching and caching. It could be discussed if other intrinsic differences between user groups could motivate differentiating pre-fetching and caching strategies.

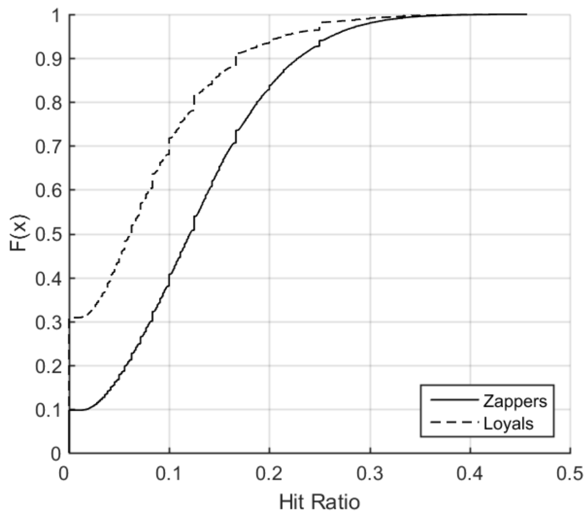


Figure 8.12: Personal cache hit ratio for zappers and loyals.

Overall, 80-85% of the sessions contain only one or two program requests, this indicating that most of the users are informed of what content is requested before starting a viewing session. Even so the analysis indicates that identifying the two user groups zappers and loyals is beneficial.

Chapter 9

Conclusions and Future Work

On the 29th of October 1969 the first message ever was sent over the ARPANET[1]. From thereon, we have, in a relatively short time span, witnessed an evolution in access to information and communication between people that is truly amassing. Not only new applications have been developed and deploy, but also have for the general public well-known applications been transferred to this new infrastructure, and the reuse of old and already available infrastructure have been made possible by new technology. These changes of technology are not always just positive. Well-known and therefore also anticipated application behaviour has changed; These new technologies bring, in most cases, better quality and fidelity to well-known application, but in many cases more severe negative impacts from disturbances. This is true not only for the end user, but also for cases when upper layer technology is transferred over to new or alternative lower layer transport links.

This thesis has dealt with some issues in this field. Studies of a disturbance's impact on packet transfer, both regarding available capacity and latency, is one example, how to increase, or at least maintain, the user's perceived quality of TV over Internet (IPTV) or Video on Demand (VoD) is another. Monitoring of quality performance parameters has an increasing importance, both for connectivity as well as for service suppliers and their customers; it is no longer possible to solve any performance problem with increased capacity. Quality of Service (QoS) and Quality of Experience (QoE) parameters on different levels in the Open Systems Interconnection (OSI) reference model have been identified, and monitoring thereof discussed.

[1] ARPANET is the packet switched network that was the basis for what from 1983 is known as the Internet

Monitoring Digital Subscriber Line (DSL) performance parameters gives a good estimate of the user's perceived quality in the single user case where the link is utilised close to maximum. This is however not enough for Orthogonal Frequency-Division Multiplexing (OFDM) technologies like Very-high-bit-rate Digital Subscriber Line version 2 (VDSL2) and G.fast, where also utilisation has an impact on for example the probability of packet loss.

DSL links can adapt to long term narrow band disturbances, i.e. broadcast radio stations or corrupt switched power supplies. Momentary onset of such a disturbance will cause changes in the bit loading per carrier, i.e. bit swapping. This will in turn affect the transport of datagrams with the odd packet loss, but after the initial phase the channel has adapted to a lower capacity. The latency is not affected, unless of course the required capacity is greater than the available. The DSL channel does not restore itself to a situation before the disturbance onset, which means that the bit loading caused by the disturbance is kept unchanged. Thus, a shorter narrow band disturbance affects the channel's capacity in the long term. The effect of disturbances with a low energy gradient regarding time, like radio signals mirrored in ionospheric layers, are not studied, but could be a matter for future studies.

Short term wideband disturbances, Electrical Impulse Noise (EIN), impact on datagram transport over DSL links has also been studied. EIN is no cause for bit swapping, but destroys one or two Multi-Carrier Modulation (MCM) frames, be it Digital Multi-Tone (DMT) or OFDM. This in turn causes bit errors in one or two datagrams, if the EIN hits an MCM frame carrying a valid datagram. The probability of packet loss is shown to not only be dependent of the EIN pulse frequency, but also the utilisation of the DSL channel.

Neither of these two disturbance cases have in this thesis lead to a suggestions of specific fingerprints related to typical disturbances. This could of course be a matter for future studies.

In the near future it is expected that available copper based access networks will be used in a multi-user environment, both for mobile backhaul and for packet based fronthaul, e.g. Common Public Radio Interface (CPRI), for small cell deployment. Not only the channels' capacity but rather the latency is of major interest for mobile front-hauling in Fixed and Mobile Converged (FMC) networks. The impact on Packet Delay Variation (PDV) from EIN, specifically in combination with the DSL system's Impulse Noise Protection (INP) functionality i.e. Forward Error Correction (FEC) and physical layer retransmit, has not been studied, but is in this context of great interest. The results point out the importance of performance monitoring and the understanding of cross layer interaction. This is probably one of the key factors for a successful FMC network, especially when considering performance management for service assurance. In this thesis, only the effect of disturbances in a single user use case is studied. In FMC networks, the situation is more complex and future work could explore these multi-user scenarios.

An identified knowledge gap is the art of performance monitoring measurements in networks with asymmetric paths like mesh networks. Another identified possible knowledge gap is the outstanding question if Internet Protocol version 4 (IPv4) and Internet Protocol version 6 (IPv6) behave differently and thus should be handled individually.

On the application level, the impact zapping has on pre-fetching and caching policies beyond the general strategy has been studied. Not only can zapping prone subscribers be identified, but also zapping prone channels exist. Thus, profiling of both subscribers *and* channels enabling adaptation of e.g. playout delay and pre-fetching and terminal or personal caching strategies per profile is suggested as a mean of increasing subscriber QoE and satisfaction. The outcome of introducing profiling has to be further analysed, though. The finding in [53] of self-generating requests is not studied here but needs to be included in a further strategy study.

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