NTNU – Trondheim
Norwegian University of
Science and Technology

Reliable Broadcast Contribution over the Public Internet

Martin Alexander Jarnfeld Markman
Stian Tokheim

Master of Science in Communication Technology
Submission date: June 2012
Supervisor: Andrew Perkis, IET

Norwegian University of Science and Technology
Department of Electronics and Telecommunications

Problem description

In professional broadcasting there is a need to transfer content from recording to editing, between studio facilities and other point to point transfers. This process is referred to as contribution. The common practice for contribution by the broadcast industry today is by using dedicated IP networks, guaranteeing a high quality lossless transfer. However, this is costly and not always available at all points and sites of interest. This thesis should investigate if and how contribution can be done over the worlds largest IP-network, the Internet. More specifically the thesis shall provide answers to whether broadcast contribution can be realized in the public Internet at a reliability and bandwidth level meeting contribution requirements to both Quality of Service (QoS) [1 - QoS] and Quality of Experience (QoE) [2 - QoE].

This should be answered in three phases.

- First, the thesis should provide a description about the state of the Internet in Norway and how it is likely to evolve in the next few years. This should be based on up to date information and interviews with the industry.
- Secondly, typical network behavior and network QoS should be measured and described through network testing of a link. This should lead to a discussion about reliability and different ways to ensure reliability.
- Third, contribution over the Internet should be tested with equipment made for professional contribution over IP contribution networks. The QoE provided to the end user by contribution should be described and discussed in relation to the network QoS.

[1 - QoS] ITU-T, G.1050: Network model for evaluating multimedia transmission performance over the Internet protocol, November 2007

[2 - QoE] Qualinet, Qualinet White paper on Definitions of Quality of Experience, May 2012

Abstract

Broadcast contribution is point-point media transfer from recording sites to local editing studios, between studio facilities and to distribution centers. The contribution phase has strict Quality of Service (QoS) requirements to reliability and bandwidth - any error might degrade end users Quality of Experience (QoE) in the consecutive distribution phase. Dedicated IP contribution networks has become the preferred technology for contribution from content creation sites. Occasionally, however, contribution happens at a site without access to a dedicated IP contribution network - in which case the broadcaster must utilize less optimal technologies. Based on IP, the public Internet may be a superior solution in some scenarios due to high bandwidth and geographical coverage - *if the internet can conform to strict contribution requirements to QoS and QoE*. This thesis attempts to give a clear answer to this question.

Our investigation was done in three parts. First, we uncovered recent Internet QoS trends in Norway. We found that the internet has become an Internet video delivery platform, which in turn has resulted in bandwidth increase in access networks. Bandwidth in residential access links now conforms to contribution requirements. ISPs make profit according to the level of offered QoS, broadcasters can therefore expect high QoS. Also, broadcasters can buy QoS guarantees, which may be a viable and safe solution.

Secondly, we recorded over 21 hours of Internet QoS statistics on a connection traversing 11 routers and one peering point. The measured level of every QoS metric (packet loss, jitter and re-ordering) conformed to professional contribution network requirements, except the rate of packet loss bursts. However, no burst above 200 ms was recorded and no two consecutive bursts happened within a 2 second time frame. Based on this, we explained how simple error control strategies can correct or mask packet loss burst with a 200-250 ms delay tradeoff and 15-30% bandwidth overhead.

Third, we did subjective tests in the Internet with two professional JPEG2000 contribution gateways delivered by T-Vips. A full movie was encoded at 70 Mbit/s, a bitrate used for very high quality contribution, and shown to a test panel of 24 participants. By analyzing questionnaires, we proved that contribution over the internet yield equally good QoE as cable TV. Also, we found that noticeable degradations due to packet loss happened once per hour on average. Furthermore, packet loss bursts below 4 ms was generally not visible to the viewers.

Because the Internet provides both the required QoS and QoE, we concluded that broadcasters can do contribution over the Internet at the required level of quality whenever this is a favorable option.

Samandrag

Kontribusjon for kringkasting er transport av audiovisuelt innhald frå plassen innhaldet vert laga til eit distribusjonssenter. Kontribusjonsfasen har strenge Quality of Service (QoS)-krav til pålitelegheit og bandbredde - ein kvar feil kan føra til dårlegare Quality of Experience (QoE) for sluttbrukar. Tileigna IP-kontribusjonsnettverk har blitt den føretrekte teknologien for sending av kontribusjonsinnhald frå plassen innhaldet vert laga. Men, tidvis skjer kontribusjon frå plassar utan tileigna IP-kontribusjonsnettverk - då må kringkastaren bruka mindre optimal teknologi. Internettet, som er basert på IP, kan vere ein overlegen løysing i somme tilfelle på grunn av høg bandbredde og geografisk dekning - *dersom Internettet kan fungera i samsvar med strenge krav til QoS og QoE*. Denne oppgåva prøver å gje eit klårt svar på dette spørsmålet.

Granskinga vår vart gjort i tre delar. Først avdekte vi nylege trendar for Internett QoS i Noreg. Vi fann at Internettet har blitt ein plattform for levering av sanntids video, som i sin tur har ført til ei auking i bandbredde som er i samsvar med kontribusjonskrav. Internettleverandørane tener pengar i samsvar med nivået av QoS, derfor kan kringkastarar forvente høg QoS. I tillegg kan kringkastarar kjøpe QoS garantiar, noko som kan vere ein levedyktig og trygg løysing.

For det andre, så registrerte vi 21 timar med Internett QoS statistikk på ein kopling som traverserte 11 ruterar og eitt samtrafikkpunkt. Resultata viste at Internettet samsvarer med krav til kontribusjonsnettverk når det gjeld pakketap, jitter og endra rekkefølgje på pakkane. Pakketap-raten er hovudutfordringa. Men, ingen pakketap-serie over 200 ms vart registrert, og to pakketap-seriar oppsto aldri innafor ei tidsramme på 2 sekunder. Basert på dette, viste vi at ein enkel feilkontroll-strategi kan vera i stand til å rette alle feil med ein 200 ms forseinking som kostnad.

Som eit tredje punkt, så gjorde vi subjektive testar i Internettet med to profesjonelle JPEG2000 (Intra-koding) kontribusjons-gatewayar levert av T-Vips. Ein heil film vart koda på 70 Mbit/s, som er klassifisert som høveleg for veldig høgkvalitets-kontribusjon, og vist til eit testpanel beståande av 24 deltakarar. Ved å analysere spørjeskjema, viste vi at kontribusjon over Internettet gir like god QoE som kabel TV. I tillegg fann vi at merkbare feil på grunn av pakketap skjedde ein gong i timen i gjennomsnitt. Vidare fann vi at pakketap-seriar under 4 ms generelt ikkje var synlege for sjåaren.

Fordi Internettet gjev både den QoS og QoE som krevs, har vi konkludert med at kringkastarar kan gjere kontribusjon over Internettet med det nivået av kvalitet som krevs når det måtte vere eit gunstig alternativ.

Contents

Nomenclature

ACK	Acknowledgment
ADSL	Asymmetric Digital Subscriber Line
ATM	Asynchronous Transfer Mode
AVC	Advanced Video Coding
BGAN	Broadband Global Area Network
CAGR	Compound Annual Growth Rate
CDN	Content Delivery Network
CPU	Central Processing Unit
DC	Digital Cinema
DCT	Discrete Cosine Transform
DOS	Denial Of Service
DTH	Direct-To-Home
DVB	Digital Video Broadcasting
DVB-T	Digital Video Broadcasting — Terrestrial
DVD	Digital Video Disc
FEC	Forward Error Correction
FTTH	Fiber To The Home
FTTx	Fiber To The x
HD	High Definition
IFs	Influence Factors
IP	Internet Protocol
IPv4	IP version 4

IPv6	IP version 6
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
ITU	International Telecommunication Union
ITU-R	ITU Radiocommunication Sector
ITU-T	ITU Telecommunication Standardization Sector
KLT	Karhunen–Loève Transform
MAC	Media Access Control
MPEG-2 TS	MPEG-2 Transport Stream
MPLS	Multiprotocol Label Switching
MTU	Maximum Transmission Unit
MVs	Motion Vectors
MXF	Material Exchange Format
NTSC	National Television System Committee
OB	Outside Broadcasting
OSI	Open Systems Interconnection
PAL	Phase Alternating Line
PBLR	Packet Burst Loss Rate
POC	Proof Of Concept
PSNR	Peak Signal-to-Noise Ratio
QoE	Quality of Experience
QoS	Quality of Service
RTP	Real-time Transport Protocol
RTT	Round Trip Time
SD	Standard-Definition
SDH	Synchronous Digital Hierarchy
SDI	Serial digital interface
SMPTE	Society of Motion Picture and Television Engineers
SNG	Satellite News Gathering

SONET	Synchronous Optical Networking
SSB	Statistics Norway
TCP	Transmission Control Protocol
TOS	Type Of Service
UDP	User Datagram Protocol
VDSL	Very-high-bit-rate Digital Subscriber Line
VLAN	Virtual Local Area Network
VoD	Video on Demand
VPN	Virtual Private Network
VSF	Video Service Forum

List of Figures

List of Tables

Chapter 1

Introduction

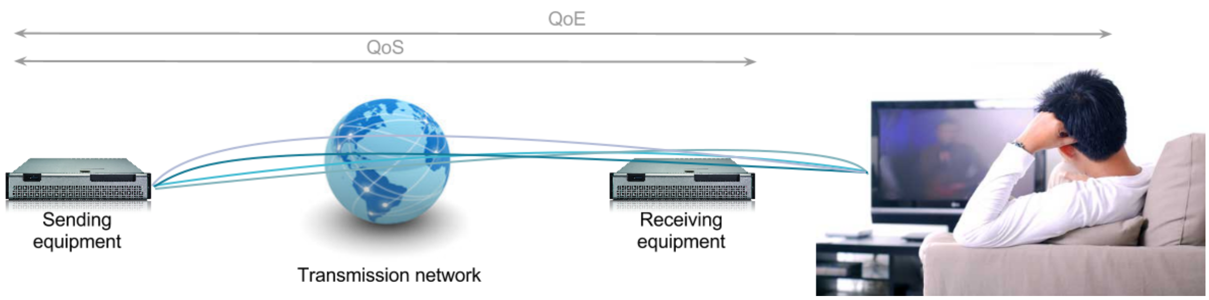
Over the last 40 years - television broadcasting have evolved from delivery of low-quality black-and-white content to crystal-clear HDTV and 3D-TV. Meanwhile, television as an amusement and information delivery platform has become vital for people. Watching TV triggers our emotions - we laugh at funny comedians, we are shocked from breaking news and we cry and empathize with people telling their sad stories. Have you ever caught yourself standing in your couch in front of the TV screaming at your favorite football team?

Broadcast of audiovisual content is actually a chain of actors, technologies and equipment [1]. At one end, the content is produced by creative minds - while at the other, end users enjoy content on various media platforms. In between we find the broadcasting system chain; camera operators, production teams (onsite or at a remote location), transmission operators, network engineers, post-production teams and commercial distributors. In general, the content is captured, produced and edited before transmitted to end users in the broadcasting chain.

Money flows from the end users to the broadcasters and to the creators. Content creators gain profit according to the level of end user amusement. Broadcasters, or providers, gain profit according to the level of Quality of Experience (QoE) offered to the consumer.

QoE is a wide term capturing user experience or delight of the delivered service [2]. User experience is determined based on more than just error-free audiovisual stream with high quality. Aspects like availability (from where, to where and to what terminal platform), economy, and contextual information like the users personality and state-of-mind also determines QoE. QoE - as the degree of delight - is therefore what makes end users willing to pay the broadcasters.

From the broadcasters point-of-view, Quality of Service (QoS) is crucial in order to deliver high QoE [3][4][5]. QoS is merely about network impairment metrics: packet delay, throughput (bandwidth), loss and jitter. Media loss and video quality degradation caused by network impairments degrade QoS - such errors directly affects end user QoE. Thus, the QoS term measures the broadcasters ability to deliver services with reliability and high quality.



If you actually recall standing in your couch screaming at your favorite football team - you perceived a high level of QoE in that moment. Indeed, high QoS is fundamental as a first step in delivering high QoE. However, you probably would not be jumping up and down in happiness because the packet loss ratio is below 0.05%. To be clear, there is a clear correlation between QoS and QoE - but QoE is ultimately what matters for the end users.

The broadcasters task is to design and operate a chain of systems delivering content maximizing QoS and QoE. This chain of systems, or the broadcasting system chain, is commonly divided into two phases; *contribution* and *distribution*. In the contribution phase, the content is transferred from the content creation site - be it a sporting event, a news coverage or a studio show - to a distribution center. In the distribution phase, the content is distributed to end users on various media platforms like TV, PC and smart phones, using distribution technologies such as satellite (DTH), terrestrial links (DVB-T), cable, Internet and wireless networks - as illustrated in figure 1.1.

This thesis is exclusively focused on the contribution phase. Contribution is basically a live point-point media transfer. The transfer can be between *permanent* locations, e.g. between a local studio and a distributor studio as illustrated in figure 1.1, or from *occasional*[1] creation sites such as a sporting event to a local studio. The overall goal in the contribution phase is to be completely transparent to the end users. This transparency is ensured by keeping the media transfer error free such that no error propagate into the distribution phase, affecting the viewing experience.

Internet Protocol (IP) networks is now the preferred contribution technology. Broadcasters implement dedicated IP contribution networks in which they have complete control. The packet-switching nature of IP gives clear advantages over traditional circuit-switching networks due to flexibility in terms of carrying a range of different content, re-routing of traffic in case of link breakdown and the ability to convey remote control and instructional traffic to the creation site. Compared to other contribution technologies like satellite (SNG) and microwave transmission - IP offers a superior level of bandwidth and minimal delay.

[1]The term 'occasional contribution' is used in the broadcasting industry to denote contribution from temporary content creation sites.

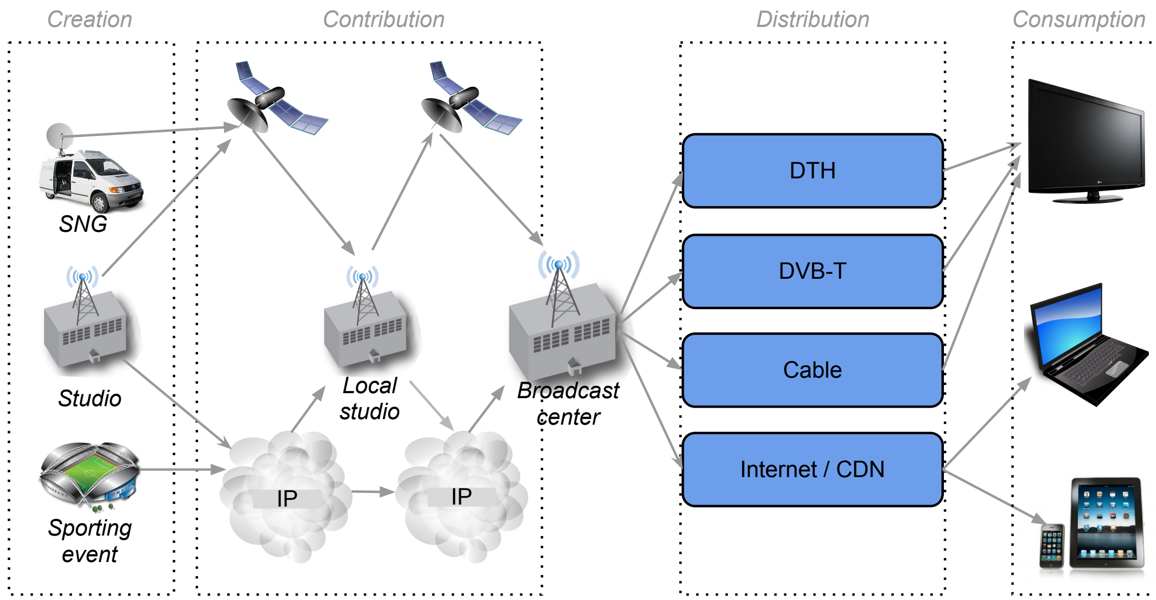


Figure 1.1: The role of contribution and distribution in the broadcasting system chain.

The high bandwidth in IP networks keeps compression of the audiovisual content to a minimum. JPEG2000 for contribution over IP has recently proved to be the technology of choice for many large broadcasters around the world - like NRK, SKY, FOX, CBS, RAY, and so forth [6]. T-Vips, a Norwegian contribution equipment vendor, have sold JPEG2000 IP gateways[2] many broadcasters around the world. Compared to other codecs, such as MPEG-2 and MPEG-4 AVC, JPEG2000 offer almost lossless compression - avoiding introduction of coding artifacts prior to re-encoding in the distribution phase [7]. Any quality degradation due to transcoding may cause perceptual degradation.

The challenge faced by the broadcasters is not merely choosing the optimal contribution technology - it is about *choosing the optimal contribution technology at the given content creation site* [8]. High-performance IP contribution networks is unavailable at some content creation sites such as minor sporting events, news coverage sites and concerts. The cost of implementing an IP contribution network at these location cannot be justified - as a rule of thumb, 5-10 events must happen annually at a location to justify this cost [7].

In this scenario, the broadcaster must use other contribution technologies than IP such as satellite or microwave transmission. Compared to IP contribution, these solutions have advantages due to geographical coverage and deployment time, but clear drawbacks in terms of flexibility, cost efficiency and bandwidth. As a consequence from the lack of bandwidth, the content must be produced and edited onsite in production trucks. These vehicles are expensive to rent and operate, and the planning and deployment time is also considerable.

Meanwhile, the public Internet[3] - the world largest IP network - has become an inevitable communi-

[2] An IP gateway, in this context, is a video encoder/decoder and IP packer setup box. It can serve as both sender and receiver in an IP network.

[3] From this point, "the public Internet" is referred to as "the Internet".

Figure 1.2: A football match is a typical occasional contribution site. Image found at fotballmagasinet.no.

cation platform for all. It is hard to imagine how our life would be without the Internet, even though this was the reality only a couple of decades ago. During this life span, the Internet has evolved from elastic content exchange, e.g. document and e-mail exchange, to delivery of bandwidth consuming services in real-time to end users. This puts demand on Internet QoS - Internet Service Providers (ISP) must deliver Internet traffic with sufficient QoS to keep customers happy.

Due to the recent emerge in Internet QoS, using the Internet as a contribution network for occasional contribution has most likely crossed every broadcasters mind. In theory, there are no limitations with this solution - that is, professional IP contribution equipment can be used in the Internet. As a concept, IP contribution in the Internet has the inherit advantages of professional IP - bandwidth and cost efficiency - but in addition has the benefit of high availability and fast deployment[4]. Contribution in the Internet can therefore be a attractive solution for broadcasters.

Most likely, using the internet as a contribution network for occasional contribution has crossed every broadcasters mind. In theory, there are no limitations with this solution - that is, professional IP contribution equipment can in principle be used in the internet. As a concept, IP contribution in the internet has the inherit advantages of professional IP - bandwidth and cost efficiency - but in addition has the benefit of high availability and fast deployment[5].

In practice, the question is whether reliability[6] and bandwidth in the Internet conforms to contribution requirements. If not, then none of the benefits mentioned above matters - the Internet would not be contribution ready.

[4] In principle, once connected to the internet IP contribution is "plug-and-play".
[5] In principle, once connected to the internet IP contribution is "plug-and-play".
[6] Reliability is the probability that an item can perform a required function under stated conditions for a given time interval (ITU-T E.800).

As far as we are aware, using the Internet as a contribution network has never been described on an academic platform. What is missing is therefore a close study of Internet reliability and bandwidth with respect to contribution - is the Internet contribution ready?

The motivation for this thesis is to be the first academic work on contribution in the Internet. Describing how a professional JPEG2000 IP contribution system perform in the Internet - as a proof-of-concept - can be first step in such an introductional study. Will the system work in the Internet? Recall that the goal in contribution is transparency in terms of absence of media degradation - how transparent will a contribution system in the Internet be for the end users? What will be the resulting QoE?

Our prior work showed that the Internet in Norway conformed to contribution requirements in short tests [9]. Internet statistics were collected from testing a handful of internet connections over a short time (minutes). This allowed comparison of recorded statistics and contribution requirements. What is missing from this study is longer tests (hours), capturing life-cycle Internet statistics. Also, a better understanding of Internet reliability; how is the commercial Internet evolving in terms of performance?

1.1 Scope

We will only consider the Internet in Norway, although the results possibly applies in other countries. We will only consider contribution of video, and leave audio unconcerned. We will conduct contribution tests using JPEG2000 contribution equipment. However, we will not evaluate JPEG2000 video quality. Only the effect of packet loss will be measured, and therefore the results may apply for other setups as well. We will only use the Internet in our testbed, avoiding use of models or simulations to test the effect of packet loss for end users.

The advantages using the Internet for contribution will be discussed, but it is outside the scope of this thesis to define realistic broadcasting scenarios or use-cases to motivate or justify our research topic. It is our belief that limiting this thesis to a specific scenario[7] with specific requirements, would also limit this thesis value. In fact, we see it the other way around; the outcome of the general and opportunistic investigation in this thesis can motivate broadcasters to utilize the Internet as a contribution network in scenarios they find suitable.

[7] With scenario, we mean a specific application (such as a football game) at a specific geographical placement.

1.2 Purpose

The purpose of this thesis is formulated in our research question;

Can contribution be realized in the Internet at a reliability and bandwidth level conforming to QoS[8] *and QoE*[9] *contribution requirements?*

Due to the lack of references or advice from previous studies, this thesis serves the purpose as an introduction thesis on contribution over the Internet. We search for a better understanding of the problem, highlighting challenges - but also opportunities - in this study. We will not pursue and solve every challenge here - rather, we will describe and discuss the overall indications and results which may stimulate further research in this fresh domain.

For clarity, we have chosen to divide our research question into three subquestions;

What is the state of Internet performance in terms of bandwidth and reliability?
The Internet is well known as a communication platform, but unknown in terms of the level of expected reliability and performance. We want to investigate how Internet performance is evolving in the commercial domain - is the Internet dimensioned and designed such that contribution can be done? This thesis will try to uncover some recent performance trends and *not* treat Internet as a black box. With such knowledge, we can also explain and generalize results from conducted contribution tests.

Can a contribution packet stream conform to contribution requirements?
This question takes on Internet performance measurements. By recording hours of packet QoS metrics (such as loss and delay) in the internet, conformance to contribution QoS metric requirements can be stated. Furthermore, the aggregated QoS metrics will point to the most common, and therefore most critical, network impairments in the internet given an IP contribution stream. Having found the most critical impairments, this thesis will point to the most important transport protocol features and also the discuss the choice between UDP and TCP as transport protocol for contribution in the Internet.

Does contribution over the Internet meet TV viewers expectations in the distribution phase?
We want to setup a real contribution system in the Internet and record both system behavior and subjective experience. Not only as a proof-of-concept, we want to prove that contribution over the Internet yield a TV service conforming to TV viewers exceptions in the distribution phase[10]. We will use the Quality of Experience (QoE) term to capture subjective feedback. It is our belief that recording QoE data from a real contribution test is a vital supplement to QoS recordings.

We will therefore investigate Internet contribution both in terms of QoS and QoE (see figure 1.3). Together, the findings from both investigations will be discussed and compared - and finally answer our main research question.

The thesis is organized the following way:

[8] Quality of Service. QoS converge towards network performance.

[9] Quality of Experience. QoE is not a well-defined concept. We will define our interpretation and application of QoE later in the thesis.

[10] Given an error-free distribution phase.

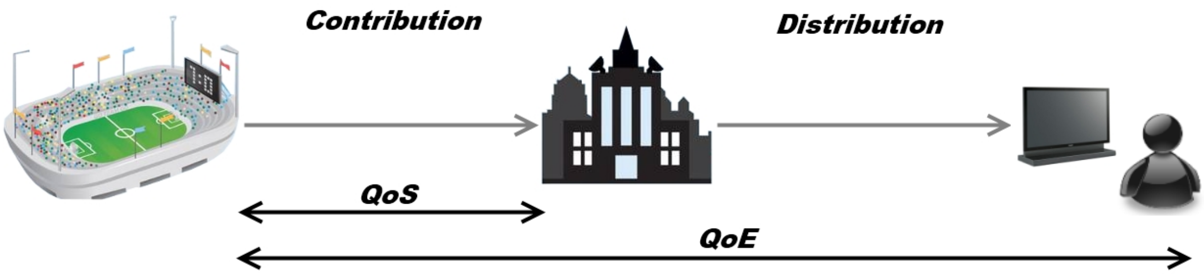


Figure 1.3: This thesis will concentrate both on QoS and QoE.

Chapter 2: The broadcast industry and contribution - gives a quick introduction to the broadcasting industry and how contribution is used.

Chapter 3: Theory - gives an overview of the theoretical background of the thesis.

Chapter 4: The Internet - the worlds largest IP network - presents the answer to "What is the state of Internet performance in terms of bandwidth and reliability?" based on available information and industry interviews.

Chapter 5: Internet performance measurements - answers "Can a contribution packet stream conform to contribution requirements?" by presenting the method and results of the network testing.

Chapter 6: Testing video contribution over Internet - describes how we did, and the results we got by doing, tests with the TVG video gateways. This answers "Does contribution over the Internet meet TV viewers expectations in the distribution phase?".

Chapter 7: Discussion - briefly discuss some of the implications of seeing the results from chapter 4, 5 and 6 collectively.

Chapter 8: Method critique - discusses the known shortcomings of the methods used in chapter 4, 5 and 6.

Chapter 9: Conclusion - sums up the most important conclusions by providing a short answer two the three subquestions and the one main question that defines the purpose of the thesis.

Enjoy!

Chapter 2

The broadcast industry and contribution

This chapter will introduce the reader to broadcasting systems and architecture. Furthermore, a presentation of contribution in the broadcasting system chain is given. This will highlight shortcomings in contribution systems used today, and in turn explain how the Internet as a contribution network may overcome some of these shortcomings - justifying our research.

2.1 The Broadcast contribution and distribution model

As mentioned, the broadcasting system chain in broken up into two separate parts, namely *contribution* and *distribution*. It is important to realize that both contribution and distribution most often is a collection of different systems. For example, in the contribution phase, the raw content is transported between a content creation site and a local editing studio, and consecutively transported to a national TV center. In the distribution phase, the content may be transported to a transcoding center, before transported to a satellite uplink station. All of these transporting systems have different characteristics and requirements.

At a top level, it is sensible to divide contribution systems into two classes, namely *permanent contribution* and *occasional contribution*. Figure 2.1 illustrates an broadcasting system chain and these two classes. Note that, contribution is usually a one-one system while distribution is one-many.

Permanent contribution

Permanent contribution refers to point-to-point contribution between permanent locations. Examples are between a local TV-studio and a TV-headquarter, or from a TV-headquarter to a local distributor system. These contribution links are permanent as they are constantly sending content between permanent locations. Furthermore, these links have very high bandwidth to avoid quality degradation of the content due to compression. Obviously, permanent contribution systems must be very reliant and stable - any link failure in the contribution phase propagates into all distribution services.

Traditionally, broadcasters often utilized ATM switching technique for permanent contribution of audiovisual content [4]. Even though ATM networks has the benefit of low risk for data loss due to rigid rules for insertion of ATM cells in SONET/SDH frames, IP is now the preferred network

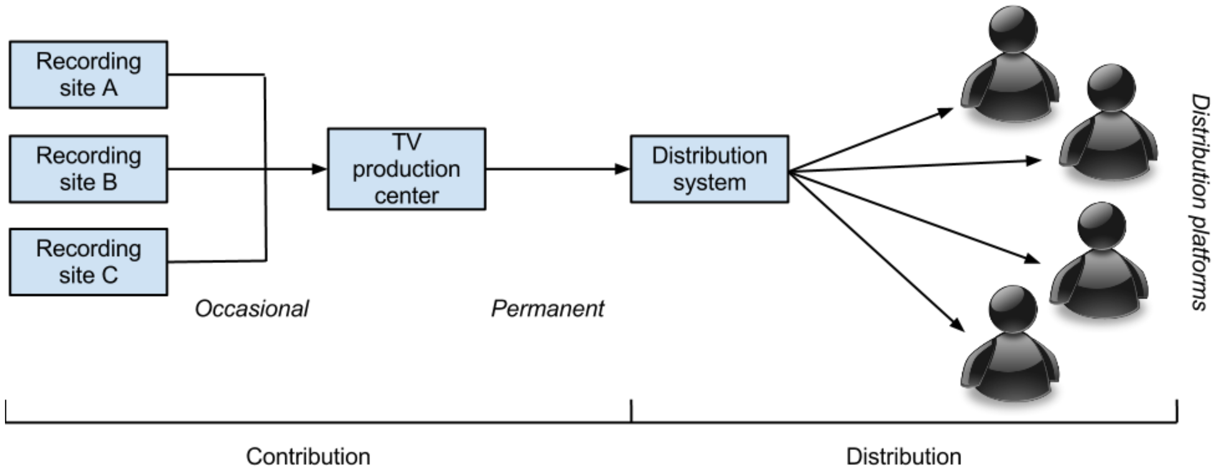


Figure 2.1: Illustration of the broadcasting system chain.

protocol for high quality video contribution. This is first of all because IP equipment is cheaper and able to carry any type of content. As far as reliability is concerned, IP can perform as well as ATM when the uncertainties associated with IP are explicitly controlled.

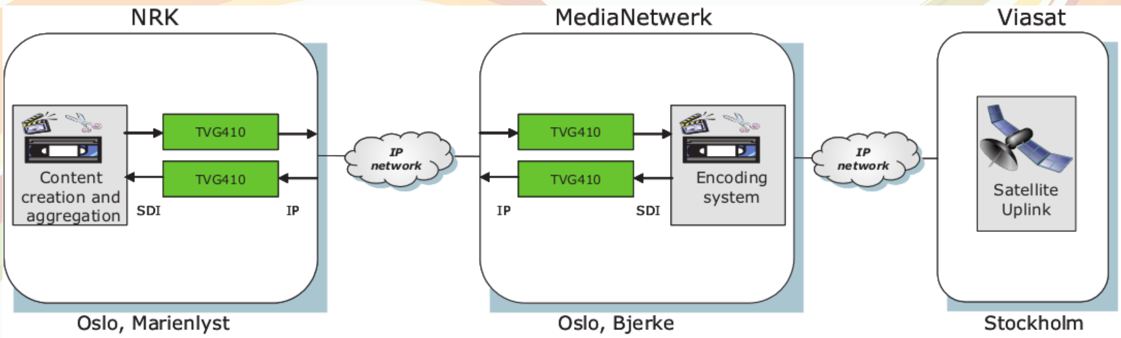
IP contribution are done over dedicated IP contribution networks[1] owned by the broadcaster, or outsourced to a contribution network provider firm. These networks are very closely designed and monitored to avoid network degradations such as media loss or loss of synchronization. In addition to high bandwidth and reliability, the advantages with a contribution *network*, as opposed to a single contribution link, is the ability to reroute the traffic in case of link breakdown. IP also allows bi-directional traffic; control traffic, e.g. remote camera control and adjustment optical camera settings, and communication, e.g. questions to an on-site reporter or instructions to operators, may flow the opposite direction of the media content in the network.

In Norway, IP contribution technology is used on a everyday basis by broadcasters [1]. For example, Media Netwerk, a contribution network operator, provides contribution services to NRK[2] based on IP technology; content for distribution is conveyed from NRK headquarters at Marienlyst in Oslo, to RiksTV[3] distributor centers in the districts. IP contribution networks have proved to be excellent in terms of uptime and reliability. On the next page, a case study of the complete broadcasting system chain for the TV-channel "SportN"[4] is presented to give a specific and accurate example.

[1]From hereon known as just "IP contribution networks".
[2]Norwegian Broadcasting Corporation - the largest public broadcaster in Norway.
[3]National television distributor company in Norway.
[4]SportN was closed in 2009, but the case highlight a common practice.

Case study - SportN

In 2005, NRK and MTG joined forces to launch a new sport channel called "SportN". The channel set to distribute Norwegian sports over satellite systems owned by Viasat. The partners decided that the content should be compressed and transported from the NRK studio in Oslo to the satellite uplink in Stockholm , from where it was distributed over DVB-S systems.



NRK chose IP-technology for contribution, provided by MediaNetwerk. The content was transmitted using a pair of MPEG-2 transmitter/receiver pairs at 300 Mbit/s sending rate.

SportN was put into service November 29th in 2005. The initial 45 days of operation, 52 packet were lost over the main link. Most of which were invisible to the viewer using error conceilment. The packet loss was acceptable, but generally contribution networks target at most 1 packet loss per day. On this particular link, error correction was later put in place to further reduce loss.

Occasional contribution

Occasional contribution denotes contribution from content creation sites. Such contribution happen sometimes only once, e.g. news coverage from a specific location, or once in a while, e.g. football played in the same stadium every weekend. Hence, *occasional* contribution.

The pool of contribution technologies used for occasional purposes is indeed heterogeneous. A brief description to the most common technologies in Norway is given next. Information here was given by [1]. Photos of the technologies can be found in appendix A.

- **IP contribution**: Large stadiums and arenas can be connected to an IP contribution network. For example, broadcasters have recently utilized IP for contribution from the Olympics (OL) - this was the case for Beijing OL and Vancouver OL [10][11]. As far as requirements and performance, occasional IP contribution networks are essentially the same as permanent contribution. One clear advantage with IP for occasional contribution is that the broadcasters demand to bandwidth can be met, because transmission technologies in IP links can provide excellent bandwidth (in the case of SportN, 300 Mbit/s IP contribution network was used). As a result, the content can be transported in very high quality avoiding quality degradation prior to the distribution phase. Remote production, i.e. transmitting of multiple camera angles, is also possible.

- **Satellite News Gathering (SNG)**: Here, content from the occasional site is uplinked to a satellite in the atmosphere, from where it is further downlinked to a broadcasting center. In the broadcasting world, it is common to use the term Satellite News Gathering (SNG) for occasional satellite contribution[5]. Satellite transmission equipment on the occasional site is typically mounted on a vehicle for fast deployment. Using SNG, a limited bandwidth is available (up to 30 Mbit/s). For content like news coverage where only one camera angle is sufficient, the content can be captured, compressed and sent on-the-fly. In this scenario, a simple van with a mounted satellite dish is sufficient (Satellite News Gathering (SNG) van).

  For more sophisticated productions, such as a sporting event or a talk show, the production must happen onsite due to bandwidth constraints. This is usually done in a production vehicle referred to as a Outside Broadcast (OB) truck. In a OB-truck, the content from multiple camera angles are edited to one signal and compressed onsite before relayed via satellite. In Norway, the main broadcasters TV2 and NRK have access to only a limited set of OB-trucks. The benefit with SNG is high geographical availability (coverage) and reliability.

- **Microwave transmission**: Here, the content is conveyed using a microwave transmitter. The signal must be aimed at a local receptor, e.g. a TV tower like Tyholt in Trondheim. Production vehicles like OB-busses may use microwave transmitters instead of satellite, but there exist also lighter and more handy equipment. For instance, NRK use a microwave transmitting system called STRATA which can be operated by one person and is very fast deployed (minutes). The maximum range for STRATA is 70 km, but this depends on the topology in the terrain as it needs a line of sight to the TV tower. Moreover, bandwidths up to 18 Mbit/s (depending on distance) may be conveyed, allowing contribution of high definition (HD) content. News and amateur sport coverage near cities (because TV towers often are located near cities) are common applications for STRATA.

- **Broadband Global Area Network (BGAN)**: A BGAN terminal is about the size of a laptop and can easily be carried and operated by one person. A satellite network, the Inmarsat I-4 network, provides almost global coverage for the BGAN terminals. This is a telephony and broadband network allowing bandwidths up to 0.5 Mbit/s. Therefore, this solution is most used for news coverage with low quality and high freshness demands. NRK often use BGAN for such purposes.

- **Mobile network contribution**: in Norway, 3G coverage for mobile terminals is quite good - especially near cities. This allows for contribution over 3G. A clear advantage is high availability and fast deployment, as a cell phone with an embedded camera may function both as a capturing and contribution transmitting device. An example here is Streambox Live, which is used by broadcasters such as NRK, CNN and FOX for some news coverage missions. Depending on the compression algorithm, this technology can deliver satisfying quality but it exclusively used for news coverage. With the introduction of 4G, such solutions is expected to be even more viable in the future.

[5] Satellite News Gathering (SNG) is not limited to contribution of news - contribution of other content is also done using SNG.

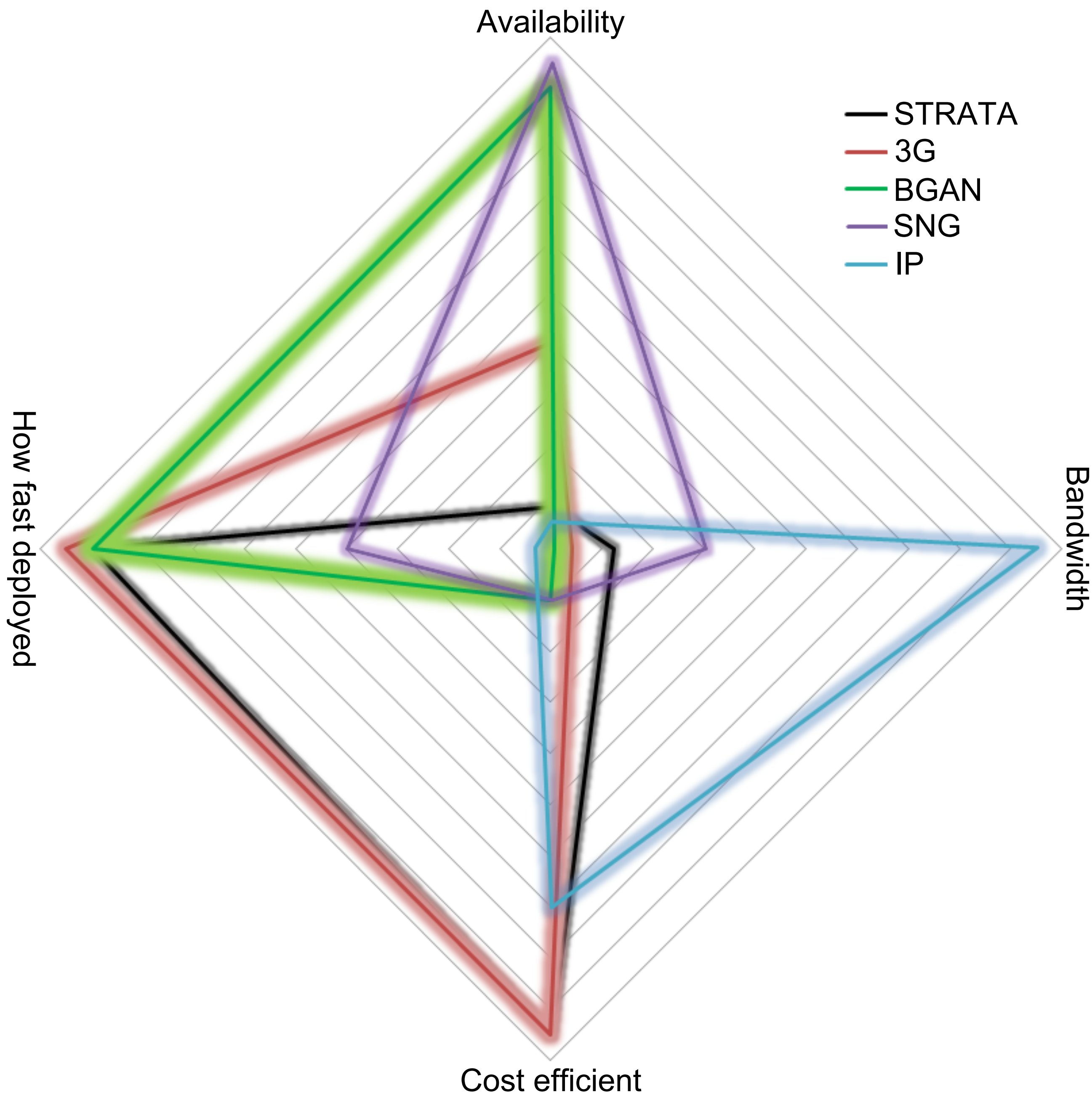


Figure 2.2: Illustration of different occasional contribution technology. Illustration based on information from [12].

Shortcomings in occasional contribution

Figure 2.2 sketch how the mentioned contribution technologies differs in terms of bandwidth, availability (geographical coverage), cost (the cost of setup and maintenance) and deployment time (or freshness, the time it takes to deploy a contribution service). As the figure suggests, there exist clear tradeoffs with every technology; there are no optimal solution.

The figure show that 3G and BGAN are exclusively suited for news due to lack of bandwidth. In news contribution, however, *the freshness exceeds bandwidth requirements* - the viewer tolerates degraded video quality, even loss in the video stream, as long as the breaking news coverage is viewable. Thus, deployment time and availability are the only requirements.

The end users expectations in terms of video quality and loss is very different for other contribution content such as sports and concerts. Here, STRATA (microwave), SNG and IP are three options. STRATA, however, has poor availability. Most often, the broadcaster must chose between IP and satellite.

As mentioned, contribution is sometimes done at temporary locations without access to dedicated IP contribution networks - *lack of availability*. IP contribution can essentially be done from everywhere, but the cost associated with implementing an IP contribution network access at the event site must be justified. As a general rule, 5-10 events must annually happen at a site to justify the implementation cost [7]. Once implemented, IP contribution networks are relatively cost efficient compared to SNG/OB.

The broadcaster most often utilize SNG and OB-trucks for production because SNG offers a high geographical coverage and a acceptable level of bandwidth. However, this solution is costly as it requires rental or purchase of the OB-bus, as well as personnel for transportation, setup and operation. The setup time is also significant as usage of OB-trucks are planned weeks in advance (usage of SNG vans, however, is very fast deployed).

2.2 The Internet as a contribution network

The Internet is the worlds largest data network, and it is based on IP. A clear advantage using the Internet as a contribution network is **availability**; in Norway 92% of all households and firms are connected [13]. Internet is in principle a non-profit network - users only pay for access - such that contribution over the internet has **low cost**. Also, IP contribution equipment (transmission system) is essentially "plug-and-play". Hence, an IP contribution system can be very **fast deployed** in the internet given an access line. There are question marks left at reliability and bandwidth; can the Internet deliver contribution services as reliant and with sufficient bandwidth compared to IP contribution networks?

If this thesis proves that reliability and bandwidth is sufficient in the Internet, it is easy to see that this solution *scores high on every scale* in figure 2.2. The Internet as a contribution network would therefore be an attractive contribution solution in some scenarios.

Chapter 3

Theory

This chapter will provide the necessary theory to appreciate and understand the main topics and principles in this report. It starts with presenting the theory behind digital video, and moves on to the basics about IP networks. This is the principles and technologies behind every IP network, including the Internet, which will be specifically presented and discussed in an own chapter, chapter 4. We then move on to IP contribution as an application of IP networks. We end the chapter with a presentation of the theory behind QoE based on the most recent advances in the topic.

3.1 Video

In this first part of the theory chapter, we go through some basic principle and practices in the topic of digital video. This will introduce some basic terms that are important throughout the thesis. We will also focus on the relationship between bitrate and video quality, and briefly discuss different ways to do subjective video quality assessment.

3.1.1 Frame rate, interlacing and resolution

A video is essentially a sequence of images, or *frames*, displayed consecutively at a certain rate, the *frame rate*. When the frame rate is higher than 15 frames per second (fps), the human brain will produce a sensation of continuity [14, p. 24]. Typical frame rates used are 24 fps for cinema, and 50 fps, much used for HDTV.

The technique *interlacing* can be used to double the perceived frame rate without increasing the bandwidth. In interlaced video, each frame is divided into two fields and updated in turn at a field rate twice the frame rate. This is a way to reduce flicker, which is normally perceivable at rates bellow 50 fps [14, p. 24]. Video that is not interlaced is called *progressive*, and is symbolized with a *p* for progressive video whereas *i* denotes interlaced video.

A video has a certain *resolution* specifying how many pixels each frame consists of. High definition TV (HDTV) is typically 1280x720 (720p/i) or 1920x1080 (1080p/i, sometimes referred to as "Full HD").

This also determines the *aspect ratio*, that is, the relationship between the number of horizontal pixels and vertical pixels. For HDTV the aspect ratio is 16:9. In broadcasting terminology, the video is normally just classified as HD, e.g 720p/i or 1080p/i, or SD, e.g. 480p/i or 576p/i.

3.1.2 Video compression

Uncompressed video contains a lot of redundancy which is a waste of bandwidth. Video compression aims at reducing the bandwidth to a level more practical for storage and transmission. A number of video compression schemes have been developed.

Video compression is either lossless or lossy. In lossless compression, the original pixel values is completely restored after decompression. Lossy compression will lose information hindering complete restoration, resulting in introduction of distortion artifacts in the video. This distortion may or may not be visible, depending on the compression factor and the efficiency of the compression.

In the broadcasting chain, the video content is very often re-compressed, called *transcoding*, prior to transmission over each link in the chain. Distortion introduced in an early phase, will propagate throughout the chain. It is therefore desirable to minimize coding distortion and artifacts in the contribution phase, before transcoding in the distribution phase.

The different video compression techniques can generally be categorized as intra coding or inter coding. Intra coding exploits spacial redundancy in each frame in the video signal, whereas inter coding exploits spatial and temporal redundancy between successive frames.

Intra coding

In *intra coding*, spatial redundancy, i.e. redundancy between adjacent pixels in one frame, is exploited to reduce the overall bitrate.

Each frame typically goes through three stages in intra coding; transformation, quantization and entropy coding. In *transformation*, a frame is transformed into another domain using a wavelet transform like Karhunen-Loeve transform (KLT), discrete cosine transform (DCT) or some other kind of transform. In the transformation domain, the pixels are represented by less correlated coefficients. As correlation essentially is redundant information, the transformation stage reduce the bandwidth of the signal without losing information [15]. Moreover, most of the energy in the frame is low frequent information and will be concentrated in a small subset of coefficients. Also, the eye is more sensitive to low frequencies than high frequencies [16].

Following the transformation stage is the quantization stage. The precision of the coefficients is reduced, but in such a way that the frame is minimally degraded. Quantization preserves more of the precision in the low frequency coefficients than the high frequency coefficients. The fact that quantization is a many-to-one process has two important consequences: First, we are able to compress the signal even more. Second, the process is generally irreversible.

In the entropy stage, even more redundancy is removed by exploiting that some representation levels from the quantization stage appear more often than others. For example, frequent levels can be coded with less bits than less frequent levels. This way, the overall bitrate is reduced. Common entropy coding techniques are Huffman coding and Arithmetic coding.

Inter coding

In addition to intra coding, most video codecs does some kind of inter-frame coding. Inter-frame coding exploits statistical dependencies between the frames, so called temporal redundancy, to reduce the overall bitrate needed to code the video. An inter coded video consists of three types of frames: I-frames that are just intra coded, P-frames that are forward predicted based on previous frames and B-frames that are bidirectionally predicted based on previous and future frames. This is illustrated in figure 3.1.

The inter-frame coding is highly efficient. The size of the P- and B-frames are about 50% and 25% the size of an I-frame [17].

The frames in an inter-frame coded video is organized in so-called Groups of Pictures. Each group starts with an I-frame, and the rest are P- and B-frames. For example can the pattern be ...[I BBB P BBB P BBB P BBB] [I BBB P BBB P BBB P BBB]... and so on. Because the P- and B-frames depend on the I-frame, en error in the I-frame may effect all the frames that are predicted from it. This will not an issue if the video is intra-only coded, since each frame will be independently coded.

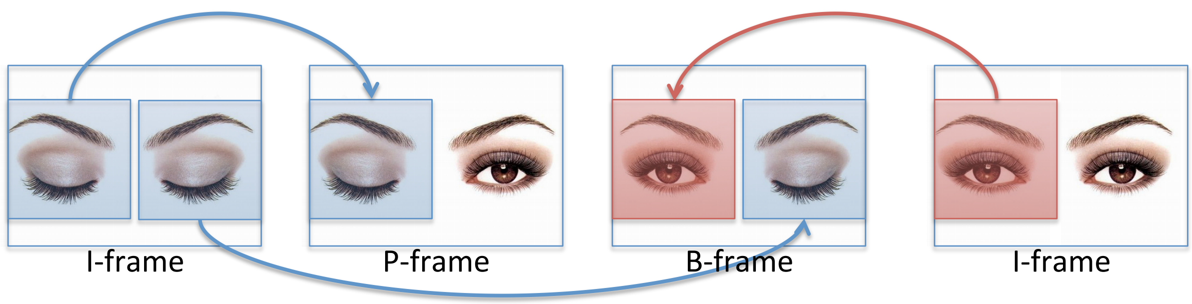


Figure 3.1: Prediction of P- and B-frames based on I-frames.
Illustrates how I-frames are used to predict P-frames and B-frames.

3.1.3 Chroma subsampling

Another way to reduce the bitrate is chroma subsampling. Each frame is normally represented by three matrices, one containing the luminance components, and two containing the chroma (color difference) components. The human visual system has low chromatic acuity [16, p. 208], that is, the human eye is more sensitive to luminance compared to chrominance. To exploit this visual redundancy, the chroma matrices are subsampled [16, p. 209]:

- 4:4:4 - no chroma subsampling.
- 4:2:2 - chroma subsampling by a factor of 2 horizontally.

- 4:2:0 - chroma subsampling by a factor of 2 horizontally and vertically.
- 4:1:1 - chroma subsampling by a factor of 4 horizontally.

In broadcasting, the video signal is usually first coded in 4:2:2 format in the contribution and production phase, before it is distributed in 4:2:0 [18].

3.1.4 Video codecs

A video codec, i.e.**cod**(er-)**dec**(oder), specifies compression and decompression of digital video[1]. A number of codecs have been developed, some of which are open standards and others are proprietary. A brief overview of the most important codecs for broadcasting is given here - namely MPEG-2, MPEG-4 AVC and JPEG2000.

MPEG-2 and MPEG-4 AVC

MPEG-2, approved by MPEG in 1994, is much used in broadcasting, DVD-Video and elsewhere. It was the codec of choice for broadcasters when broadcasters started digitalizing, both for contribution and distribution [18].

MPEG-4 AVC, also known as *H.264* was released as part 10 of the MPEG-4 standard in 2003. It was designed to meet the growing demands for better video quality at low bitrates in TV broadcasting, Internet streaming and teleconferencing, to name a few [19]. Many countries, Norway included, now use MPEG-4 AVC for their digital terrestrial television distribution.

MPEG-2 and MPEG-4 AVC are block based and use DCT transformation. This leads to a characteristic blockiness when the video is too harshly compressed. It also uses inter-frame coding to reduce the overall bitrate. So called Motion Vectors are computed by

finding the best matching blocks in the frames. The vectors describe how the matching blocks have moved, so that the information in the block does not have to be repeated.

MPEG-4 AVC uses much the same techniques as MPEG-2, and thus suffer from the same artifacts. But the techniques, e.g. the motion estimation, is more refined. Therefore, it generally preforms better than MPEG-2, specially at low bitrates.

JPEG2000

JPEG2000 is a still image compression standard developed to meet new demands in a range of different domains. Although JPEG200 share a similar name to its predecessor JPEG - it is based on quite different principles.

[1]It is common practice to only define decoding part in the codec specifications, allowing different encoding implementations.

The transformation in JPEG2000 is based on wavelets. This is essentially a process where the whole frame is filtered both vertically and horizontally. The outcome of this process is low frequent and high frequent information in both vertical and horizontal direction. This filtering process is often repeated two or more times producing sequentially ordered layers - the lower layers define low frequent, "base" information, while the higher layers define high frequent "enhancement" information.

JPEG2000 has a video extension called *Motion JPEG2000*. Motion JPEG2000 is a intra coding video compression scheme as each frame is coded using JPEG2000 and no prediction is done between the frames. Therefore, Motion JPEG2000 require a higher bitrate compared to inter coding schemes like the MPEG family to produce similar picture quality.

The advantage with Motion JPEG2000 is the low compression latency (latency due to inter prediction is avoided), superior image quality on high bitrates [20] and scalability in terms of quality and spatial resolution. As JPEG2000 compression is tile-based, as opposed to block based in the MPEG family, blockiness artifacts will not pollute the video. The most common artifact is blurring introduced when too much of the high frequent information is removed.

Motion JPEG2000 is bandwidth consuming, and is most used in professional domains such as Digital Cinema (DC) and contribution. In 2005, the Digital Cinema Initiative, a joint venture of Disney, Fox, Paramount, Sony Pictures Entertainment, Universal and Warner Bros. Studios, chose Motion JPEG2000 as the new compression standard for DC [21]. As DC has high demands on video quality and virtually no bandwidth requirements, Motion JPEG2000 was a viable option. The resolution in DC is 2K (2048 x 1536 pixels) or 4K (4096 x 3072 pixels), and its maximum total bitrate is 250 Mbit/s [22].

Motion JPEG2000 has also become a great success in contribution as the standard is the codec of choice for many large broadcasters around the world [6]. The most important benefits of Motion JPEG2000 in contribution is the low coding latency and the minimal quality degradation avoiding transcoding-distortion in the broadcasting chain. Common Motion JPEG2000 bitrates for contribution applications are 100-300 Mbit/s for HD and 35 Mbit/s for SD [7].

We will refer to "Motion JPEG2000" as just "JPEG2000" in the rest of this thesis.

3.1.5 Bitrates

It is useful to have a feel for the relationship between different bitrates and qualities. Table 3.1 gives an overview of common distribution phase technologies and the bitrates they operate on. In comparison, the maximum bitrate for MPEG-2 compressed video on DVD is 9.8 Mbit/s [23] and the bitrate of a BluRay-Disc is 36 Mbit/s.

Table 3.1 show technologies that are typical in the distribution phase of the broadcasting chain. Contribution will have different (generally higher) requirements to the video. These requirements will be discussed in section 3.3.1.

	Video Resolution	Uncompressed bitrate	Compressed bitrate
NTSC[2] video	480x480 @ 29.97fps	168 Mbit/s	4 to 8 Mbit/s
PAL[3] video	576x576 @ 25fps	199 Mbit/s	4 to 9 MBit/s
HDTV	1280x720 @ 60fps	1327 Mbit/s	18 to 30 Mbit/s
HDTV	1920x1080 @ 30fps	1493 Mbit/s	18 to 30 Mbit/s

Table 3.1: An overview of bitrates of compressed and uncompressed video for common TV technologies. Based on table in [24, p. 6].

3.1.6 Influencing factors for video quality

Besides compression and properties like resolution, frame rate and interlacing, a number of things might influence the quality of the video. *Additive noise* may be introduced in the process of capturing the video, as may *blurring and loss of detail*. *Color bleeding* is a form of blurring that happens in the color channels. The colors will appear to flow into each other. This is to name a few.

When video is transported over a network, packet loss may introduce errors in the video. The effect of a packet loss depends on the way packet loss is handled by the receiver, for example: A black frame is inserted for the missing frame, the previous frame is repeated or a frame is estimated based on past and future frames. Also, this is often done on a basis of the part of the frame that is effected, for example a block or a slice, and not necessarily the whole frame. This type of handling of errors is called *error concealment*. As mentioned in section 3.1.2, errors in a frame used for prediction might propagate to the P- and B-frames. Thus, the exact effect of packet loss on the overall video quality will vary from codec to codec. Since JPEG2000 does not use inter-frame coding, the effect of an error will probably be less severe than for inter coded codecs.

3.1.7 Transport of uncompressed video

The video signal also needs to be transported from the camera into the video compression device, e.g. video gateway, and from the decoder to a display. Because the video will be uncompressed, and thus have high bitrate, standards that supports high bandwidths have been developed. Most infamous is perhaps the HDMI standard. The bandwidth of the latest version of the HDMI standard is 10.2 GBit/s, making it suitable even for 4K video. In the broadcasting industry, the Serial digital interface (SDI), and advances of the SDI standard, HD-SDI and 3G-SDI, are much used. They are backward compatible, so that video outputted on an SDI or HD-SDI interface may be inputted on a 3G-SDI interface.

	Max bitrate	Example Video Formats
SDI	360 Mbit/s	480i and 576i
HD-SDI	1485 GBit/s	720p and 1080i
3G-SDI	2970 GBit/s	1080p

Table 3.2: Some of the different SDI standards, their bandwidth and possible applications. Based on information from [25].

3.1.8 Assessment of video quality

To assess video quality, it is common to use subjective quality assessment methods. A number of participants are in some way asked for their opinion on the quality of a video clip. In *double stimulus* techniques, a reference clip is used. The reference can be explicit, and the participants are asked to rate the clip in relation to the reference, or unknown (hidden), in which case they rate both clips on a scale or by preference. Double stimulus techniques are typically used for short videoclips, e.g. 10 seconds. For longer video clips, up to 30 minutes, *single stimulus*, or *simultaneous double stimulus* techniques with continuos rating of the video are used [26].

From the results provided in this thesis it is evident that none of the subjective assessment techniques mentioned above are suitable for our problem: In a real scenario, unprovoked visible packet loss will appear relatively seldom, about once an hour in our tests, so we need something more long-term than 30 minutes. Also, the video quality is likely to be perfect in long periods of time, and bad in the very short periods when packet loss occurs. We therefore looked for a more aggregated measure than video quality, and found Quality of Experience, described in section 3.4, to be suitable.

3.2 IP networks

In the context of data communications, a network is an interconnection of devices or hosts. In order to communicate, each host must use the same communication protocol and have an unique address.

A data network is hierarchically constructed in layers, according to the OSI model. A great introduction to the OSI-model is given by [27]. At layer-2, hosts communicate through one or more switches using hardware addresses (MAC). A Internet Protocol (IP) network implements layer-3. As illustrated in figure 3.2, a layer-3 IP network interconnects layer-2 networks forming a network of networks. Hence, internet.

In an IP network, devices have a unique IP address and communicate based on the rules specified by the IP protocol. In addition to hosts (end users), IP routers directs flow of information through collection of networks.

This section will present the IP protocol, IP Quality of Service (QoS), Real-time video over IP and transport protocols used in IP networks.

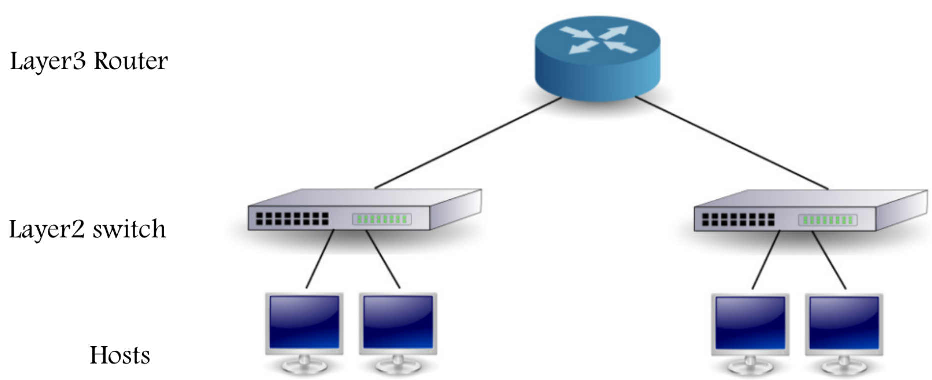


Figure 3.2: An IP network implements OSI layer-3 and interconnects layer-2 subnets.

3.2.1 IP - The internet protocol

The Internet Protocol (IP) is a standard for directing and routing flow of information between devices in an IP network. The basic unit of information is one *packet* constructed by appending an IP header (figure 3.3) to a variable length data payload. A router in the IP network accepts packets from a network node (a host or another router), and forwards this packet to another node based on the *Destination address* field in the IP packet header. Upon reception at the destination node, the *Source address* field specifies the sender. An IP packet is much like an envelope; it only holds an address and is able to carry any type of content.

An IP network was designed with some clear goals. In order to serve a scalable internet, it was designed to be *flexible*. Flexible both in terms of interconnecting networks with unequal design and performance and in terms of carrying all kinds of services - some of which used dedicated networks

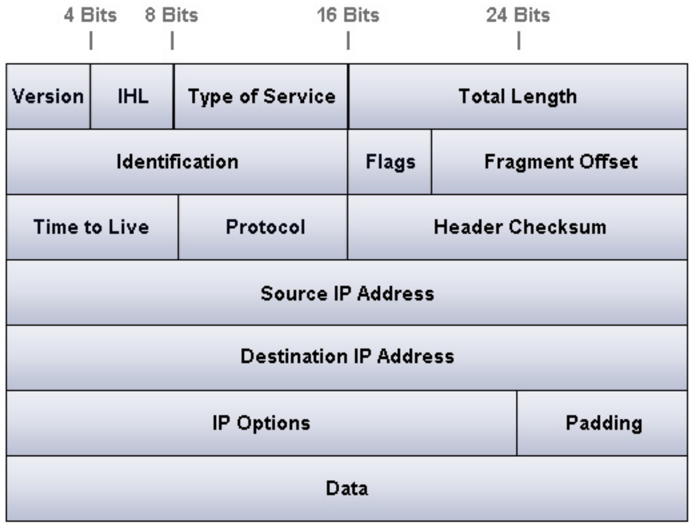


Figure 3.3: The Internet Protocol (IP) header.

a decade ago (like phone, cable television, etc). This flexibility is ensured by the packet switching, connectionless design of IP.

Complex logic is placed at the network edges (i.e. hosts) in an IP network, while the network is connectionless and works in best-effort. Best-effort means that the network is unreliable in terms of guaranteeing delivery or latency. Once a packet is sent from router A to router B, B does not feedback packet arrival confirmation to A. Packets lost between A and B are therefore never retransmitted by A (at the IP layer, that is).

The lack of logic in the network may result in *network congestion*. Congestion is essentially traffic overload in the network causing variable delay and in the worst case dropping (deletion) of packets in network routers. Congestion avoidance techniques are not implemented at the IP layer - it must be implemented at the transport layer (layer-4).

The simple and abstract design of IP allows application developers and network architects to extend the IP feature set with additional features in the transport layer (layer-4) and the application layer (layer-5). For example, retransmission, if applicable, is implemented in the transport layer (layer-4).

IP is an open standard inviting equipment vendors to competition, keeping IP equipment cheap. IP-networks can be very reliant and fast if designed and maintained properly. For example, QoS tools and real-time support can be implemented for reliability and performance.

3.2.2 IP Quality of Service (QoS)

The Quality of Service (QoS) term is used in various domains such as telecommunications, IP network operation, multimedia and so forth. As a result, the term is used quite differently depending on the context. To avoid confusion, this thesis will use the QoS term strictly in conformance to ITU-T recommendations.

ITU E.800 defines QoS as "The collective effect of service performance which determine the degree of satisfaction of a user of the service". The end-to-end QoS term is an collective evaluation of network performance, terminal equipment performance and user perception.

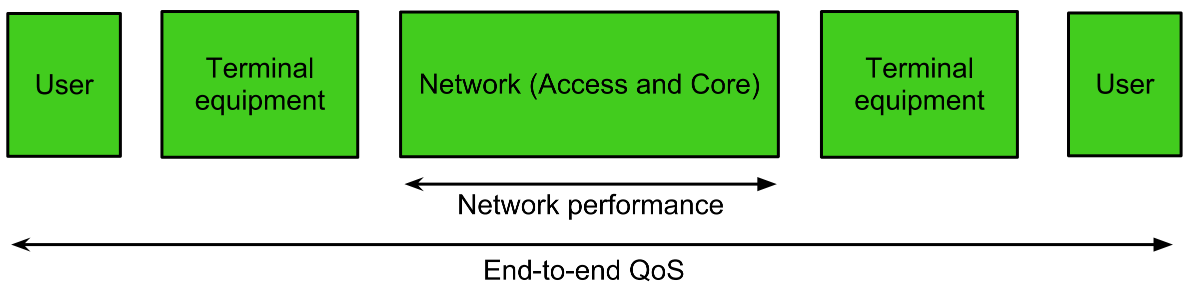


Figure 3.4: The end-end QoS term as defined in ITU E.800.

It is reasonable to assume that the terminal equipment used in IP contribution does not introduce overall performance degradation, and can therefore be neglected in the QoS term in this thesis. Moreover, user perception is also neglected in context of QoS - this thesis will use the QoE term to describe user perception. *The QoS term is therefore used for network performance in this thesis.*

Quality, in this context, is defined in ITU E.800 as "the totality of characteristics of an entity that bear on its ability to satisfy stated and implied needs". The QoS term numerically evaluate the ability of the network to satisfy the specified needs - stated in QoS metrics (like packet loss, jitter and delay) defined in the section 3.2.2.

Recall that a simple IP network works in best-effort. In other words, the network may take on unbounded QoS metric values for an end-end service. This is due to the uncontrolled, unmanaged nature of an IP network where the hosts in the network compete for resources on the same premises. The QoS terms allows customers to state *QoS requirements*, providers to state *QoS offered/guaranteed* and both customers and providers to state *QoS delivered/achieved* - expressed in descriptive terms using QoS metrics. Thus, the QoS term allows customers and providers to bound QoS metrics in the best-effort IP network.

This section will present the QoS metrics used in this report, and how QoS can be guaranteed in an IP network.

QoS metrics

The QoS metrics are objective measures on service performance. As the QoS term is equal network performance in this thesis, the QoS metrics used here are essentially measures on network performance. This thesis will follow ITU recommendation G.1050 where five QoS metrics for network performance are proposed;

1. **One way delay (ms)** is the time it takes for a packet to traverse the network. This parameter is dependent on the average load on the network (i.e. queuing delay), number of router hops as well as physical propagation and transmission delay.

2. **Jitter (IPDV) (ms)** is the variation in packet delivery delay. This is caused by variable delay in router queues, and also when packet takes a more time consuming path. IP packet Delay Variation (IPDV) gives the absolute transmission delay variation as the difference between the upper and lower packet delivery delay:

$$IPDV_{total} = IPDV_{upper} - IPDV_{lower} \tag{3.1}$$

   where $IPDV_{upper}$ and $IPDV_{lower}$ are maximum and minimum packet delay, respectively, in a 99.9% percentile set.

3. **Packet loss (%)** is the percentage of packet lost in the network, i.e. packets that never arrive. In ITU G-1050, this metric is called "Random packet loss".

4. **Packet loss burst (ms)** is called sequential packet loss in ITU G-1050. This metric is given in ms, not the number of packets in the packet loss burst, such that the metric applies regardless of packet sending rate.

5. **Packet loss burst rate (PLBR) (1/s)** is the rate of packet loss burst occurrence. PLBR is called "Rate of Sequential Packet loss" in ITU G.1050.

$$PLBR = \frac{packet\ loss\ burst\ count}{time(s)} \tag{3.2}$$

Note that the QoS metrics are renamed to, hopefully, more intuitive names here compared to G.1050. In the context of network performance, It is sensible to define two additional network parameters here;

- **Sending rate (data/second)**: is the rate of outgoing data per time from the sender.

$$Sending\ rate = \frac{bits\ sent(Mega)}{time(s)}(Mbit/s)$$

- **Throughput (data/second)**: is the rate of incoming data per time at the receiver.

$$Throughput = \frac{bits\ received(Mega)}{time(s)}(Mbit/s)$$

IP QoS tools

This section presents tools which network designers can implement in order to guarantee a level of QoS in accordance to QoS requirements. There are in general three such tools [5];

- **Network provisioning** where the network is scaled according to the user activity to ensure enough bandwidth to services. QoS requirements is therefore met, or attempted met, by ensuring enough resources to the service.
- **Service prioritization** where each host or type of process is classified into different priority levels. A packet flow is then handled with priority in the network routers according to the class it belongs to. Thus, the highest prioritized IP packet in the incoming router buffer will typically be forwarded first. This is typically done by setting the Type of Service (TOS) field in the IPv4 header, or equivalently Traffic Class in the IPv6 header. According to the router policy the higher prioritized packets will be handled with favor.
- **Resource allocation** where resources are allocated in the network routers to a flow of packets from a specific service. The network routers along the path reserve buffers and resources to the packet flow creating a tunnel between the sender and the receiver. Examples here are V-LAN configuration and Multiprotocol Label Switching (MPLS). MPLS is a very popular scheme. Here, a packet is labelled to inform the routers about the flow it belongs to. A network router forwards the packet based on this label as opposed to long network addresses which demands more processing delay. MPLS is actually a compromise between connection-oriented networks like ATM and connectionless networks like IP; the establishment of a MPLS tunnel is a connection-oriented approach in an IP network, but IP is able to reroute the traffic in case of tunnel breakdown.

3.2.3 Real-time video over IP

In computer science terms, 'real-time' refers to the concept of delivering data with timing constraints. That is, data must arrive at the receiver before a *deadline* or, equivalently, within a *delay budget*. Real-time delivery of video allows end users to watch the content while it is created at the content creation site. This is essential for content like news programs and sporting events because the freshness of the content is very valuable to end users.

Push versus pull delivery
It is important to clarify the difference between *pull-based* and *push-based* real-time delivery of video.

Internet media streaming is generally pull-based. The receiver *pulls*, i.e. requests, content from the server in the desired quality/sending rate. The overall aim is a smooth video service; the video receiver (PC, TV, SmartPhone, etc.) pulls video quality according to receiver characteristics (screen, processing power, etc.) and network throughput. Smoothness is ensured by relaxing the delay requirements allowing the receiver to buffer seconds of content to avoid frame freezing in periods with low throughput. Also, retransmission of lost media packets is possible due to low delay requirements. Examples of pull-based services are Youtube, TV2Sumo, NFL Game Pass and so forth. We did a simple test with the "NFL Game Pass" service delivered by *www.nfl.com* - it showed that the receiver (our PC) buffered around one minute of content. In effect, the receiver could playback content even if the network throughput is zero for up to one minute.

In *push-based* real-time video delivery, the video content is *pushed* from the server to the listening clients. Compared to pull-based, push-based delivery gives more immediate video viewing experience because the delay is much lower. This requires, however, a reliable network and that virtually no packet loss occur. Retransmission is generally not possible due to low delay requirements - hence, lost media packets directly degrades the end user viewing experience. Therefore, push-based video delivery is most used in reliable IP networks like IPTV networks.

IP contribution is a push-based real-time delivery system. This section will therefore introduce principles around push-based video delivery. It is important to emphasize that pull-based Internet video applications falls outside this discussion - and the remainder of this thesis.

Push-based real-time video system

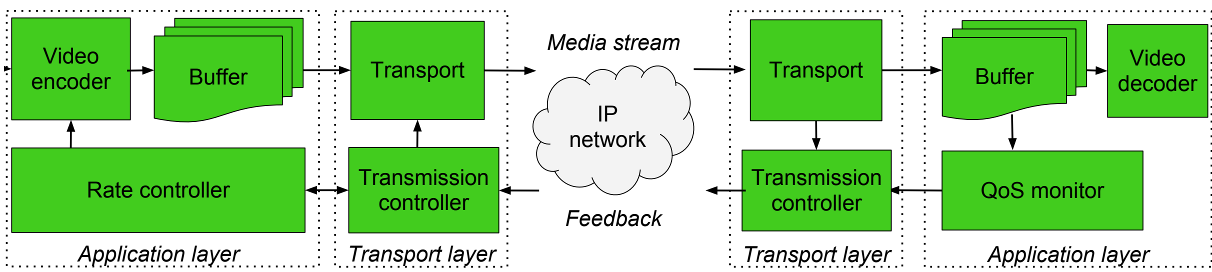


Figure 3.5: Example of a setup for a real-time video system.

Figure 3.5 illustrates a push-based system for real-time delivery of video content. The overall idea is to have a media stream from the server to the client, and a feedback stream in the opposite direction. Feedback information contains report on experienced network and media quality from the client. Based on feedback, the server can adjust the video bitrate and sending rate on-the-fly such that reliability and performance is achieved. This is known as *rate control*.

The figure is based on information from [28]. We do not claim, however, that this system is a general framework for push-based video delivery. It is our proposal to how a contribution service can be delivered with rate-control in best-effort networks. The remainder of this section will introduce every module in this system. Later in this thesis, the importance and design criteria for each module is analyzed in order to achieve a reliable contribution service in best-effort networks.

I. IP network

The IP network in the figure is a best-effort network without QoS guarantees. The video service may therefore experience unbound QoS metric values.

II. Video encoder/decoder and Rate controller

Input to the video encoder module is typically uncompressed video, and the video decoder outputs the received video to a screen. This figure is not limited to a specific codec, but it is taken for granted that the video codec can alternate the video encoding bitrate on-the-fly. The rate controller contains decision logic for alternating the encoding bitrate based on feedback from the transmission controller.

III. Buffers

Both the client and server have a buffer where content is saved temporary. At the very least to cope with jitter smoothing out the playback rate at the client. However, the larger buffers, i.e. a larger delay budget, the more resilient system. This is because the receiver is less vulnerable to network throughput fluctuations. Given that retransmission is implemented, the receiver may tolerate packet loss periods of length equal or less than the buffer size. Also, the sender may transfer at a higher sending rate than playback rate for a time equal or less than the sender buffer, refilling the receiver buffer which is possible because *there are packets to send in the server buffer*.

The buffer is also a network throughput indicator. If the buffers are filled faster than the playback rate, then this is a clear indication that the network can deliver the video with higher bitrate/sending rate.

IV. Transport and transmission control

This is the transport layer (layer-4 in OSI-model) - implementing a simple transport module, where the packets is sent, and the more complex transmission control, where the packets are monitored.

The transmission controller can track lost packets and calculate packet delay. This requires two transport features, namely *ordering* and *timing*. Ordering is essentially implementation of packet sequence numbers allowing the receiving transmission controller to both track lost packets and also order incoming packets correctly. Timing is implementation of timestamps allowing synchronization and delay calculation as the difference between the sending and receiving timestamp.

The two most critical features in the transmission controller, however, are *congestion control* and *error control*. These features are presented next.

IVa) Congestion control serves the purpose of varying the sending rate such network congestion does not cause packet loss and the network avoids congestion collapse. Congestion collapse happens when the network is completely overloaded - in this case, network resources are wasted on transporting packets that never will arrive anyway. Thus, *the server tries to adapt the sending rate to avoid packet loss caused by congestion*. Network degradations, such as packet loss and jitter, are used as network congestion indications which make the sender adjust the sending rate.

The main design choice for a congestion control is first of all if congestion control is required; if the best-effort network is never congested, then congestion control is of less importance. Secondly, how smoothly the sending rate should be adapted to network congestion. Obviously, adapting the sending implies adaption of video compression. The congestion control module must therefore vary the rate hard enough to avoid congestion collapse, yet smooth enough to avoid video codec failure and considerable drop in subjective quality.

IVb) Error control handles packet loss. Packet loss are unavoidable in best-effort network and introduce significant subjective quality degradation. Two basic error control strategies are 1) forward error correction (FEC) and 2) retransmissions of lost packets [28].

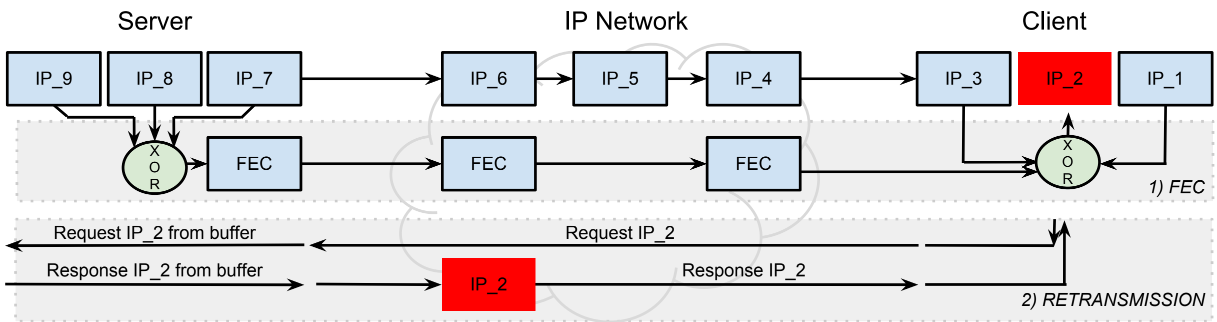


Figure 3.6: Illustration of error control strategies - FEC and retransmission. The blue and red boxes illustrates IP packets.

The principle of FEC is to add a parallel error correcting stream capable of reconstructing lost packets at the receiver. A simple FEC-XOR scheme may produce the correcting stream (see figure 3.6). See appendix E for a detailed introduction of FEC. In general, FEC is suitable for correcting random packet loss and minor packet loss bursts. For example, a (20,5) FEC stream can correct packet bursts up to 20 packets (approximately 5 ms at 70 Mbit/s). FEC introduce bandwidth overhead (in the range of 5-20% SJEKK DETTE) and delay roughly proportional to the maximum correctable packet loss burst. FEC is most suited for lightly congested networks such as professional IP contribution networks.

Retransmission of lost packets is a straightforward strategy; the sender retransmits lost packets either if no ACK (packet arrival acknowledgement) is received (TCP-like retransmission), or if the receiver explicitly requests retransmission of a lost packet (UDP-like retransmission) via the transmission controller module. Retransmission is the most effective error control strategy because only useful information is sent. However, extensive delay is introduced - at least the time the receiver waits for the lost packet, plus one RTT (request+resend). If the retransmission request is lost, the receiver must request again and thus even longer delay is introduced. In a dedicated IP contribution network the retransmission delay is unacceptably high. For contribution of non-interactive content[4] over the Internet, retransmission cannot be ruled out if it provides superior reliability.

3.2.4 IP network transport protocols

This section will introduce the two most common transport protocols in IP networks, namely Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). Furthermore, this section introduce how transport protocols are customized with the desired features. The focus is to introduce real transport protocols that can implement the transport layer in the video delivery system illustrated by figure 3.5.

[4]Interactive content such as an interview require very low delay.

TCP

The Transmission Control Protocol (TCP) is a connection-oriented, stream-based transport protocol between two hosts. It is by far the most popular protocol in IP networks (in terms of data transmission). TCP implements all basic transport protocol features. This section, however, will only review the important strategies for this thesis, namely TCP error correction and TCP congestion control.

Error control in TCP is done by retransmission. In TCP terms, this is known as *reliable transfer*. Reliable transfer implies that delivery is guaranteed; the receiver acknowledge (ACK) reception of each packet such that missing segments are detected and retransmitted. TCP will never give up on retransmission of a packet, implying that TCP is not concerned with delivery delay.

With TCP congestion control, the sender tries to adapt the sending rate, the *congestion window*, according to the level of packet loss. In other words, packet loss is used as indicators on network congestion.

The congestion window varies throughout the connection. In the start-up phase called *slow-start*, the rate is exponentially increased. The next phase is called *congestion avoidance*. Here the sending rate is additively increased for every successful packet reception. In the case of packet loss, the sending rate is divided by 2 - and then the slow-start phase[5] is re-initiated. These principles are illustrated in figure 3.7.

As a result, the number of bytes in-flight between the hosts never exceeds the sending window and the congestion window. The primary goal for TCP is to allow scalability while avoiding network congestion collapse.

UDP

The User Datagram Protocol (UDP) is a very simple protocol as it only provides features for directing data flow between hosts - along with checksum for consistency check. Implicitly, UDP is a stateless protocol without handshaking and negotiation, reliability, ordering (sequence numbers), flow control and congestion control. As a result, UDP datagrams may arrive out of order, appear duplicated or lost without notice and retransmission. The UDP sending rate is constant - it can only be varied in the application layer at the sender. Using UDP is essentially close to bypassing the transport layer because only the required features are provided.

Customizing transport protocols

Essentially all other transport protocols are either an *extension of UDP* or a *modification of TCP*. Modifications generally happens in the transport layer, while extensions happens in the application layer. For simplicity here, *a transport protocol refers to the collections of strategies that defines transport behavior*. Figure 3.8 illustrates the point that a lot of transport protocols exists, and they

[5] This is the simple explanation. In practice, the receiver will send duplicate ACKs - essentially asking the sender to retransmit the lost packet (fast retranmisit, TCP Reno). If no retransmission happens before time-out, then the rate is set to half.

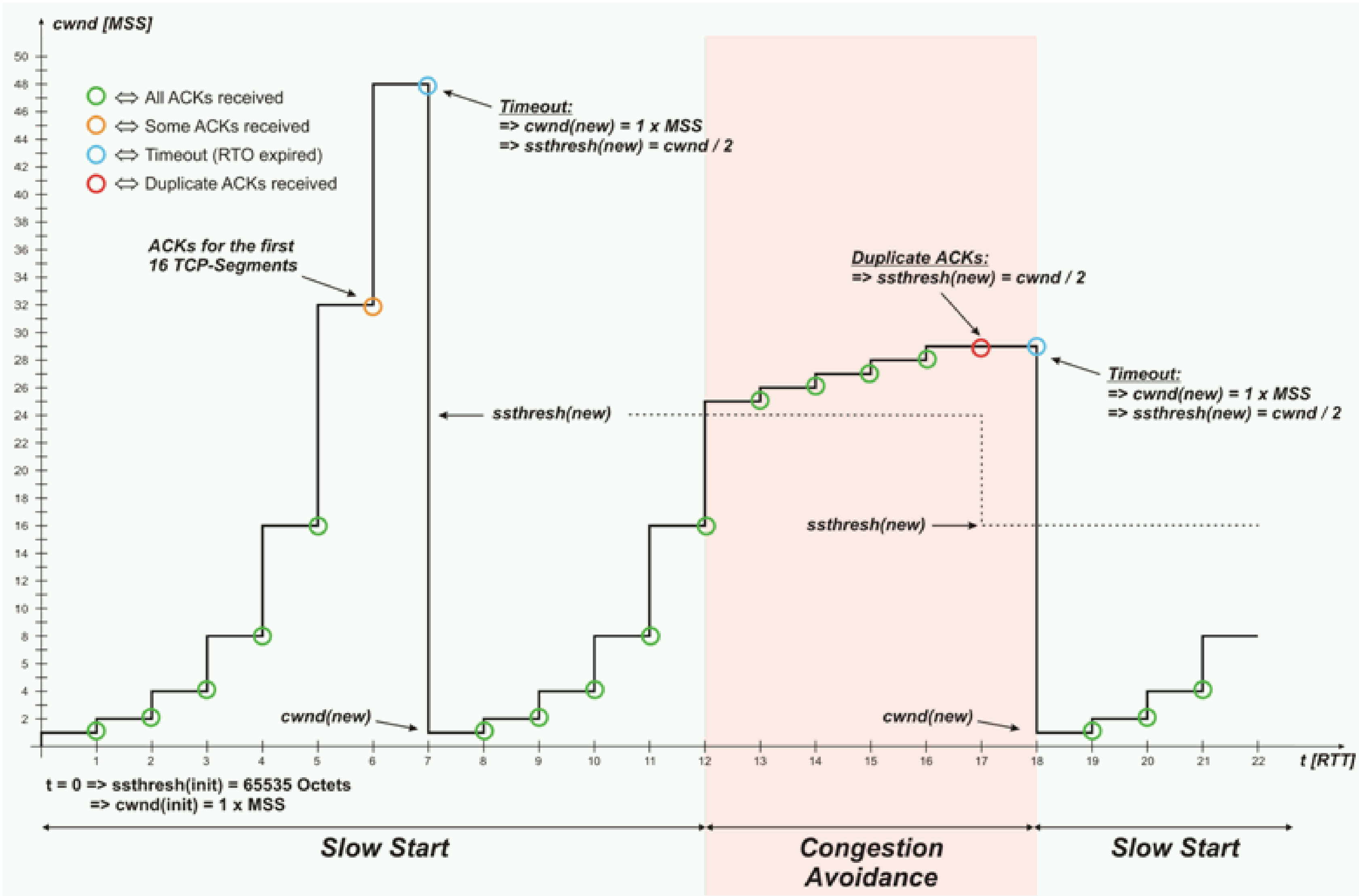


Figure 3.7: TCP slow start and congestion avoidance (illustration by http://www.inacon.de/).

differ in the implemented transport features.

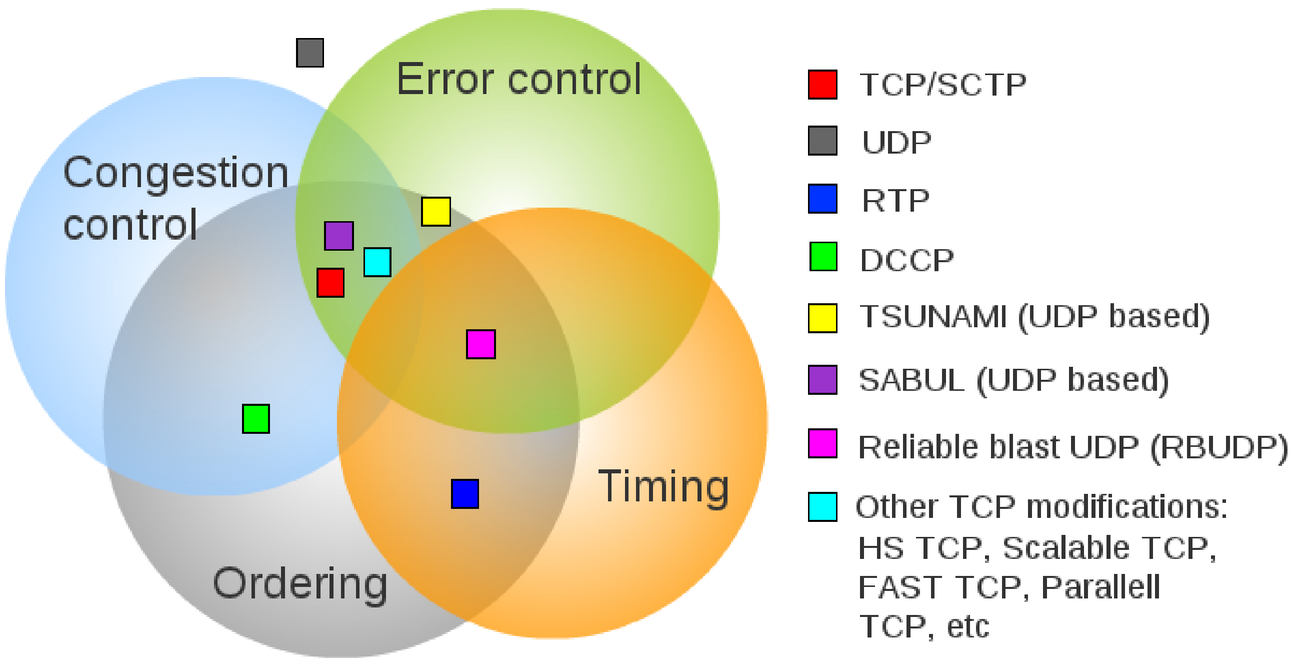


Figure 3.8: Illustration showing features for different transport protocols.

In general, it is a question of *what transport protocol features and tweaks are required* as opposed to *finding a stand-alone protocol that approximately meets the system requirements*. As mentioned; congestion control, error control, ordering and timing are the four most important transport features for a contribution system in best-effort networks. A aim in this thesis is to find out whether TCP or UCP should be used in the transport layer in figure 3.5, and which transport features should be modified or extended.

3.3 IP contribution

IP contribution networks are designed and monitored to conform to a strict contribution requirement level, guaranteeing a level of QoS to the IP contribution packet flow. An IP contribution network is far beyond a best effort network as the uncertainties associated with IP are controlled.

How QoS is guaranteed in contribution network, is very context dependent. QoS can be guaranteed solely with proper network provisioning in a dedicated IP network [7]. That is, by implementing enough bandwidth, QoS can be guaranteed because the contribution operator firm has complete control of all the traffic in the network and can therefore assure that no queueing will happen.

In some cases, explicit QoS tools like MPLS and V-LAN configuration is required *on top of proper network provisioning*. This allows the QoS marked traffic to have priority in network routers. MPLS is a very powerful strategy for large core networks with QoS offerings.

Another important tool in order to guarantee QoS, is traffic engineering. Traffic engineering is manual monitoring and routing of traffic. The main goal is to ensure distribution of traffic in the network, while providing the redundancy required. A motivation for contribution over IP is the ease of traffic engineering due to the flexible nature; the traffic may easily be rerouted in case of error or queueing. Because the IP contribution firm has complete control of the IP network, all media streams are surveilled by at a centralized location. It is also possible to send test streams (probes) in the network to detect errors before the broadcasting starts at a location.

For a discussion on advantages and disadvantages with IP contribution, as well as case studies, see chapter 2. This chapter will focus on contribution requirements.

In general, broadcasters, contribution network providers and customers have not agreed upon a standard set of contribution requirements [7]. This is because some of these standards is sometimes unknown to customers and broadcasters. In addition, the requirements to contribution networks are sometimes very easy to state; no error should occur.

Nevertheless, this thesis will use a proposed set of professional contribution requirements from Digital Video Broadcasting (DVB) and ITU-T. T-Vips, a Norwegian contribution equipment vendor, use these requirements and advised us to do the same. One can argue that professional contribution requirements are too strict for some occasional contribution setups in the Internet. However, it is the authors belief that using professional contribution requirements in this report defines a proper framework. Later, it can be discussed whether the requirements put on occasional contribution should be more relaxed.

3.3.1 IP contribution requirements

For contribution bandwidth requirements, the document "High level contribution codec commercial requirements by DVB will be used [29]. For contribution network requirements, ITU-R G.1050 will be used [3]. This section will present contribution requirements from these documents.

Contribution bandwidth classes

Digital Video Broadcasting Project (DVB) is an organization of broadcasters, manufactures, network operators and others committed to design open technical standards for delivery of digital television.

[29] defines 4 bandwidth classes for contribution shown in figure 3.9:

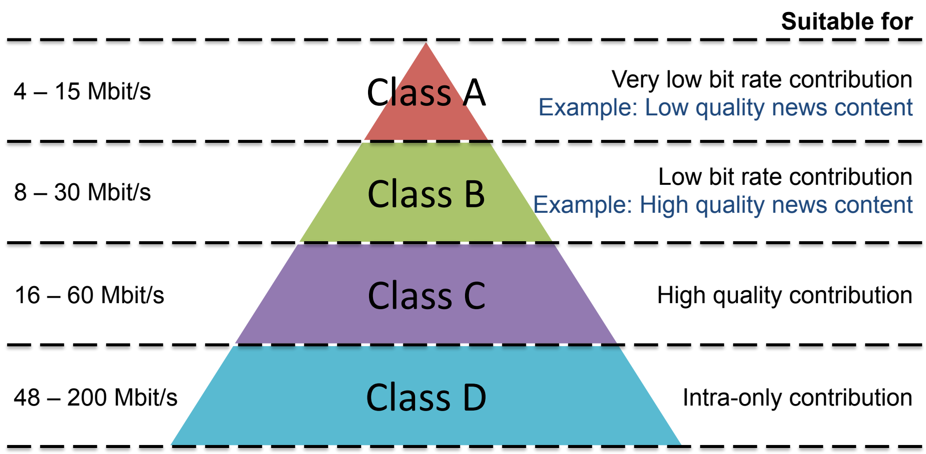


Figure 3.9: The contribution bandwidth classes defined by DVB.

Bandwidth class A,B and C is meant for inter-coding such as MPEG-2 or MPEG-4 AVC, while class D is for intra-coding using JPEG2000 or MPEG-4 AVC in intra-only mode.

Image format	**Class A**	**Class B**	**Class C**	**Class D**
1080p/60fps	N/A	30	60	200
1080p/30fps	10	20	40	120
1080p/24fps	10	20	40	120
1080i/30fps	15	30	60	150
720p/60fps	15	30	60	150
576i	4	8	16	48
480i	4	8	16	48

Table 3.3: Minimum Contribution Video bitrate in Mbit/s (DVB CM-AVC [29]).

It is important emphasize that table 3.3 states the *minimum decoding requirements* for the different classes. For example, for 720p/60fps for class D (intra) - the decoder should at least be able to decode up to 150 Mbit/s. Implicitly, bitrates below 150 Mbit/s is also decodable. The table therefore indicates the supported bitrates for each class and image format.

Contribution network profiles

ITU-R G.1050 defines impairment range for transmission of audio/video streams over IP networks [3]. It defines three network profiles and associated QoS based on the level of management and bandwidth.

- **Profile A:** Well managed network with no over-committed links which employs QoS in routers as well as traffic engineering.
- **Profile B** Partially managed network that minimizes over-committed links and has one or more links without QoS guarantees and/or traffic engineering.
- **Profile C:** Unmanaged network such as the Internet that includes over-committed links and one or more links are without QoS guarantees and traffic engineering

Impairment type	**Unit**	**Profile A**	**Profile B**	**Profile C**
One-way latency (delay)	*ms*	20 - 100	20 - 100	20 - 500
IPDV	*ms*	0 - 50	0 - 150	0 - 500
Packet loss	*%*	0 - 0.05	0 - 2	0 - 20
Maximum Packet loss burst	*ms*	n/a	40 - 200	40 - 10000
Packet loss burst rate (PLBR)	*1/s*	n/a	0.001	0.1
Reordering	*%*	0 - 0.001	0 - 0.01	0 - 0.1

Table 3.4: Network profiles (ITU-R G.1050 [3]). Note that profile A does not tolerate packet loss bursts (hence n/a). PLBR states the maximum average occurrence of packet burst loss.

It is important to note that the network profiles in G.1050 describes network performance in general. As far as we are aware, there exist no standard defining network requirements for a contribution network - especially not for contribution over less reliable network such as the Internet.

However, the Video Service Forum (VSF), which is an organization for broadcaster, network operators and manufacturers proposed network profile A for live video transport [4]. Profile A is therefore the requirements to a professional contribution network in this thesis.

Network profile C is aimed at best-effort networks, like the Internet. The QoS limits defined in network profile C is therefore what can be expected using the Internet as a contribution network.

This thesis will use the network profiles in G.1050 as a scale on which the Internet can be pinpointed as illustrated in figure 3.10. Later, it can be discussed whether this framework give meaningful results.

In the remainder of this thesis, the network profiles in G.1050 is referred to as "contribution network profile requirements".

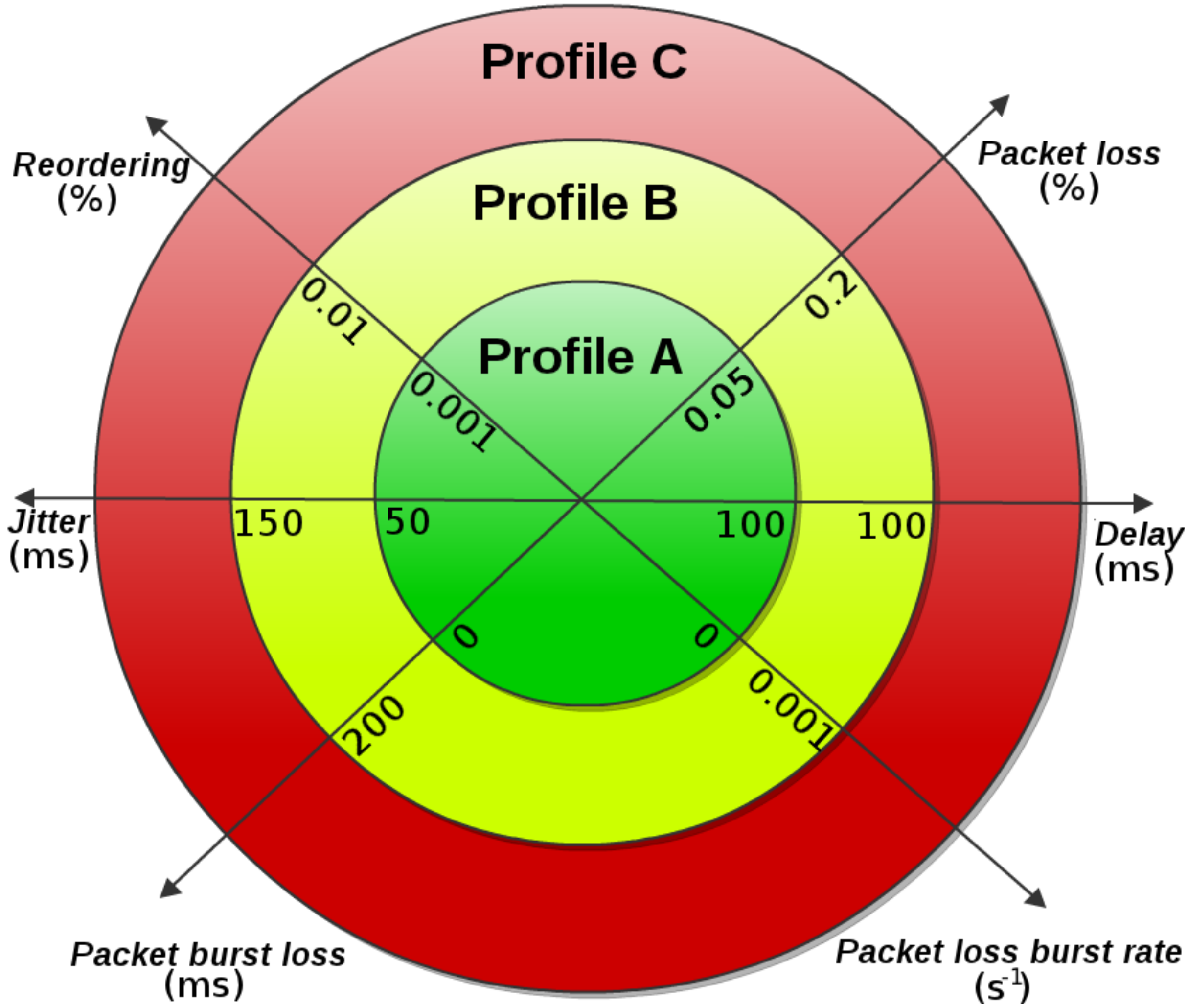


Figure 3.10: Illustration of network profile requirements in ITU-R G.1050.

3.3.2 Contribution transport protocol format

The SMPTE 2022 family defines unidirectional transport of video in IP networks, and is often followed by IP contribution equipment vendors [8]. This section will present a bird's-eye view of the standards.

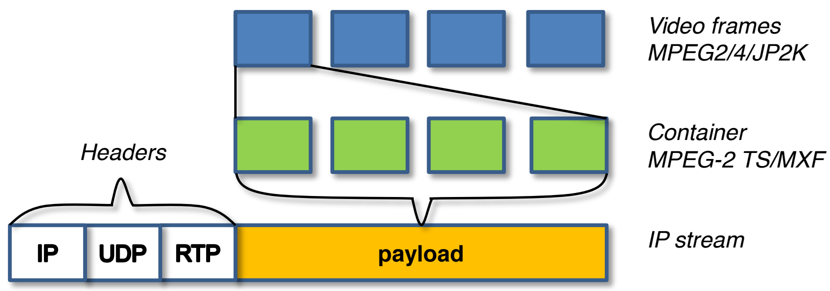


Figure 3.11: Contribution transport format.

Video frames, coded with MPEG-2, MPEG-4 AVC or JPEG2000, are packed into *container* packets. The MPEG-2 TS container format is used for MPEG-2 and MPEG-4 AVC (SMPTE-ST 2022-1, 2022-2) and a MXF container is used for JPEG2000 (SMPTE RP-2008). Generally, a container format multiplex video frames and audio - along with headers for synchronization and ancillary data. This allows perfect reconstruction of the incoming media fractions at the receiver.

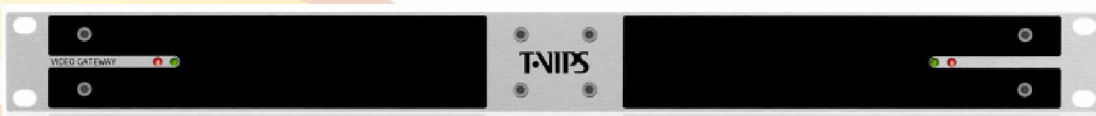
Regardless of codec, i.e. MPEG-2, MPEG-4 AVC or JPEG2000, the SMPTE standard propose packing of media payload in a RTP/UDP/IP transport core.

A set of container packets are pasted as payload in an RTP packet. As mentioned, RTP provides sequence numbers, allowing ordering and tracking of lost packets, and timestamps for delay calculation. Then, the RTP packet is inserted into an UDP segment and finally the UDP segment is inserted into an IP packet. The IP packet is never above 1500 bytes (MTU) avoiding fragmentation in IP routers.

Errors are generally not acceptable, therefore support for some sort of FEC strategy is recommended (SMPTE ST 2022-1), but there are applications where occasional errors are preferable to the overhead of the FEC, so manufacturers may support a non FEC mode. The FEC stream is implemented as a parallel IP stream to the media IP stream.

One noticeable aspect here, compared to a push-based real-time video system described in 3.2.3, is the lack of logic placed in the transport and application layer in the contribution real-time system. This is because the contribution network, as opposed to a best-effort Internet with varying performance, have QoS tools implemented in the network routers. Hence, rate control and congestion control is not applicable because the contribution network is guaranteeing throughput and reliability. The video encoding settings, e.g. bitrate and image format, are *not varied throughout the contribution session*. Therefore, TCP is not used due its fluctuating sending rate and congestion control. A simple

Case-study: T-VIPS 450 JPEG2000 Video Gateway



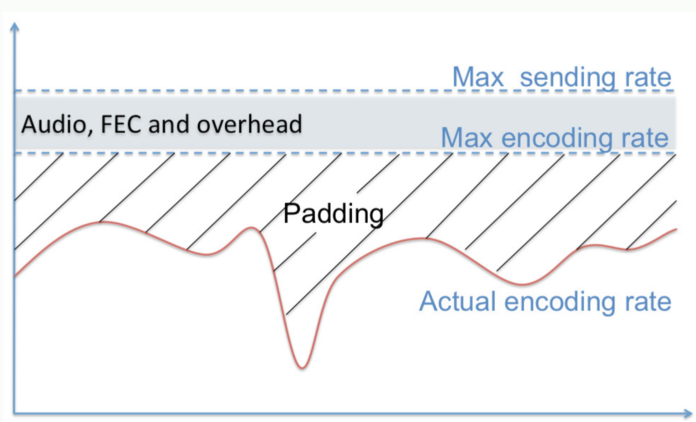
TVG450 is a JPEG2000 video gateway produced by the Norwegian company T-Vips. This gateway can function as both sending and receiving gateway for professional broadcast contribution over IP networks. The gateways accepts SDI/3G-SDI video stream as input - for example a video stream from an on-site camera. 4 SD video streams, 2 HD streams or 1 HD and 2 SD streams can be output simultaneously. TVG450 can produce IP streams at a rate between 25 Mbit/s to 450 Mbit/s.

Once connected to an IP network, the gateway need no further setup - it is simply "plug and play". The gateway settings is easily operated on-the-fly via a web interface (HTTP).

The video encoding bitrate can either be in constant or variable mode. In constant mode, padding is used on top of the encoding bitrate resulting in constant sending rate. In variable mode, the sending rate will follow the fluctuations in the encoding rate. The encoding rate will vary on a frame-basis according to JPEG2000 encoding efficiency; frames with low level of information is coded much more efficiently than frames with more information. As a result, the sending rate ranges from close to 0 Mbit/s up to the maximum sending rate. This is illustrated in the figure below.

The transport core in TVG450 conforms to the SMPTE family. The gateway multiplex video, audio and auxiliary data into one MXF/RTP/UDP/IP stream. Optionally, a parallell FEC (SMPTE 2022-1) IP stream can be toggled. FEC is able to correct a limited number of sequentially lost IP packets. This transport core is introduced in section 3.4.2.

This particular gateway is in use by many large TV operators, for instance; mediaNetwerk, SKY Italy, A1 Telekom Austria, RTL, FOX Sports Australia, etc.



3.4 Quality of Experience

The term Quality of Experience (QoE) has been developed to meet the need for a broader way to describe a communication service than the network-centric Quality of Service (QoS) [2]. A number of definitions have been proposed in attempts to capture the full extent of the concept, and this is still work in progress. We have chosen to stick to the most up to date proposal, found in [2]:

> **Definition:** *Quality of Experience (QoE)* is the degree of delight or annoyance of the user of an application or service. It results from the fulfillment of his or her expectations with respect to the utility and / or enjoyment of the application or service in the light of the user's personality and current state. In the context of communication services, QoE is influenced by service, content, network, device, application, and context of use.

[2] also points to three main groups of factors that influence the QoE (IFs): Human IFs, System IFs and Context IFs.

Human IFs are factors related to the users, like their background and physical, mental and emotional state.

The *System IFs* are factors related to the technical part of the system. These are further grouped into four groups:

- Content-related: Content type and content reliability.
- Media-related: Configuration factors like encoding and frame rate.
- Network related: QoS network parameters like bandwidth, loss and throughput.
- Device related: Specifications of the system devices, like screen resolution of a display or computational power of a decoder.

Context IFs are factors related to physical, temporal, social, economic, task and technical characteristics. For example location and space (physical), time of day and duration (temporal), number of people present (social), cost and subscription type (economical), single- or multitasking (task) and relations to other systems (technical).

In this thesis we are interested in the network related System IFs, and how these influence the overall QoE. More precisely, we are interested in the link from the network related System IFs, through video quality, to QoE. This is illustrated in figure 3.13.

Unlike QoS, QoE lacks standardized measures and ways to do measurement. Where QoS has clearly defined terms like PLBR and IPDV, QoE has no such definitions for now. A common way to measure the *degree of delight or annoyance* in fields of social studies like psychology is through survey. We have also seen examples of surveys being used to measure QoE in other studies [26]. Therefore, we choose to develop our own survey based on the principles of QoE when we measure QoE in chapter 6.

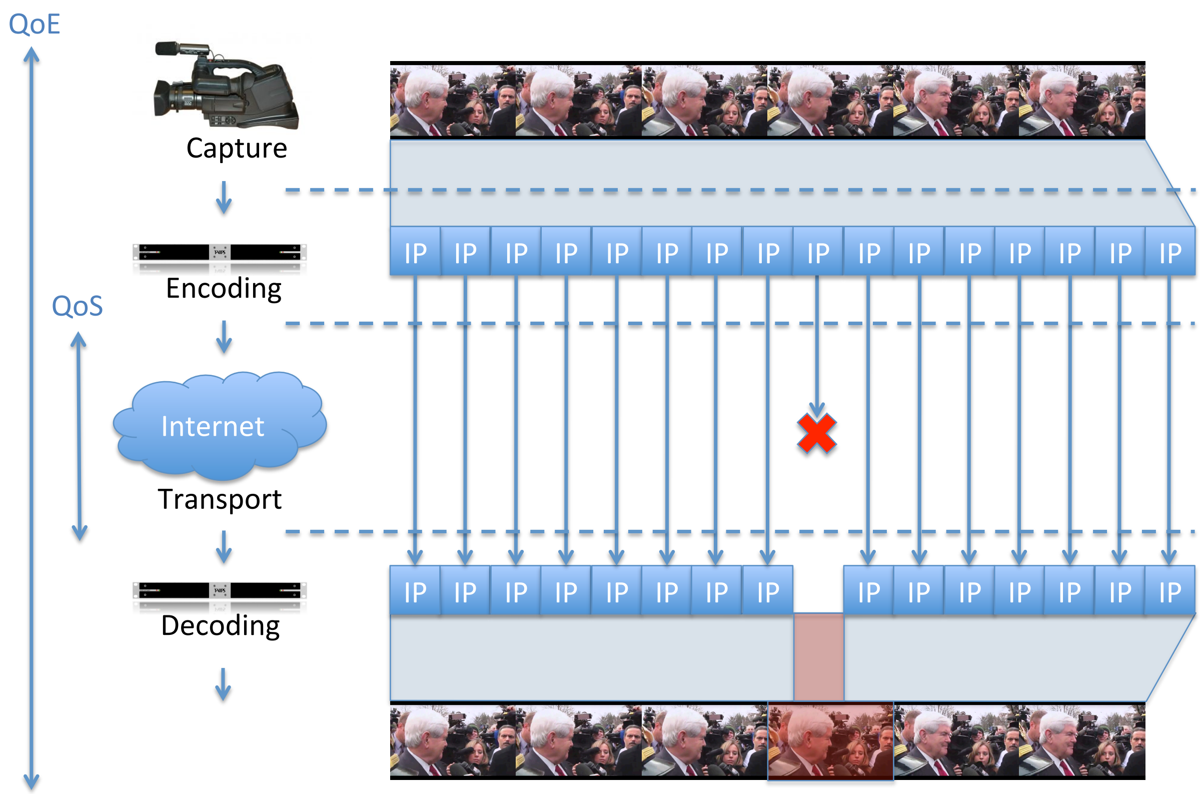


Figure 3.12: A complete video capture and transport system with a camera capturing the video, video gateways compressing, sending, receiving and decompressing the video. It also shows that a packet loss will effect parts of the video. The relation between QoS and QoE is indicated on the left, based on figure 3 in [30].

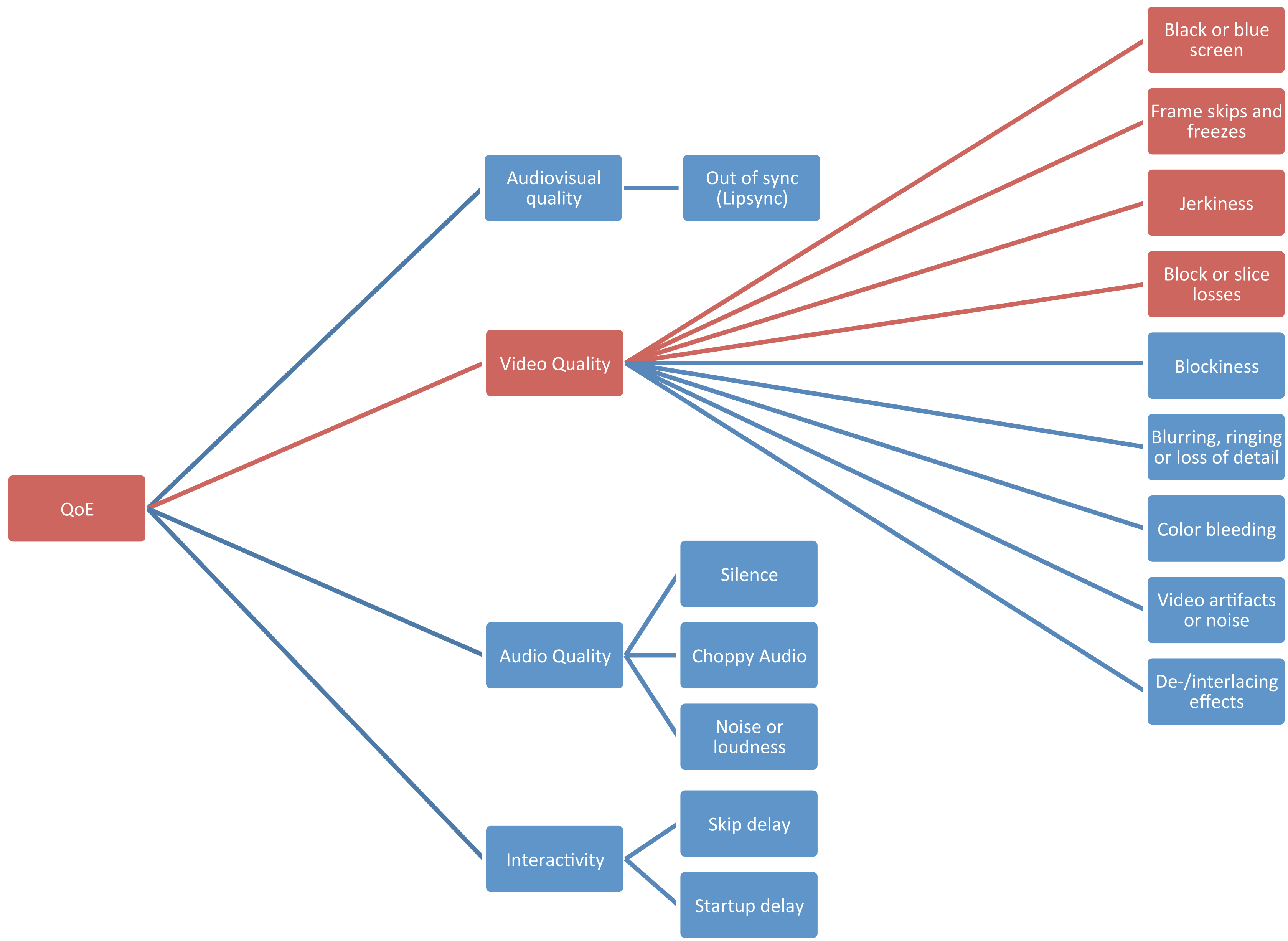


Figure 3.13: Different scenarios where System IFs might effect the overall QoE. The network related, our area of interest, are shown in red. Based on [26, p. 13 - 18].

Chapter 4

The Internet - the worlds largest IP network

The purpose of this chapter is to answer the first subquestion from the introduction, namely, *What is the state of Internet performance in terms of bandwidth and reliability?*

The Internet is, by far, the worlds largest IP network with over 2 billion users worldwide [31]. Over the last 20 years, the Internet has evolved from a research project to an inevitable communication platform for all.

A clear goal in this thesis is to avoid treating the Internet as a *black box* - that is, a network with unknown interior characteristics and response. By investigating the main design and dimension trends in the commercial internet, we can first of all give a clearer answer to whether the Internet is contribution ready. Secondly, we can understand and explain results from contribution testing. To keep this study fresh and recent, we have intentionally used references newer than 2009.

This section is organized as follows. First, Internet topology and terminology is introduced in section 4.1. Next follows a discussion on the key to the Internets success in section 4.2. Then, and most importantly, follows a discussion on recent trends with respect to Internet performance and QoS in section 4.3. Finally, we end the chapter with a discussion on how the Internet can serve as a contribution network.

4.1 Internet topology

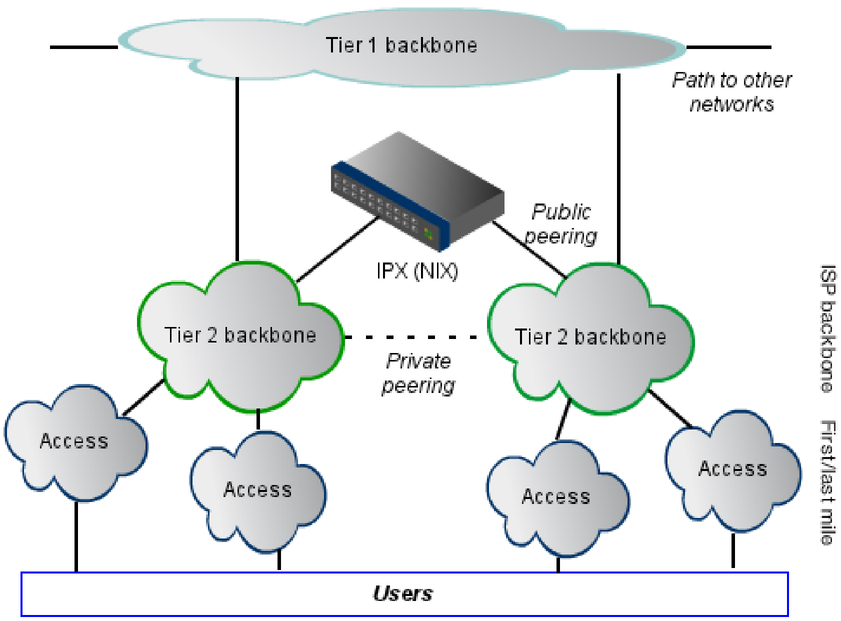


Figure 4.1: Illustration of Internet topology.

Figure 4.1 illustrates basic Internet topology. The Internet consist of 34,000 different IP networks (Autonomous Systems) which indeed are heterogenous in terms of size, capacity and redundancy[1][31]. Below follows a brief introduction to the different classes of IP networks that together forms the Internet.

End-users are connected to the Internet in access networks. These networks are operated by Internet Service Providers (ISP), except for some large firms and institutions operating their own access network. The links from end-users terminals to access network are called *first-mile* or *last-mile* - depending on whether traffic is uploaded (first-mile) or downloaded (last-mile). The most popular access lines are copper lines (xDSL) and fiber optical lines. Furthermore, access networks keeps a state for every end-users, ensuring that the purchased amount of bandwidth is provided. Today, as access networks and first/last-mile bandwidth is rapidly on the rise, access networks are not necessarily the bottleneck[2] between a pair of hosts [31]. This was the case a decade ago when slow access technologies like ISDN and ADSL dominated.

Tier-2 networks are often national ISPs - like Telenor or Ventelo. Usually, the spine of a Tier-2 network is an advanced high-capacity core network [32]. Access networks, owned by the Tier-2 firm or customers thereof, are connected to this core network. Traffic between end-users with the same Tier-2 ISP may therefore flow internally in the Tier-2 network. Tier-2 networks are often very redundant and have high capacity. A flurry of different services are offered end-users in Tier-2 networks - like IPTV, on-demand video, cloud services and music services. Internally in Tier-2 core

[1] Redundant networks continue to function in the presence of single points of failure.
[2] A bottleneck is the limiting component, like a link or router, in the path between a pair of end-users.

networks, services can be delivered with QoS guarantees to customers.

Externally, Tier-2 ISPs reach other ISPs and their end-users through either *peering* or by *transit*. A *peering point* is a network node where two or more ISPs exchange traffic. Tier-2 ISPs can setup private peering points for traffic exchange (illustrated by dotted lines in figure 4.1). Alternatively, regional and national ISPs can reach each other through *public peering points*. In Norway, there exists 6 such public peering points[3], known as NIX (Norwegian Internet Exchange) [33]. Through these points, all the major ISPs in Norway (approximately 60 ISPs) exchange traffic. In general, no QoS is guaranteed over public peering points.

A Tier-2 network can only reach a limited set of Tier-2 networks through peering. For complete, world-wide reach - Tier-2 networks must buy transit from Tier-1 networks. Tier-1 networks are large, high-performance networks directly connected to the Internet backbone, and in some cases can be considered a part of the Internet backbone. Examples of Tier-1 networks are Sprint, TeliaSonera and Level3. As a part of the Internet backbone, Tier-1 networks can reach all other hosts in the Internet. Hence, when a Tier-2 network buys transit from a Tier-1 network - this essentially means that the Tier-1 network is responsible for world-wide delivery of the Tier-2 traffic. In some cases, Tier-2 networks can buy QoS guarantees from a Tier-1 network - for example for content distribution to distant hosts.

4.2 The key to Internet success

Based on the above presentation of Internet topology, it is evident that the internet is a highly distributed and hierarchical model. No organization *owns* the Internet or defines omnipresent rules and policies. The question is then, how could the internet get so successful in terms of availability and performance despite the lack of control?

The answer to this lies first of all in that design principles behind IP, as introduced in section 3.2.1, still holds. The key word is *flexibility* - both in terms of services and networks. Indeed, the Internet has proven to serve different networks well, and also more and more sophisticated services have been implemented.

IP networks were designed to have logic placed at the edges as opposed to in the network core. From this criteria, TCP was formed with the main objective to allow scalability while avoiding congestion in networks - *without explicit policies across networks*. Today, roughly 90% of the data in the Internet is sent using TCP as transport protocol. We did a simple statistic study at caida.org, and found that 94.88% of all traffic though a peering point (equinix) in San Jose (USA) over a month was sent using TCP [34]. Figure 4.2 illustrate this.

Although the Internet initially was a research project, Internet survival and expansion is purely due to development of successful business models - ensuring that money flow into the networks from the first and last mile users. Generally, the more successful business models and services implemented in the ISP network - the more money flow - which allows ISPs expand their network with capacity and redundancy to implement even more services. Therefore, Internet evolution will take the most economically profitable path. The next section will present state of Internet trends.

[3] Tromsø(TIX), Trondheim (TRD-IX), Bergen (BIX), Stavanger (SIX) and two points in Oslo (NIX1,NIX2).

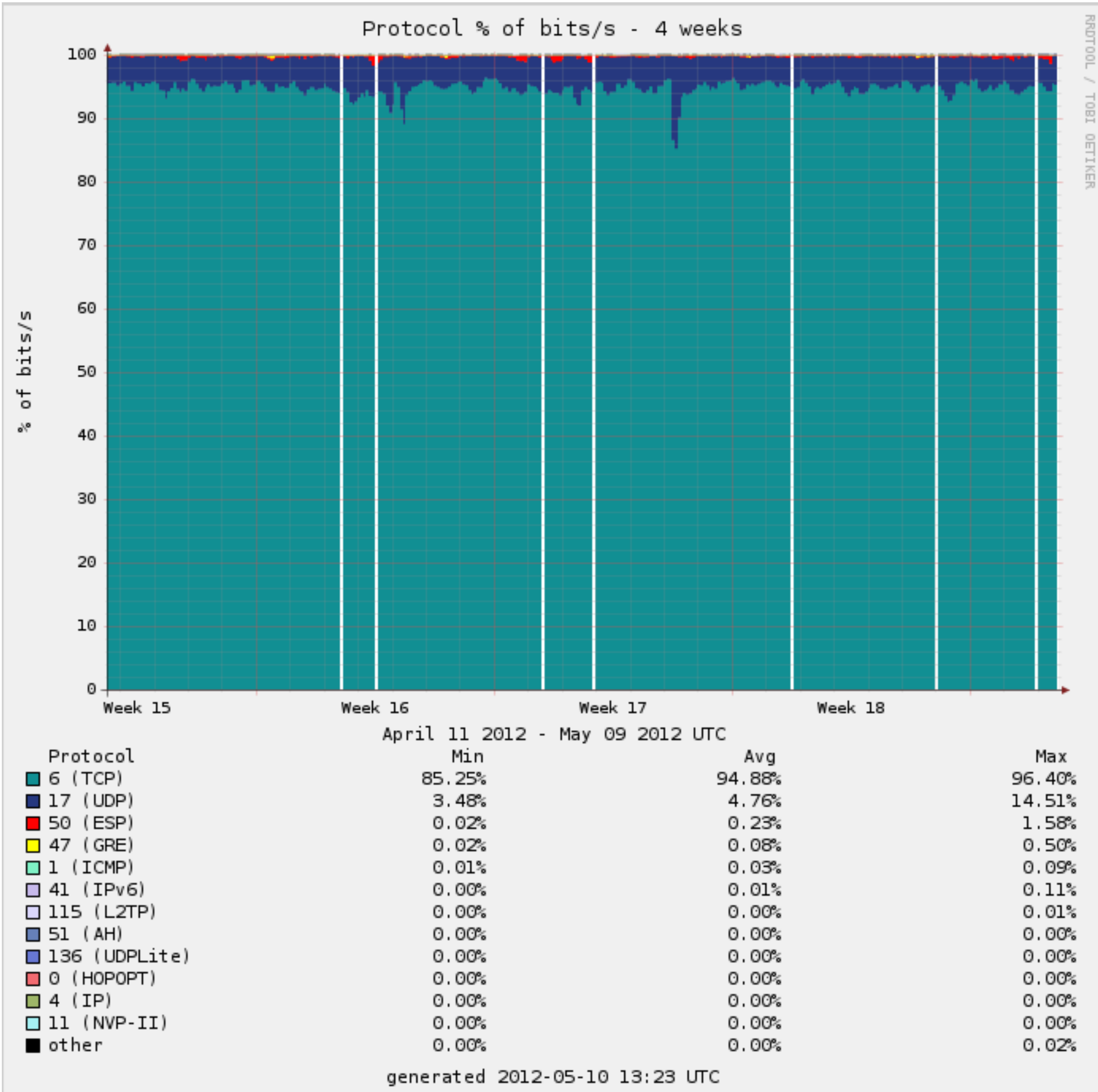


Figure 4.2: Graph showing usage of Internet protocols in terms of data (%). The data was recorded in a public peering point in San Jose (USA) called *equinix* between April 11 and May 09, 2012. The graph was generated by the authors on May 10, 2012. This service is provided by http://www.caida.org/data/realtime/ .

4.3 Internet trends - bandwidth, performance and QoS

This section will present Internet trends with respect to performance and reliability. At the end of the section, it should be more clear what can be expected of the Internet and if todays Internet is contribution-ready in terms of reliability and bandwidth.

Some of the main trends are [35][32][36][31][37];

1. In the recent years a traffic explosion has been witnessed, mostly due to delivery of Internet video. This, in turn, has led to a dramatic increase in first-mile bandwidth and reliability.
2. The best-effort principle still holds in the Internet. Congestion is limited by reducing router hops and peering points between hosts. Adaptive video streaming has proven to collaborate well with best-effort networks.
3. Tier-2 networks are very complex and well-dimensioned offering many services to customers.
4. There exist business models in Tier-2 networks allowing users to buy QoS guarantees.

1. Traffic and bandwidth explosion

Global IP traffic has increased eightfold over the past 5 years, and is expected to increase fourfold over the next 5 years according to Cisco [36]. Internet video (PC and TV) traffic has increased most dramatically over the last years, and now accounts for approximately 50% of Internet traffic. Figure 4.3 illustrates both total Internet traffic growth and the rapid growth of Internet video traffic. Video-on-demand (VoD) services is one of the big reasons for this massive growth. By 2015, Cisco anticipates that Internet video accounts for more than 60% of internet traffic [36].

Implicitly, as bandwidth demanding video services is emerging in the Internet, end-users Internet access lines bandwidth (last-mile) must keep pace. This has resulted in a vast growth in high-capacity access lines like vDSL and fiber lines (FTTx). In Norway, Telenor introduced vDSL - "very fast broadband" - in 2011 with capacity up to 70Mbit/s downlink. Also, the share of FTTx lines is on the rise. Fiber lines is generally superior to copper (xDSL) due to reliability and capacity [38]. In 2011, 17% of Norwegian households had FTTH connection compared to 9% in 2009 - an increase of 88% over just two years according to SSB [13][39].

As a result, the average first/last-mile capacity in Norway between 2004 and 2011 annually grew with 40% according to "Post- og Teletilsynet" [40]. In this time span the average downlink capacity increased from 1.5 Mbit/s to 12 Mbit/s. By 2015, "Post- og Teletilsynet" expects that 70% of Norwegian households have 40 Mbit/s average capacity [40]. Given the challenging topology in Norway (mountains, fjords and areas with low population density) the average capacity can be somewhat misleading. In cities and areas with high population density the average capacity is higher. For example in Oslo, 100 Mbit/s capacity is offered by ISPs at a reasonable price [41].

In summary, the Internet is evolving from elastic content exchange, i.e. delivery of content without timing deadlines, to delivery of bandwidth consuming Internet video services. ISPs are therefore offering access links with more bandwidth.

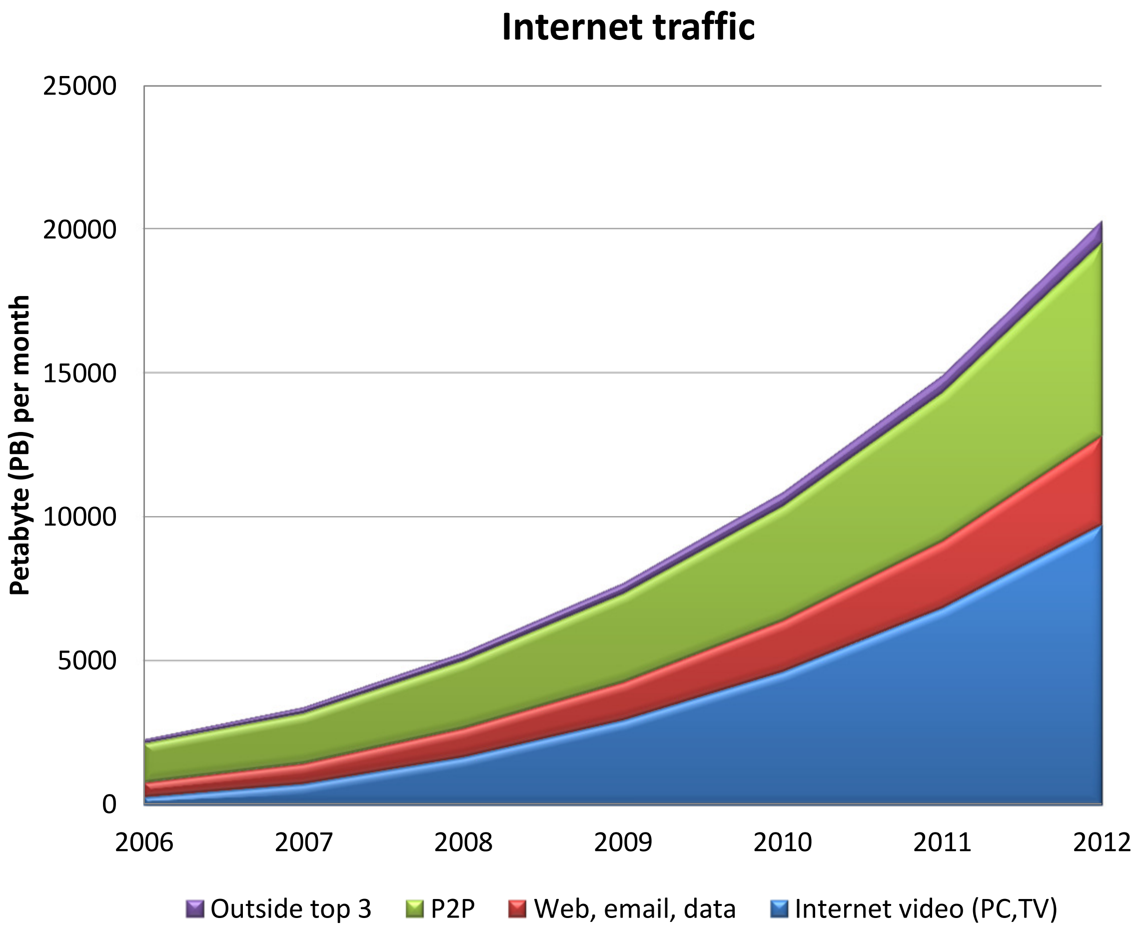


Figure 4.3: Illustration of Internet traffic growth based on data from Cisco [42]. The "outside top 3" category includes Gaming, VoIP and video communication. The illustration show that Internet traffic growth the past 7 years have been dramatic. The compound annual growth rate (CAGR) was 46%. Furthermore, the illustration show Internet video (PC and TV combined) now accounts for the largest Internet traffic type. From only 12% in 2006, Internet video now accounts for roughly 50% of Internet traffic. Cisco anticipate the share of Internet video to be 62% by 2015.

2. Best-effort principle still dominating

Recall that an IP network, at its core, works in best-effort; traffic are treated equally and congestion impacts every data flow. Elastic services like e-mail are perfectly handled in a best-effort network. This is due to the fact that elastic services are *opportunistic* - the content is delivered when network resources are present. The delay caused by congestion does not degrade the quality of the elastic service.

Ever since the rise of real-time applications in the Internet, however, some have claimed that QoS guaranteed is an absolute requirement for timely delivery [43]. However, lateral QoS guarantees is not a general practice across networks and peering points today [35]. The reason for this is, while administrators internally in an IP network easily may guarantee QoS, general QoS policies across peering points is politically and economically difficult. For a flow to have QoS guarantees across public peering points and in other networks, this essentially demands that all the affected ISP must negotiate QoS-policies.

Lateral QoS guarantees is not an absolute criteria for reliable and timely delivery - reliability can be ensured by approaching the problem differently.

The first solution is simply to ensure enough bandwidth along the path. As introduced above, this is solution is popular as Internet bandwidth is quickly on the rise. Tier-2 networks are in general provisioned such that best-effort traffic experience high reliability and throughput [32]. In fact, even in some professional IP contribution networks, reliability is ensured solely with an over-provisioning approach [7].

Bandwidth adaptable services has proved to collaborate well with best-effort networks. A specific example is adaptive video streaming where the host pull (TCP) content in a desired bitrate according to local resources (CPU, memory) and experienced network performance. At congested times, the application pulls a lower bitrate to avoid network congestion, while keeping the playback fluent and smooth for the end user. In fact, giving QoS guarantees to adaptive services may have devastating effects in the network as the prioritized service will claim more and more network resources [35].

But the key solution is to rethink *where* in the Internet the distribution of content, or traffic, originate from. For example, distributing a VoD service to a vast set of users from the same server would potentially cause massive congestion. With a distributed model, however, traffic is spread across the Internet and thus resource competition between users is reduced. They key is therefore to limit router hops, especially peering points.

A recent and very popular trend is to distribute traffic in Content Delivery Networks (CDN) [35]. Here, a content provider typically pays a Tier-1 provider for content distribution into Tier-2 networks. As opposed to a delivering the content from a centralized content server to end-users across the Internet, the content is first delivered with QoS guarantees to proxy servers placed in Tier-2 networks - and then delivered to hosts in that Tier-2 network. Figure 4.4 illustrate distributed content delivery.

As a result, hosts experience reliable services because the number of router hops and peering points are kept to a minimum. Conversely, traversal of a higher number of router hops and peering points with unknown performance, the more vulnerable the packet flow is to degradations such as loss and jitter. This is referred to the middle-mile problem [31].

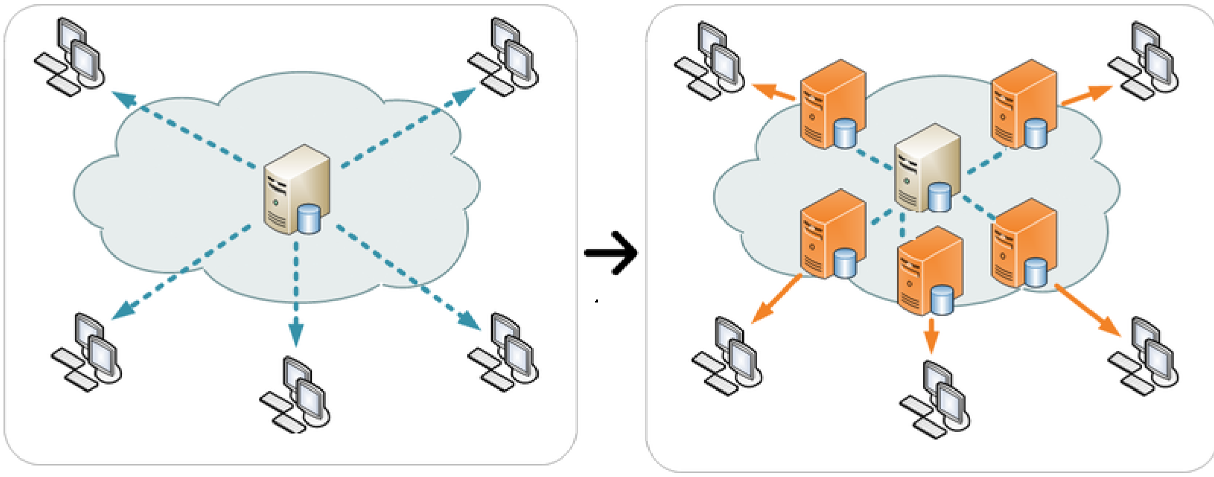


Figure 4.4: Transition from centralized content distribution to Content Delivery Networks (CDN). Figure found at http://cdn-comparison.com/ .

3. Reliability and high performance in Tier-2 networks

Tier-2 networks are far beyond a best-effort network in terms of provisioning, redundancy and technology [32]. As a general fact, Tier-2 networks profit by implementing a more complex network. This is because a complex network is able to reliantly deliver services to their customers (or end users), keeping customers happy.

As a specific example of Tier-2 core network design, a case-study on the Telenor IP/MPLS core network, *BRUT2.0*, is included on the next page.

What is noticeable in this case-study, is the strict dimensioning requirements. Essentially, congestion is avoided because the level of link load must never exceed 70%. It is only in the case of link failure that congestion may occur[4], because the traffic needs to be rerouted to already loaded links. Hence, *congestion* and *packet loss* are not terms, or requirements, used by Telenor IP designers. It is merely a matter of proper dimensioning and link failure avoidance.

As a result of the proper dimensioning, BRUT2.0 have approximately zero packet loss.

Furthermore, this Tier-2 network is dimensioned and fully equipped with router technology and policies to allow QoS-marked traffic. This guarantees performance to services or customers, also in the case of link failure.

Another interesting aspect in the case-study, is the best-effort dimensioning requirement. Because at most 30% of link capacity should be used for QoS-marked traffic, 40%[5] of the network is devoted to best-effort traffic. Therefore, best-effort traffic flows reliably in this Tier-2 network. Many Internet video services are therefore seamlessly delivered as best-effort traffic - for instance, TV2Sumo [35]. It is only in case of link failure when best-effort traffic may experience congestion.

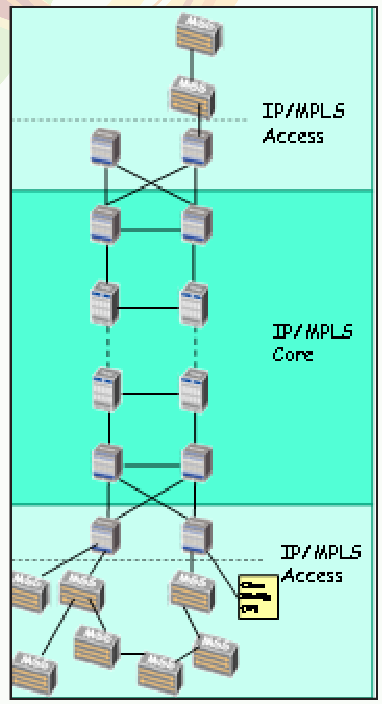
[4] Link failure causing congestion will only affect best-effort traffic. QoS traffic avoids congestion even when link failure occurs due to network redundancy.

[5] Max load level is 70% and max QoS load is 30%, resulting in at least 40% capacity for best-effort.

Case study: BRUT2.0 Telenor Core Network

Telenor is planning and building a new IP/MPLS core network, known as "Brut 2.0". The network should be fully operational by 2013 with over 200 nodes with switching capacity as high as 5.2 Tbit/s. More and more services are carried by this "all-purpose IP network". In addition, traffic is classified according to type of customer; private, firm and wholesale. Thus, the BRUT2.0 is planned to be sufficiently reliable and redundant.

Requirements for BRUT2.0:



Delay

Delay in non-congested traffic scenario:	Max 20 ms / node
Delay in voice service class:	Max 100 ms / node
Delay in business class:	Max 300 ms / node
Delay in best-effort:	Max 10000 ms / node

Jitter

Jitter in non-congested traffic scenario: Max 10 ms / node
Else up to max delay (see above)

Dimensioning

- Non-congested network for critical applications: QoS shall be implemented such that critical customer's SLA are sustained under network congestion.
- Total traffic should never exceed 70% peak capacity on links.
- Sum of traffic class Voice, Premium and Business should never exceed 30% peak capacity on links.

Failure handling

- High availability shall be achieved by redundant infrastructure in core and distribution layer
- Nodal uptime 99.999
- 50 ms convergence in case of link failure (new path discovery delay)
- Links in core must handle all priority traffic + minimum 50% of best-effort traffic if traffic doubles due to failure of neighbouring links

4. Performance and QoS for sale

Tier-2 core networks may offer QoS to professional (or business) customers. As more and more services are becoming web-based - like cloud services - firms need to *trust the ISP* to deliver reliable and secure Internet access. Therefore, a firm would in some cases like to buy QoS guarantees from a Tier-2 provider. Essentially, there are two alternatives for purchasing QoS in Tier-2 networks:

1. **VPN-services with performance guarantees**
2. **Rental or purchase of dedicated lines**

VPN with QoS

In a Virtual Private Network (VPN), secure tunnels are maintained between sites (site-to-site VPN) or from a remote user to a site (remote access VPN). Thus, even though the end-users are geographically separated, they operate as in a private network. Many Tier-2 networks offers such site-to-site VPN solutions with performance and security guarantees. Figure 4.5 illustrates this concept.

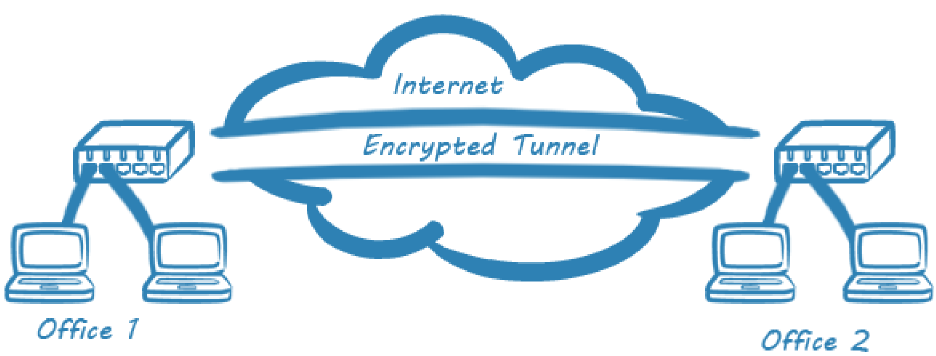


Figure 4.5: Illustration of site-to-site VPN. Figure found at http://www.onemetric.com.au/ .

As a specific example, Telenor offers a solution called *Nordic Connect*. With this solution, a firm can buy dedicated capacity between sites and also remote access. These sites can be placed wherever in the world[6]. The tunnels are created in Telenors IP/MPLS-based core network. Inherently with MPLS, QoS is guaranteed such that, in case of network congestion, users within a Nordic Connect VPN will be prioritized over best-effort traffic in Telenors core routers. Furthermore, within the VPN, real-time traffic can be labelled and prioritized over elastic traffic. As a result, such a solution can offer point-point solutions with reserved and guaranteed level of bandwidth and uptime over the Internet.

Nordic Connect is just one example of Tier-2 VPN services - other Tier-2 networks around the world offer similar solutions. In the case of video contribution, a solution like Nordic Connect may be less costly and more available solution over professional contribution networks. At the same time, such a

[6]Telenor uses its core network for VPN traffic in the Nordic countries, but can also provide worldwide VPN solutions in cooperation with BT Group Telecommunications.

solution offer guaranteed performance and reliability making it more suited than best-effort solution for contribution services vulnerable to network degradations. Finally, such a VPN solution can be offered over a limited time span (for instance, a month) making it attractive for contribution from an one-time event such as a music festival or a football cup.

Rental or purchase of dedicated lines

As mentioned, due the rapid growth of bandwidth consuming applications - ISPs and third-parties are implementing clusters of fiber lines all over the world. Usually, fiber implementation takes a over-provisional approach - more capacity than what is currently needed is implemented. Hence, selling capacity has become viable and cheap and is now a practice of many major telecommunications companies.

Telecommunication companies have established business models to offer capacity rental. A customer can rent a point-point link over which the customers traffic flow in isolation - as opposed to VPN solutions where traffic flows together with other traffic over the same links in the Tier-2 core network.

Specifically in Norway, Telenor offer rental of point-point lines with transmission systems (referred to as *Digital point-point connection* or without transmission systems (dark fiber). Users may buy lines with a capacity ranging from 64 Kbit/s to 1000 Mbit/s with guaranteed uptime.

4.4 Results and discussion

The purpose of this chapter was to introduce recent Internet bandwidth and reliability trends from a contribution perspective.

The first requirement is bandwidth. As section 3.3.1 stated, high-quality contribution require bandwidth above 15 Mbit/s. This can definitely be realized in the Internet, as ISPs now offer 100Mbit/s access links. As stated, [40] expects that access link bandwidth in Norway should average 40Mbit/s by 2015.

Thus, contribution over the Internet is really a question of reliability - can the Internet meet the contribution requirements introduced in section 3.3?

In a best-effort network and over peering points, congestion may occur. As mentioned, typical combat for congestion today is to implement much bandwidth along the lines and also distribute traffic through CDN. The key to reliability is to minimize the number of router hops and peering point traversals. Inside a Tier-2 network, the best-effort traffic can expect reliability as these networks are designed to avoid congestion for best-effort networks. This is absolutely an indication that contribution over the best-effort Internet is realizable; *reliability is ensured by bandwidth over-provision*.

Purchase or rental of dedicated lines and VPN solutions may be an excellent choice for professional contribution demanding QoS guarantees. Not only guaranteeing QoS, this solution protects against DoS attacks, network link failure (the QoS traffic has first priority also in the case of rerouting) and congestion at access nodes. In fact, some professional IP contribution networks already use this solution; Media Netwerk rent capacity, both dedicated capacity and VPN, from Ventelo in some cases to expand their network [7]. Such rental, however, is kept to a minimum in IP contribution networks because complete control is desired. That is, complete control of bandwidth booking and link monitoring is only possible inside the IP contribution network.

Chapter 5

Internet performance measurements

In the introduction, we asked *Can a contribution packet stream conform to contribution requirements?* This chapter will measure Internet QoS metrics such as packet loss, reordering, delay, jitter and throughput for a packet stream between a pair of Internet hosts, and compare the measured QoS to contribution QoS requirements. The motivation for this is to investigate whether the Internet is contribution ready, and identify challenges using the Internet for this purpose.

In addition to comparing the measured QoS metrics to QoS contribution requirements, this section will capture *typical network impairment patterns*. Such patterns are typical occurrences of errors - for instance; length of a typical packet loss burst, delay and loss correlation, a typical time distance between consecutive losses, etc. Having captured such patterns, we can discuss if the Internet is contribution ready. Also, the most common degradation patterns will point to the most critical transport protocol features for contribution.

5.1 Background

Measuring Internet performance in terms of throughput, packet loss and jitter - referred to as network impairments - is an important yet a difficult topic of research. In practice, network performance tests are used at a everyday basis by hosts, network architects, network application designers and network engineers to test or verify network bandwidth and reliability. In research, such tests can be used to capture Internet impairments which, in turn, allow better routing and transport protocol design. In short, network performance tests are actively used by all participants in the Internet, but differs vast in motivation, methodology and consistency.

Paxson did one of the first, influential studies on typical Internet impairments by tracing 20,000 TCP bulk transfers between 35 Internet sites [44]. *He found that packet loss within a short time frame are correlated*. In other words, packet loss often appear in packet loss bursts. Nguyen et al. verified this by proving that *packet loss is correlated within 1000 ms* [45]. This study analyzed more than 100 hours of packet traces from PlanetLab measurements over the Internet. Furthermore, the results showed that packet loss runs (bursts) and successful packet runs are uncorrelated; one cannot predict loss runs from successful runs or vice versa. These findings are most interesting, indeed, but these test were conducted with UDP stream with a small packet size (40 bytes) with few packets per seconds (50/sec). Therefore, these results may not necessarily directly translate to video contribution over IP using maximum packet size (MTU) and much higher bandwidth. This is further investigated here.

In our prior work on contribution over the Internet, network impairments were recorded between Trondheim and hosts at Oslo, Bergen, Lillehammer, Alta and San Diego [9]. A RTP/UDP/IP packet stream conforming to SMPTE contribution requirements (see section 3.3.2) was sent between a pair of hosts. Each test lasted for one minute. The impact of various packet sizes (up to MTU, 1500 bytes) and various bandwidth (up to 25 Mbit/s, as constrained by the first-mile) were analyzed. The results showed first of all that very similar network impairments were recorded regardless of geographical placement of the hosts. No packet loss were recorded over any connections. Furthermore, the results strongly suggested that packet size equal MTU should be used, as a smaller packet size resulted in loss due to a higher number of packets per time. What is missing from this research is longer recordings - hours instead of minutes - capturing more complete data for a given connection. Also, testing with higher bandwidth than 25 Mbit/s should be investigated.

In chapter 4, we showed that first-mile bandwidth is rapidly increasing. Average first-mile bandwidth is now 12 Mbit/s, and is expected to be 40 Mbit/s by 2015. In cities and areas with high population density, access links with 100 Mbit/s bandwidth is offered. This section should investigate the reliability and performance of such high bandwidth access links.

The investigation can be done with a client-server application allowing active measurements of packet flow QoS. To guarantee that the best-effort Internet is tested, the packet flow should traverse one public peering point. The measured QoS metrics should be compared to contribution QoS requirements introduced in section 3.3.1 namely contribution bandwidth classes (DVB) and contribution network profiles (ITU-T). Using these requirements, the tests in this chapter can more easily be categorized and compared.

As stated above, we expect to find some network impairment patterns. Although the Internet is basically a collection of networks with unequal performance, Internet workings and design is based on some principles which may result in typical impairment patterns. For instance, as described in chapter 4, the best-effort principle applies for connections traversing peering-points. Also, roughly 90% of the data traffic is transported over TCP and Tier-2 networks are generally over-provisioned.

Given such network impairment patterns, a discussion on the most critical transport features can be done. As mentioned in 3.2.3, congestion control and error control put requirements on every other module in a real-time delivery system for video. This chapter should therefore discuss the importance of each feature based on the collected data. This, in turn, impacts the choice of transport protocol - TCP or UDP - for a contribution system over best-effort Internet.

5.2 Test hypothesis

We have chosen multiple hypothesis in our testing.

Hypothesis 1: The Internet conforms to contribution network profile B requirements.

Refer to table 3.4 for contribution network profile B requirements. Recall that the Internet as a best-effort network is classified as contribution network profile C. If the hypothesis is proven, i.e. Internet conforms to profile B, then Internet QoS is better than what is defined in ITU G.1050. Profile A was not chosen as a target because this profile does not tolerate packet loss bursts which likely will occur in the Internet.

Criteria: This hypothesis is confirmed if the majority of QoS statistics conforms to the QoS metric requirements in contribution network profile B.

Hypothesis 2: Packet loss in the network is the most critical QoS metric.

Packets must be present at the receiver within the playback deadline. This deadline is missed if the packet is lost in the network, i.e. packet loss, or if the packet arrive too late due to extensive delay (jitter). This hypothesis suggests that no jitter (IPDV) will result in a missed playback deadline given a jitter buffer. The maximum jitter is set to 50 ms which is the requirement in contribution network profile A[1]. The impact of this hypothesis is that, if proven, packet loss is the most critical QoS metric and error control to cope with packet loss is more important than implementing longer buffers to cope with jitter in a Internet contribution system.

Criteria: This hypothesis is confirmed if IPDV (see section 3.2.2) is below 50 ms in the tests. Data from all UDP tests should be used to prove this.

Hypothesis 3: Packet loss bursts is less than 1000 ms in length.

This hypothesis suggests that no packet loss bursts is longer than 1000 ms. If proven, the result has quite important impact on congestion control implementation, because the network is only lightly congested. Also, if proven, error control is much easier to design and implement because the a packet loss burst will never be longer than 1000 ms.

Criteria: This hypothesis is confirmed if packet loss bursts using UDP is consistently less than 1000 ms.

[1]Normally a jitter buffer of size 50-100 ms is used in professional contribution. Our IP contribution gateway equipment, TVG430/450, allows up to 100 ms jitter buffer [46][8]

5.3 Test methodology

The Internet performance measurement was so called *active-passive*. *Active* in that a real IP packet flow was sent from the sender (typically an event) to the receiver (typically a broadcast center) allowing measurements and analysis - *Passive* in that the packet payload was meaningless data.

The sender and receiver ran an application able to transmit or receive UDP and TCP streams. This application was developed prior to this thesis; design and user interface details are described in [9]. Essentially, the sender pushed an UDP[2] or TCP stream of data to the receiver. Both the sender and receiver recorded packet headers (outgoing or incoming) using Wireshark [47]. This allowed later analysis of packet loss, delay, jitter and re-ordering using ad hoc software developed in C (programming language).

Due to the increase in Internet bandwidth, contribution bandwidth class D (DVB requirement, see section 3.3.1) is now theoretically possible in the commercial Internet. We therefore wanted to test with a packet flow conforming to bandwidth class D and investigate the reliability at these sending rates. The conducted tests was done with sending rates at 70, 60 and 50 Mbit/s which are within bandwidth class D.

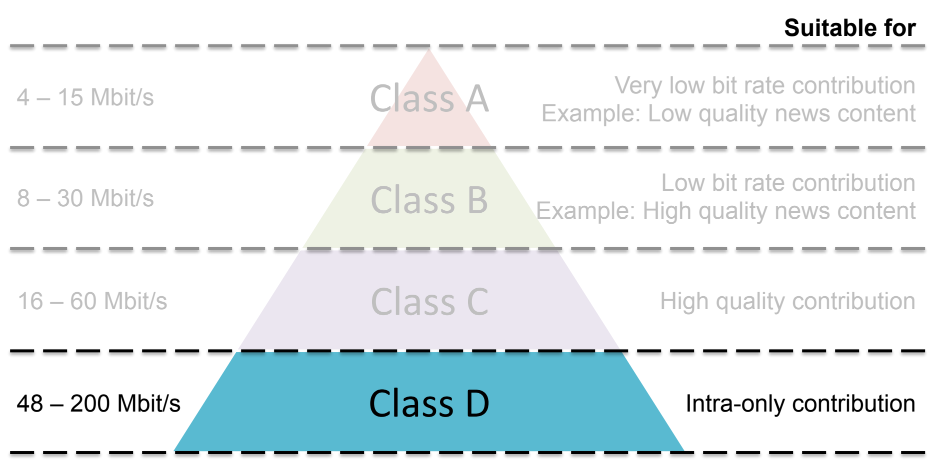


Figure 5.1: Testing was conducted in contribution bandwidth class D at 70, 60 and 50 Mbit/s sending rate.

In addition, tests with 25 Mbit/s sending rate were conducted conforming to contribution bandwidth class C. We did this to allow a discussion on whether a drop in sending rate results in more reliable contribution.

It is important to point out that we separate results from discussion. The results section is solely based on data objectively proving or disproving our hypothesis - no discussion, or subjective elements are included in the results section. Therefore we chose hypothesis that can be evaluated objectively, for instance conformance to a specific contribution network profile. The results are used as a basis in the discussion section, where more indecisive topics are presented.

We criticize the testing methodology in section 8.2.

[2]The UDP packets have sequence numbers allowing tracking of sent and received packets.

5.4 Test setup

We chose to only test one connection in our testing, i.e. the sender and receiver were placed geographically permanent in the Internet. This is criticized in section 8.2.

Choosing the route, i.e. connection between the sender and receiver, was difficult. We wanted to test the Internet using commercial access and core networks and one peering point. This guarantees best-effort and competition with other commercial traffic. The receiver was therefore placed at private home at Prinsen in Trondheim, with Canal Digital (Telenor) as ISP. The access link was a VDSL connection limiting the sending rate to 70 Mbit/s. The sender was placed at NTNU in Trondheim. NTNU, as any other Norwegian university, has Uninett as ISP. Uninett is a private, high-capacity network for research and educational use. Appendix D shows that this core network is very lightly loaded (around 20%) and have much bandwidth (10Gbit/s).

Figure 5.2 illustrates the route. The figure is briefly explained step by step below.

The packets flowed from the sender (on the top) to the receiver (at the bottom) - as illustrated by the green arrows. The traffic entered the Uninett access network in which a couple of routers (illustrated by blue boxes) forwarded the traffic. Next, the traffic entered the Uninett core network. Three routers forwarded the traffic into the peering point NIX1 in Oslo.

Next, Uninett and Telenor peered at NIX1 in Oslo - illustrated by the green box. For reasons unknown, the peering happened at NIX1 in Oslo - as opposed to a TRD-NIX in Trondheim where the both sender and receiver were placed. As far as test setup, this was actually great because the traffic traveled longer (approximately 1000 km) and more routers were in the path; this possibly gave more varying results. The NIX1 box in the figure show traffic volume with respect to time for NIX1. uio.no allows users to dump statistics from NIX1 [33] - this dump, in particular, is from a normal week in April (2012). The graph show similar traffic patterns on weekdays (expect friday) with a peak between 18:00 and 22:00. Therefore, each test was conducted in this time period. By testing in the most busiest time frame, we possible captured worst-case Internet performance.

Inside the Telenor core network, the traffic traversed five routers from Oslo to Trondheim. Finally, the traffic entered the access network where two routers forwarded the traffic to the receiver location.

The arrows on the left specifies bandwidth and delay, respectively, for a part of the route. The bandwidth was found by running Internet performance test provided by nettfart.no. This website allows bandwidth measurement into NIX1 in Oslo. The maximum bandwidth for the connection was therefore 71 Mbit/s. The delay data was found by using a traceroute (see [9] on traceroute description). The total delay for the connection was 21 ms.

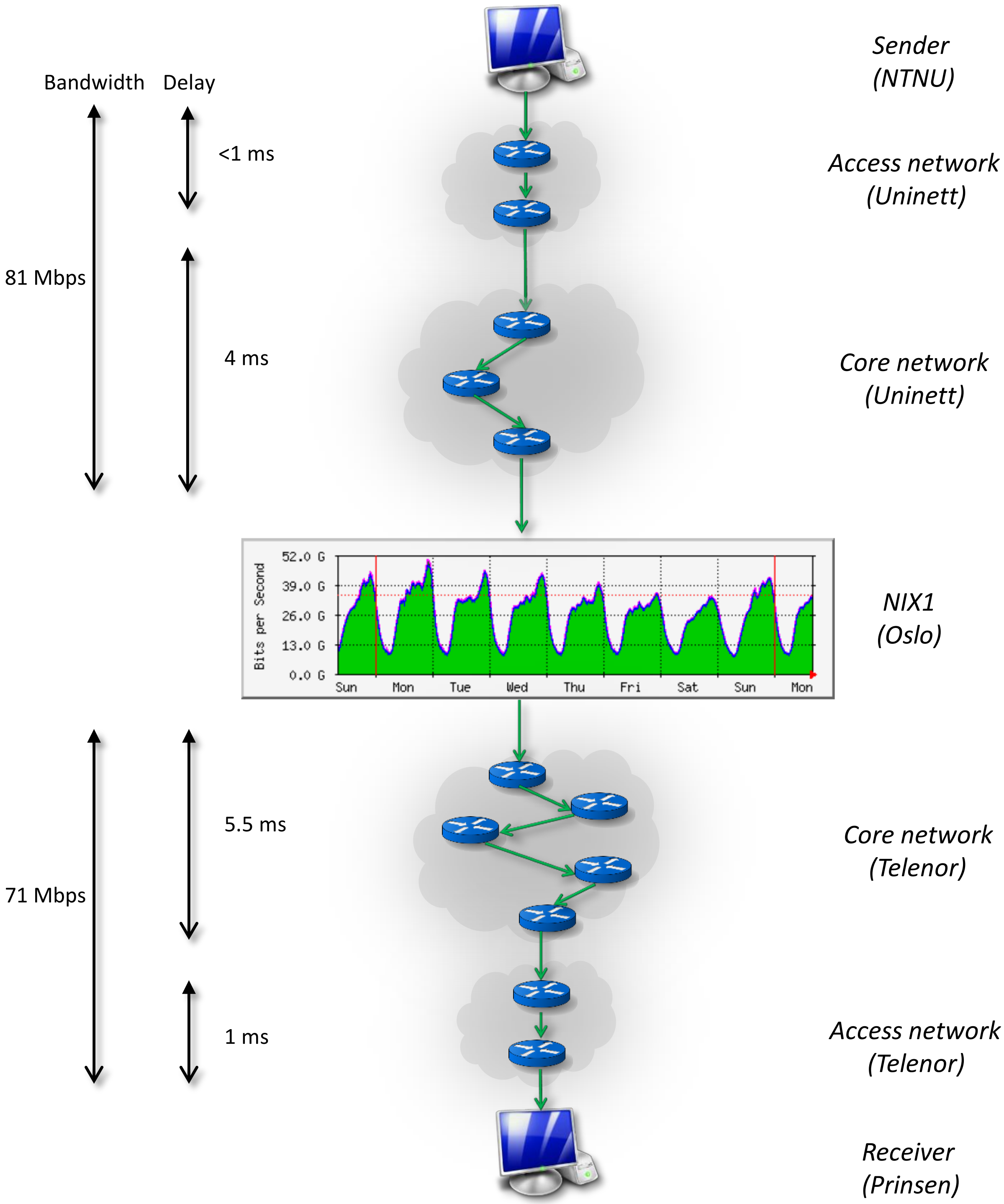


Figure 5.2: Figure showing the route between sender and receiver.

5.5 Results

This section will present the results for each hypothesis. Totally, 11 tests over 11 days were conducted at 25, 50, 60 and 70 Mbit/s sending rate. Totally, over 21 hours of network statistics were collected.

Hypothesis 1
Hypothesis: The Internet conforms to contribution network profile B requirements.

To appreciate the results, the reader should restudy the contribution network profile requirements in table 3.4. Detailed results from all test are presented in appendix B.11. The main findings are presented in table 5.1 and illustrated in figure 5.3.

The key findings illustrated in the table are:

- The QoS metrics packet loss rate (per cent), IPDV, reordering and random packet loss (singles) from all tests conform to contribution network profile A. Thus, in terms of these metrics - the results was consistent and conform to professional contribution requirments.
- No packet loss burst length exceed contribution network profile B requirement (200 ms).
- PLBR, i.e. average occurrence of packet loss burst per time, is the most critical network impairment. The PLBR values in the tests are consistently placed on the border between contriution network profile B and C. Six tests (3,4,6,7,9,11) conform to profile B while 5 tests (1,2,5,8,10) conform to profile C.

As a conclusion, the tests are split between contribution network profile B and C. Therefore *the hypothesis is both proved and disproven*. PLBR is the most serious network impairment, and is the most important problem to solve going forward.

Hypothesis 2
Hypothesis: Packet loss in the network is the most critical QoS metric.

The question here was whether packet loss that are lost in the network (never arriving) are the most critical QoS metric, compared to packet loss (missed deadline) caused by extensive delay (jitter). The requirement was 50 ms from contribution network profile A. As figure 5.3 (upper-right) illustrates, the highest measured IPDV from all tests was 43.9 ms. It is therefore safe to say that *no packet loss occurred due to jitter* given a 50 ms jitter buffer. Therefore, *this hypothesis is confirmed*.

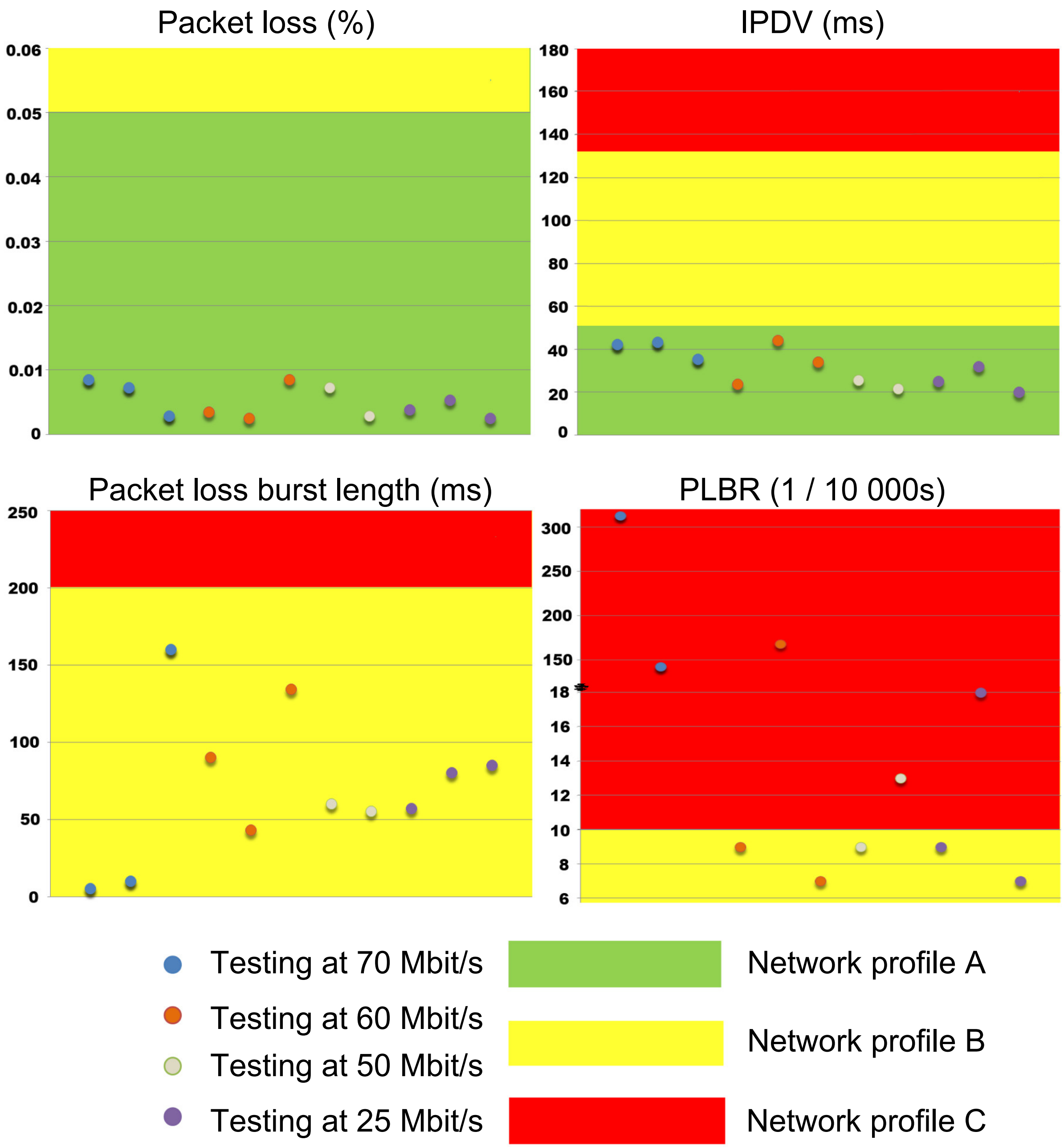


Figure 5.3: Plots for all tests versus contribution network profile requirements. Each test is represented by a dot in each graph. The legend is printed above. Upper-left: packet loss percentage. Upper-right: IPDV in ms. Lower-left: maximum packet loss burst in length (ms) for each test. Lower-right: Packet loss burst rate (PBLR), note that the axis is truncated. In summary, the graph show that packet loss and jitter is within profile A. Packet loss burst length is within profile B. PLBR is divided between profile B and C.

Test setup				Resulting QoS metrics			Packet loss distribution			
Test	*Sending rate*	*Duration*	*Packets sent*	*Packets lost*	*IPDV*	*Reordering*	*Singles*	*Packet loss burst length*		*PLBR (1/s)*[a]
								$<$ 200ms	200ms$<$	
1	70 Mbit/s	1.5 hrs	32141919	2701 (0.0085%)	42.1 ms	3 (0%)	12	169	0	0.0312
2	70 Mbit/s	1.5 hrs	32141919	2318 (0.0072%)	43.0 ms	2 (0%)	5	77	0	0.0142
3	70 Mbit/s	1.5 hrs	32141919	1690 (0.0028%)	35.1 ms	2 (0%)	5	5	0	0.0009
4	60 Mbit/s	1.5 hrs	27550464	1047 (0.0038%)	23.6 ms	2 (0%)	0	5	0	0.0009
5	60 Mbit/s	1.5 hrs	27550464	1459 (0.0053%)	43.9 ms	2 (0%)	7	91	0	0.0168
6	60 Mbit/s	1.5 hrs	27550464	1663 (0.0060%)	33.9 ms	1 (0%)	1	4	0	0.0007
7	50 Mbit/s	4 hrs	61276447	1679 (0.0027%)	25.3 ms	3 (0%)	6	13	0	0.0009
8	50 Mbit/s	4 hrs	61276447	1917 (0.0031%)	21.4 ms	2 (0%)	12	19	0	0.0013
9	25 Mbit/s	1.5 hrs	11489299	207 (0.0017%)	25.0 ms	2 (0%)	0	5	0	0.0009
10	25 Mbit/s	1.5 hrs	11489299	397 (0.0034%)	31.8 ms	2 (0%)	3	10	0	0.0018
11	25 Mbit/s	1.5 hrs	11489299	297 (0.0025%)	19.9 ms	2 (0%)	1	4	0	0.0007

Table 5.1: QoS statistics for all tests.

Cells are colored green, yellow and red depending on conformance to contribution network profile A, B and C, respectively.
[a]Packet loss burst rate: the rate of packet loss bursts (1/s).

Hypothesis 3
The packet loss bursts is less than 1000 ms in length.

Figure 5.4 plots every recorded packet loss burst for all tests. In this figure, each dot represent a packet loss burst. This figure clearly show that every packet loss burst is below 1000 ms in length. Therefore, *this hypothesis is confirmed*. In fact, every packet loss burst was below 200 ms - which is the requirement for contribution network profile B.

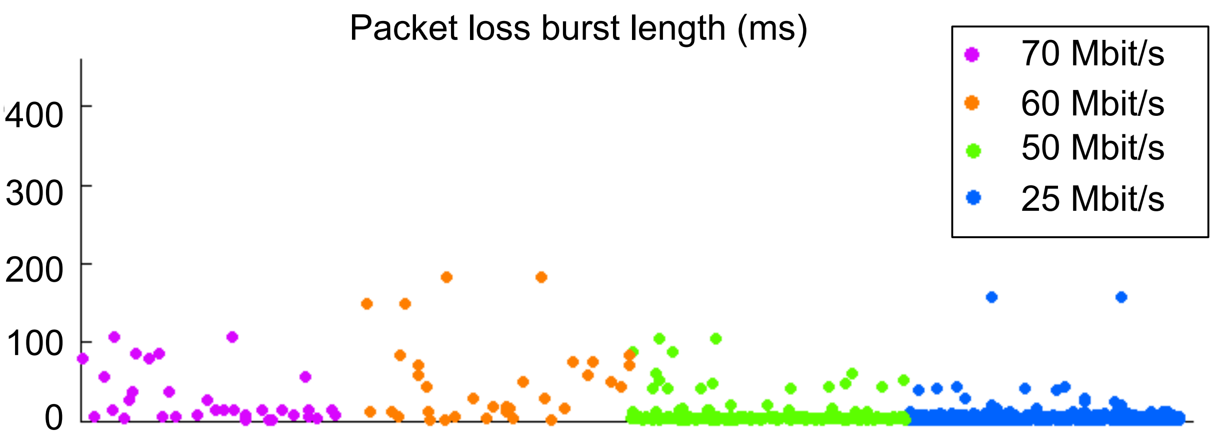


Figure 5.4: Plot of all packet loss bursts from all tests.

Figure 5.4 also show that a lot of small packet loss bursts occurred when testing at 70 and 60 Mbit/s, because the graph is very dense around 0 at these sending rates. This was actually caused by Test 1 (70 Mbit/s) and Test 5 (60 Mbit/s) where a lot of small bursts occurred. This was not, however, the case in other test at the same sending rate (Test 2,3 and Test 4,6) in table 5.1.

In summary, the test were consistent in that no burst exceeded 200 ms. Packet loss burst frequency and length, however, cannot be generalized in our tests due to high variability.

5.6 Discussion

Recall our research question; *Can a contribution packet stream conform to contribution requirements?* In terms of contribution requirements, we have used contribution network profile B and contribution bandwidth class D as the target in our testing. Even though these are not necessarily requirements from a broadcaster, it is sensible to have a clear test framework and target.

The results show that the testing does not consistently conform to contribution network profile B, as the results are categorized in contribution network profile B and C. The rate at which packet loss bursts occur (PLBR) is the one metric hindering the Internet to consistently conform to contribution network profile B. Can this problem be solved such that our research question is confirmed?

Congestion control

Can congestion control solve the problem with PLBR? Recall that congestion control imply varying the sending rate according to network loss and delay levels. One success criteria for congestion control is that *dropping the sending rate results in less packet loss* (resulting QoS metrics). Arguably, the purpose of congestion control is also for network *fairness* and scalability; at congested times it seems fair that every host drops the sending rate to avoid network traffic overload. However, fairness is not within the scope of this report.

Low levels of packet loss and jitter were consistently measured in every test, which indeed is a sign of low congestion levels in the networks the packets traversed. This was actually no surprise. In chapter 4, the commercial Telenor core network (BRUT2.0) was described - the same network core used in testing. It was stated that this core network is dimensioned to be at most 70% loaded. Uninett is even less loaded; appendix D show that the link between Trondheim and Oslo is not more than 20% loaded. As a result, we expected the network have low congested levels, and this was confirmed.

By studying table 5.1, there is no consistent tendency that dropping the sending rate from 60-70 Mbit/s to 25 Mbit/s solves the PLBR problem because the high PLBR values were present regardless of rate. For example, in Test 3 (70 Mbit/s) the PLBR level was as good, or better, compared to Test 9 (25 Mbit/s) and Test 10 (25 Mbit/s). Furthermore, Test 6 (60 Mbit/s) had lower or equal PLBR compared to all tests at 25 Mbit/s. The test results are variable and suggests that congestion control, i.e. dropping the sending rate, does not necessarily solve the packet loss burst problem because the problem remains when sending at 25 Mbit/s.

TCP versus UDP

Another important discussion is the choice of transport protocol for contribution over the best-effort Internet. As mentioned, TCP carries roughly 90% of all data in the Internet, but it is not necessarily the preferred protocol for contribution.

Given our test data, using TCP as a transport protocol would yield a very fluctuating sending rate. Figure 5.5 illustrate the distribution of the distance (time) between two consecutive packet loss bursts from all tests. The figure clearly show that packet loss bursts tend to cluster as most bursts are separated by 10-50 seconds. This would cause periods with massive fluctuation in the TCP sending rate due to congestion control; for every packet loss burst, TCP would dramatically reduce the sending rate and use seconds to re-establish the sending rate. As a result, the throughput in these periods will be critically low and may cause buffer starvation and system failure.

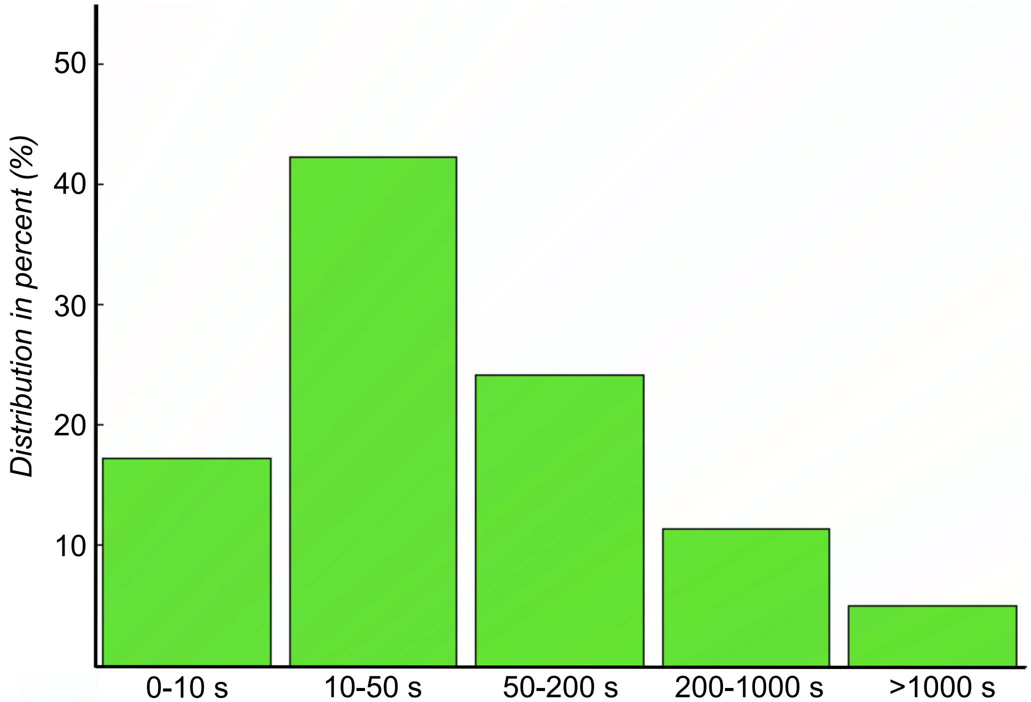


Figure 5.5: Distribution of the distance (time) between two consecutive packet loss bursts from all tests.

To confirm the fluctuating sending rate with TCP due to congestion control, we did a TCP throughput test. This test was done using the same setup (described in described in 5.4) as the UDP tests displayed in table 5.1. The test lasted from 17:00 to 23:00 on a Wednesday evening.

Figure 5.6 show the resulting throughput from 17:00 to 17:30. First of all, the figure confirms the fluctuating sending rate with TCP. Secondly, the time to re-establish the sending rate after a packet loss is quite considerable. This is illustrated by the red graph; here a packet loss occurred and it took 13 seconds to re-establish the rate. In this 13 seconds period, the average throughput was only 14 Mbit/s. Third, the average throughput was very low in some periods; the figure points at a 3 minutes interval where the average throughput was only 22 Mbit/s.

Recall that the UDP tests in table 5.1 was done over the same connection at the same time of day as the TCP test. In common for all UDP tests was a stable throughput at the specified sending rate (see appendix B.11). Therefore, it is reasonable to assume that *a UDP flow would have been stable under the same network conditions as the TCP test*.

In summary, UDP is preferred over TCP due to less fluctuating throughput. It seems like good solution to use UDP without congestion control for contribution over the Internet. However, the problem with packet loss bursts is yet to be solved. Therefore, UDP must be extended with some kind of error control to enhance reliability.

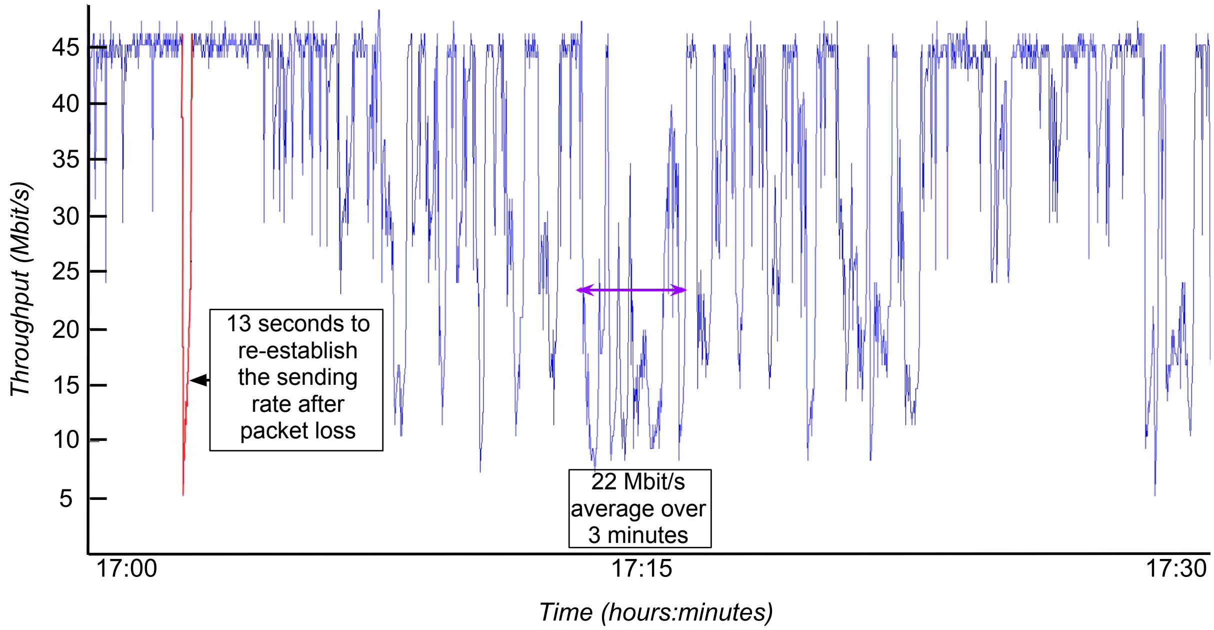


Figure 5.6: TCP throughput recorded 15/02/2012 from 17:00 - 23:00. Here, the period from 17:00 - 17:30 is illustrated.

Error control

Error control handles packet loss. As mentioned in section 3.2.3, two possible error control strategies are Forward Error Correction (FEC) and retransmission. Can these strategies solve the problem with packet loss bursts?

From the testing, two clear facts have been revealed about the packet loss burst patterns:

1. No packet loss burst is above 200 ms in length (see figure 5.4)
2. No packet loss burst happens within 2 seconds after the previous packet loss burst. This is illustrated in figure 5.7. Here, the distance in time between the successive packet loss bursts are plotted. The minimum value is 2 seconds[3].

Based on our results, FEC as we know it from professional contribution, is useless for contribution in the best-effort Internet based on two facts; The level of correction is too low compared to the burst length. The FEC settings used in professional contribution typically cannot correct bursts above 5 ms. Increasing the level of protection, increases the calculation delay at the sender. Given that bursts of 200 ms happens at most once every 2 seconds - the FEC buffering delay at the sender is 2 seconds which is high. The second fact (2) is, given that packet loss burst occur between time *a* and *b* in the media stream, then packet loss often occur in the FEC between *a* and *b*. In this case, the FEC stream is totally corrupted and useless. Because short congested periods (200 ms) recorded, this will happen.

[3]Test 2.

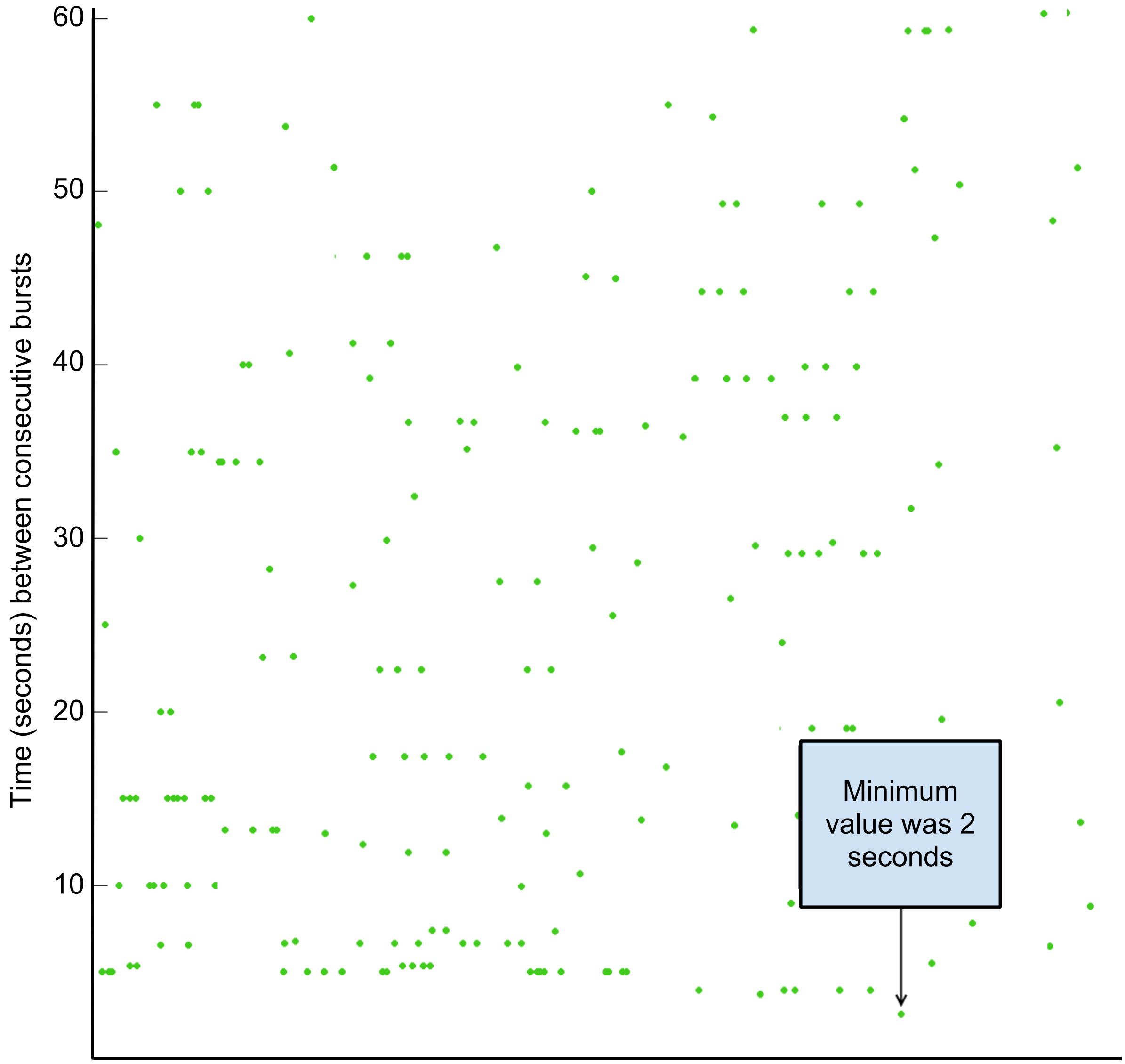


Figure 5.7: Scatter of time between consecutive packet loss bursts from all tests. Only distances below 60 seconds are showed here.

There is no question that a retransmission strategy would solve the given packet loss burst problem. The question is rather at what cost in terms of delay. Table 3.4 show that 100 ms is the maximum delay budget for contribution network profile B. In our testing, the network is sometimes congested for a time interval above 100 ms - in which *all* data is lost, including retransmission requests. Retransmission in this scenario require a delay budget above 200 ms to cope with the recorded packet loss bursts. Increasing the delay above 100 ms, however, would result in failure to conform contribution network profile B because of delay requirements - even though the burst problem is solved.

In summary, conformance to contribution network profile B seems difficult. To enhance reliability, given our test data, one must *enlarge the delay budget above 100 ms* for error control strategies to be successful. A delay budget above 100 m, however, conforms only to contribution network profile C.

At this point it is sensible to leave the contribution network profile requirements. As mentioned, these are intended for network performance in general and may not apply for all occasional contribution setups. That is, for many occasional contribution applications, a larger delay budget can be allowed [8]. So the question is, how much must the delay budget be increased to solve the PLBR problem? We will illustrate two possible solutions to find the answer to this problem.

Retransmission: A simple retransmission strategy is illustrated in figure 5.8a). The figure illustrates an example with a packet loss burst of 200 ms - which is the worst case in our findings[4].

The packet loss burst starts at time *a*, and the receiver waits until the packet loss burst stops at time *b* before requesting a retransmission for the packets lost in the interval (*a*,*b*). The retransmission request takes half the roundtrip time (RTT) and is done via the feedback channel (illustrated in figure 3.5). The first retransmission packet is sent immediately (no latency), thus arriving at the receiver half the RTT later. *The complete retransmission process takes therefore the time equal the packet loss burst length, in this case 200 ms, plus RTT.*

As mentioned in section 3.2.3, retransmission require buffers such that the content in the buffers is played back while retransmission happens. Immediately after the retransmission, the buffers need to be refilled - or else the next packet loss burst would result in buffer starvation. As an example given 70 Mbit/s first-mile bandwidth, appendix C shows that 62 Mbit/s should be used for media and 8 Mbit/s should be used for buffer refilling. Here, the given buffer size was set to 250 ms (RTT was set to 50 ms) and the buffer refilling time budget[5] was set to 2 seconds because this was the minimum recorded time distance between two consecutive bursts in our testing.

Delayed backup stream: a possible solution could be a delayed backup stream as illustrated in figure 5.8b). The idea is that the main media IP stream with video frames coded at some bitrate is sent on one port, and a copy of the main media stream at a much lower bitrate is sent on another. The delayed backup stream is thus an independent video stream, *not* an error correcting stream, with fair video quality for the end users. Furthermore, this backup stream is delayed such that when packet loss burst occur in the media stream, a backup version of the lost frames with fair quality arrives some time after. The playback system, which delayed until the arrival of the backup stream, may therefore chose content from the backup stream when the media stream is corrupted.

[4] The longest recorded burst was 169 ms in Test 1.

[5] This is the maximum time it should take to fill the buffers.

Here, it is sensible to chose the backup stream delay to be 200 ms - because this was the largest packet loss bursts. Hence, *the delayed backup stream would always arrive uncorrupted given our test data* and therefore the playback will not fail. *The total delay is therefore 200 ms*.

The tradeoff is obviously the drop in video quality when using the backup stream. However, it was found in [48] that a sudden drop of video bitrate using JPEG2000 may not be visible to the viewer. Specifically, it was found that alternating the bitrate from 30 Mbit/s to 15 Mbit/s and back to 30 Mbit/s for a time window of 2000 ms - the subjective quality remains unaffected in most cases. This ultimately implies that using a main media stream at 30 Mbit/s and a backup stream at 15 Mbit/s - the packet loss bursts in the main media stream is masked by the parallel delayed backup stream.

In summary, the tradeoff between enhanced reliability and delay in error control was found to be in the range of 200 ms, because no packet loss burst longer than 169 ms was recorded. Because no two bursts occurred within a 2 second time window, it is guaranteed (in our test data) that retransmission requests or a parallel delayed backup stream arrives uncorrupted.

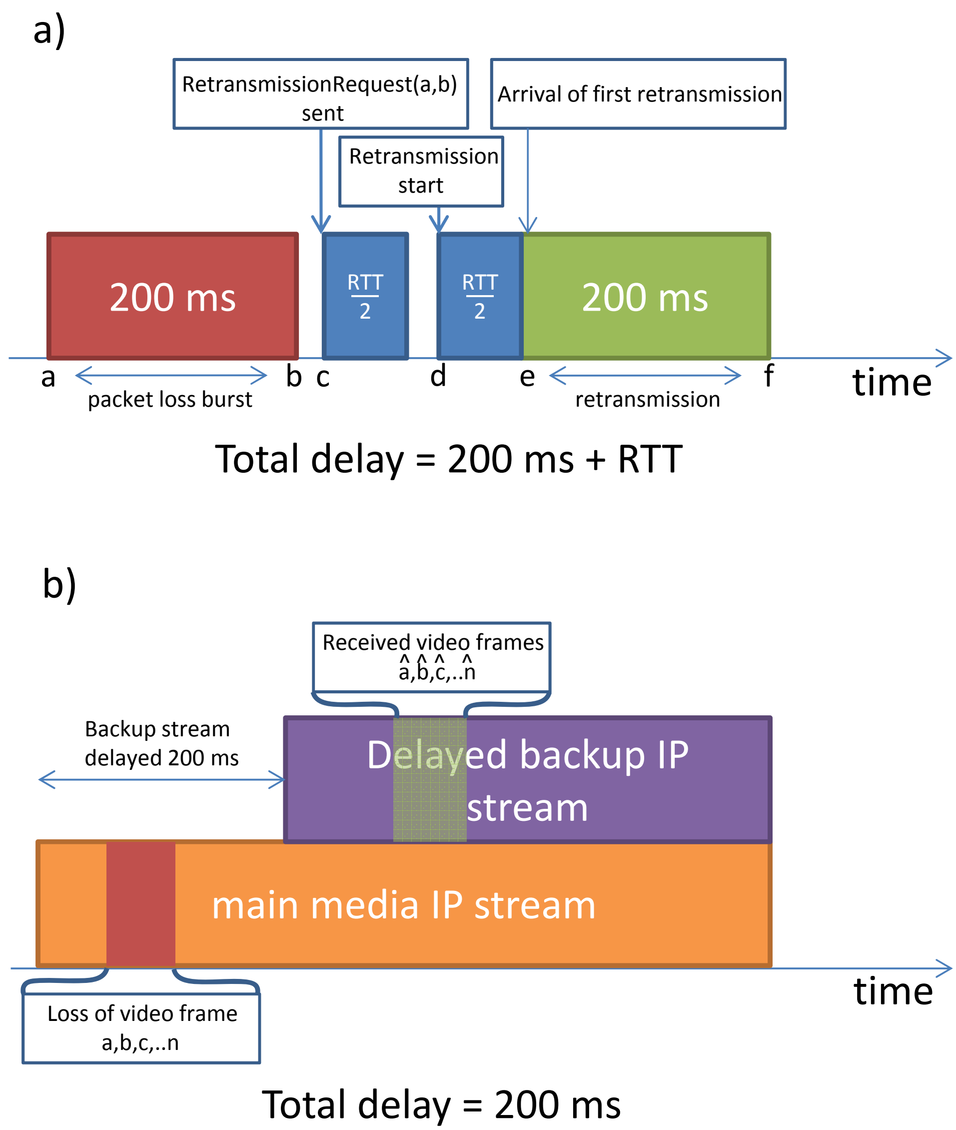


Figure 5.8: a) A simple retransmission solution, b) Error correction solution using baseline correction stream.

5.7 Summary

In the introduction, we asked: *can a contribution packet stream conform to contribution requirements?*

This chapter has shown that in terms of packet loss (singles), jitter and re-ordering - the packet streams indeed conform to contribution requirements (contribution network profile A, contribution bandwidth class D).

The packet loss burst and the rate thereof is the only reason why approximately half of the tests conform to contribution network profile C and contribution bandwidth class D. Contribution network profile C is probably outside the range of what is tolerated in occasional contribution.

However, the research question asks if a packet flow *can* conform to contribution requirements. This chapter has shown that an UDP contribution stream with error control, absolutely can yield reliability conforming to contribution requirements - *if the broadcaster accepts a delay and bandwidth overhead*. A retransmission strategy with approximately (depending on RTT) 250 ms delay tradeoff and 10% bandwidth overhead was illustrated to correct the packet loss bursts. Alternatively, a parallel delayed base stream can mask the packet burst length with minimal subjective degradation, 200 ms delay tradeoff and approximately 30 % bandwidth overhead.

Chapter 6

Testing video contribution over Internet

The last question we asked in the introduction was: *Does contribution over the Internet meet TV viewers expectations in the distribution phase?* This is definitely an important and relevant question for broadcasters who want to use Internet contribution in their TV production. Their main goal with respect to quality is to meet or exceed their customers expectations. Thus, our challenge is to construct and conduct tests to answer this.

We will do this through two tests: A "Proof Of Concept" (POC) test, and a subjective test. In both of the tests we will use equipment made for contribution over IP contribution networks. This equipment is completely compatible with the Internet, so the setup is basically "plug and play".

The POC test will focus on the occurrence of packet loss *as it happens*. A small test panel of both experts and amateurs will record and rate periods of noticeable degradations. This will give a feel to how the video was perceived, and outline the level of quality provided by the system.

The subjective test aims at pinpointing the level of quality more precisely. Unlike the POC test, the quality rating will be done in retrospect. That is, a larger set of participants will watch a complete movie and surveyed thereafter. Central for the design of the survey is the concept of Quality of Experience, presented in section 3.4. In our case, "watching TV" is the experience we want to measure, and the "level of quality" is the quality of this experience. To further strengthen the integrity of our results, we developed a research protocol prior to the testing. This clearly describes the background and the test setup and how the results are to be presented and analyzed.

To our knowledge, no academic study on contribution over the Internet has been conducted. Therefore, this thesis has not only focused on exploring the topic in a broad manner, but also on exploring different test methods to get the most enlightening results. Since this is a fresh topic in an academic setting, we seize the opportunity to suggest a general technical test framework that can be used for further studies.

6.1 General technical test framework

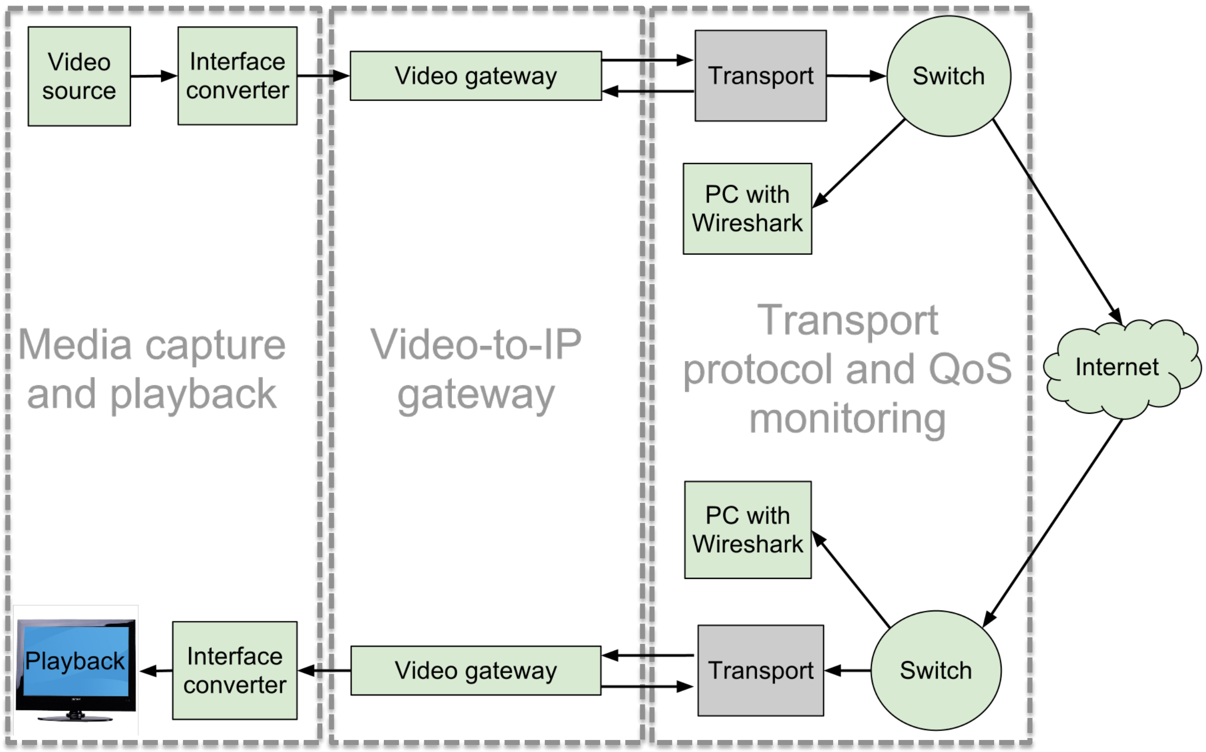


Figure 6.1: A general technical setup for testing contribution over the Internet.

Figure 6.1 shows the technical setup we suggest for conducting research of contribution over Internet. The different components in the figure are abstracted into three groups:

Media capture and playback
Consists of a video source, a monitor to playback the resulting video and interface converters between the video gateways and the video source/playback system. In actual contribution, the video source would be a camera with SDI output capturing live scenes, but in a test setup, a PC with prerecorded video material is sufficient. The video material should be of high quality without visible encoding artifacts.

Video-to-IP gateway
Every contribution system over IP has two video gateways; one video-to-IP gateway in transmit mode compressing raw video material and output IP packets, and one IP-to-video gateway in receive mode that takes in IP packets, decodes the compressed video and output uncompressed video.

Transport protocol
As defined in section 3.2.3, a real-time system over best-effort IP network may employ a transport module that does online rate control and implement transport protocol features like error control and congestion control. Such features will not be tested in this chapter. The task is rather to investigate how the system works without such transport features. Later it can be discussed how different transport features may yield better QoE. The transport box in figure 6.1 is therefore bypassed in our

testing.

Network logging in a PC with Wireshark
The port mirroring switches mirrors, or duplicates, the media stream. One media stream can therefore be directed to a PC with a network analysis program, e.g. Wireshark, where packet headers are logged. As this is done at both the sender and receiver, it is possible to do post-analysis by comparing the packet headers measuring QoS metrics like jitter, delay and packet loss. This measurement can later explain subjective results.

Using this framework, video can be sent and received exactly as in a dedicated IP contribution network using professional contribution equipment. As QoS metric measurements and QoE measurements are done simultaneously, it is possible to describe system behavior, network behavior and end users perception in every test.

6.2 Proof of concept (POC) test

The POC test was designed to outline the level of quality resulting from contribution over the Internet. It also gave us the opportunity to experiment with the technical setup and different subjective quality assessments.

Using an Android application, developed by us for this specific purpose, the participants logged visible packet loss and rated intervals of 15 minutes. Two movies were shown, and the test lasted for nearly 7 hours in total.

6.2.1 Technical setup

In the following tables we sum up the specific details of our technical setup. This is done in relation to the general technical test framework:

Media capture and playback		
Video source	PC with the video material in file format.	The DVI converter was mounted on as a 720p60 display.
Video material	• The Lord of the Rings: The Fellowship of the Ring (3h 1m) • The Lord of the Rings: The Two Towers (3h 50m)	Ripped from BluRay-Disc. MPEG-4 AVC encoded at 32 Mbit/s.
Interface converters	• Gefen DVI to HD-SDI Scaler box • Black Magic Mini Converter SDI to HDMI	
Display	Samsung plasma TV	50” display, able to show 720p60.

Video to IP gateway		
Sending gateway	TVG450	Set to send at a constant rate of 70 Mbit/s. Encoding with JPEG2000.
Receiving gateway	TVG430	Compatible with TVG450.

Transport protocol and QoS monitoring		
Switch	Two DLINK 16-Port Layer2 EasySmart Switch	Abel to do port mirroring.
Network analyzer program	Wireshark	The post analysis was done with a program we made ad hoc.

JPEG2000 coded HD video was sent over the Internet at a constant sending rate of 70 Mbit/s. It is important to notice that with the settings and equipment we used, the video was of very high quality as long as there was no packet loss. That is, there were no visible degradations due to encoding, compression or other parts of the system then the Internet. It was, for the most part, as if the video had been played on a BluRay-player directly connected to the display.

70 Mbit/s is categorized intra-only contribution - class D - in the contribution bandwidth requirements defined in section 3.3.1. This also means that it can be used for everything Class A, B and C systems can be used for for example high quality news content.

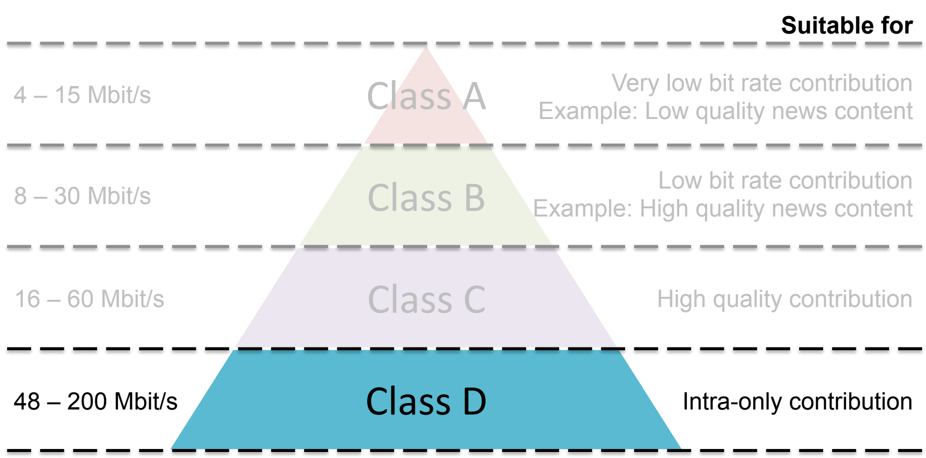


Figure 6.2: 70 Mbit/s contribution is classified as contribution bandwidth class D suitable even for intra-only contribution.

Audio is outside the scope of this thesis, but is still an important part of the watching experience. Therefore, the soundtrack was played back locally. A copy of the video file was played back on a separate computer on the receiving side. The soundtrack from this copy was manually synced up with the video at the beginning of the test. It is worth noticing that in a realistic scenario, the audio would probably be sent over the same channel as the video, and thus experience the same periods of packet loss. The combined effect of video and audio degradations will naturally be more displeasing than just one or the other.

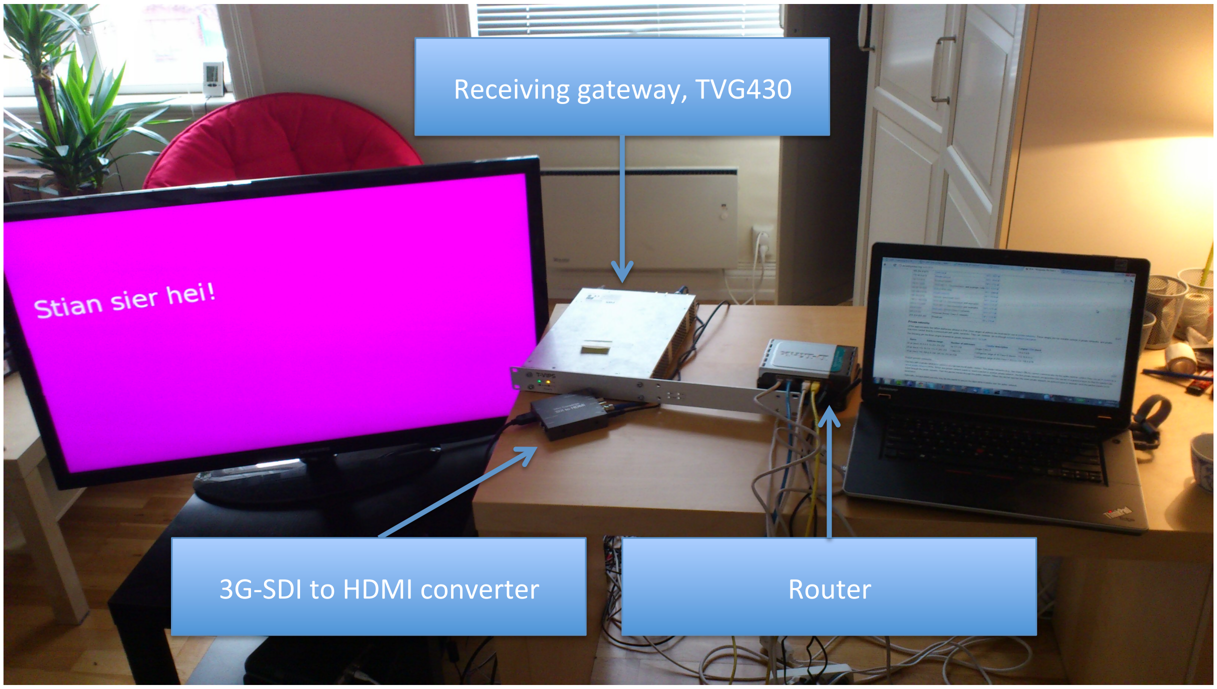


Figure 6.3: Parts of the technical setup on the receiver side. From left, the display, at the moment showing a test signal generated by the sending gateway, the gateway, 3G-SDI to HDMI converter, the router and the computer used for network logging.

6.2.2 Participants

A study with subjective measurement is always highly dependent on the size and composition of the test panel, i.e. the participants. In general, a larger test panel will give more reliable results. Also, the participants should be unbiased and reliable as far as this is possible. A more extensive discussion about composition of a test panel is provided in the method critique in chapter 8.

In the POC test, the test panel consisted of five persons: Two experts[1] and three amateurs. Everybody was students in the range of 24 to 28 years.

6.2.3 Subjective quality logging

To do subjective quality assessment, we developed the application 'VisioRate'[1]. It works on any smartphone running the operating system Android. It lets the user: (1) Register noticeable degradations immediately, (2) register severe degradations intervals such as frame freeze or synchronization problems lasting for seconds and (3) rate the impact of degradations the last 15 or 30 minutes (15 was used in the test) on a scale from 1 - "Very annoying" to 5 - "Imperceptible". For (1) and (2), the user had to actively record the degradations when they appeared, and for (3) an alarm went off every 15 minutes, and they were prompted for a rating.

[1] We served as both experimenters and experts in this test.

[1] Available on Google Play under the name VisioRate

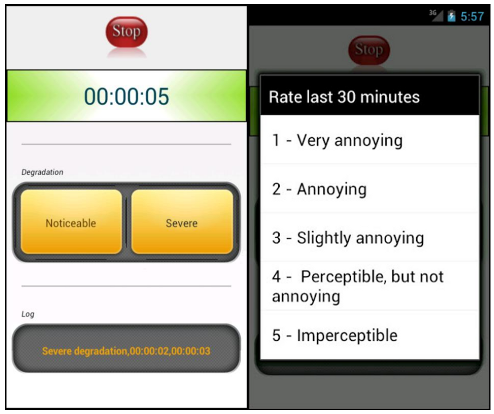


Figure 6.4: Screenshot of the application VisioRate developed and used for subjective testing.

All the participants had Android phones, and installed VisioRate before the test started. Unfortunately, the application had stability issues on some of the phones, so that only three of the five records were complete at the end of the test (two expert record and one amateur record). This was enough to provide meaningful results in the POC test, but we did not risk using VisioRate for the subjective test. Despite this, we definitely think that smart phone applications can be easy and affordable way to do subjective quality testing if work is put into making the application stable and reliable.

6.2.4 Results

This section will present the results from the POC test. First of all, the network data is displayed in table 6.1. To make it easy to compare and discuss in relation to the network tests in chapter 5, we present it in the same format as we did there. For convenience, we show the network data from both the POC test and the subjective test in the same table.

The first thing to notice, is that the results in table 6.1 look very much alike the results from the network tests in table 5.1. We see that, just like in the network test, the QoS parameters conform to contribution network profile A for packet loss ratio, IPDV, reordering, single- and longer packet loss bursts, and that short packet loss bursts conform to class B. The PLBR is split between class B and C, exactly as in the network tests. We have thus measured the same network behavior with two different methods. This is in itself a strong evidence that the methods we used to measure network behavior have high reliability.

Test setup			Resulting QoS metrics			Packet loss distribution			
Test	*Duration*	*Packets sent*	*Packets lost*	*IPDV*	*Reordering*	*Singles*	*Packet loss burst length*[a]		*PLBR*[b]
							< 200ms	200ms <	
POC1	3h 01m	83186470	198 (0.0002%)	15.86 ms	2 (0%)	1	4	0	0.0003 s
POC2	3h 50m	106294500	340 (0.0003%)	21.7 ms	3 (0%)	1	5	0	0.0002 s
Monday	2h 01m	56390589	386 (0.0006%)	21.02 ms	1 (0%)	5	36	0	0.0050 s
Tuesday	2h 01m	56390589	414 (0.0007%)	22.68 ms	3 (0%)	14	23	0	0.0032 s
Wednesday	2h 01m	56390589	407 (0.0007%)	12.28 ms	3 (0%)	2	8	0	0.0011 s
Thursday	2h 01m	56390589	362 (0.0007%)	24.46 ms	2 (0%)	1	7	0	0.0009 s

Table 6.1: Aggregated QoS results from the POC and subjective test.

Cells are colored green, yellow and red depending on conformance to contribution class A, B and C, respectively

[a] The length of one consecutive run of packet loss. Obviously, the number of packets lost in one run depends on the packet sending rate. Therefore, time is used as a common metric as it is independent of packet sending rate

[b] Packet loss burst rate: The time between two consecutive packet loss burst. Here, the minimum PLBR is given for each test

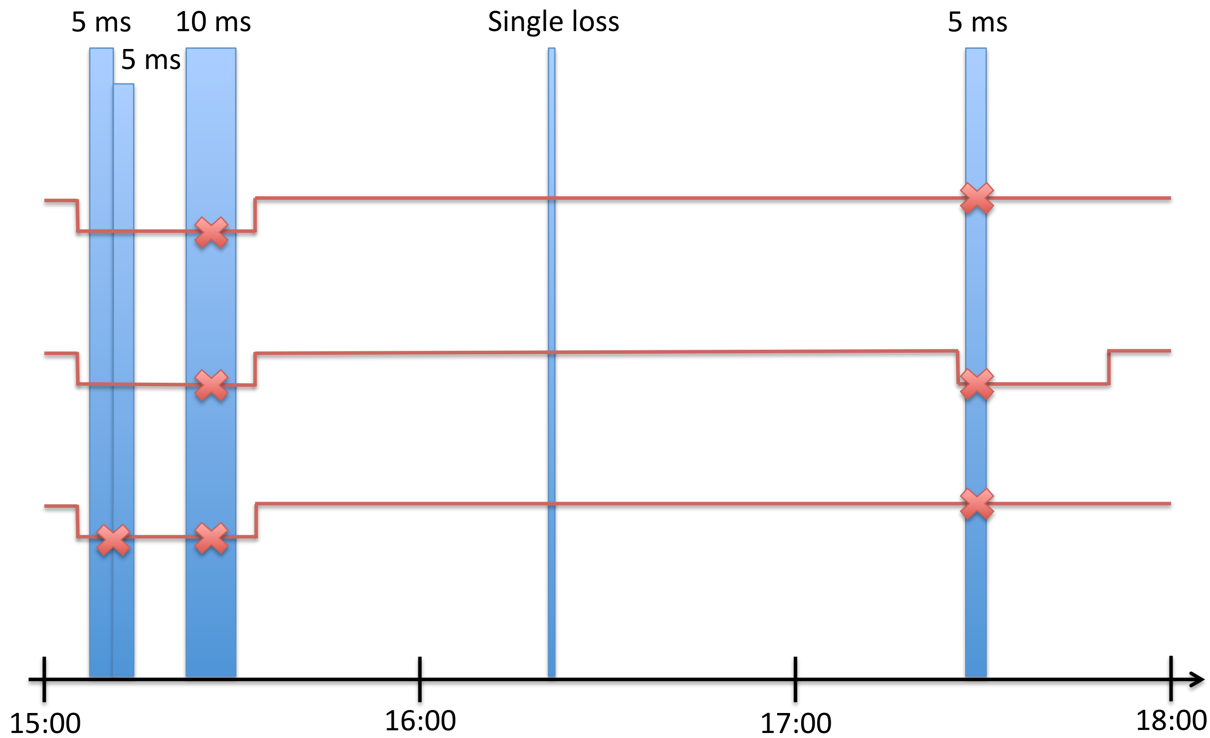


Figure 6.5: Subjective results and network results from "Fellowship of the Ring". The blue bars shows when packet loss bursts and a single loss occurred, and their width is proportional to the length of the burst. The red line shows the interval rating, only "5 - Imperceptible" and "4 - Perceptible, but not annoying" was recorded. The red crosses shows when each participant registered noticeable degradation. No "severe" degradations were recorded. The top two are the experts.

Figure 6.5 and 6.6 combines the network and subjective results. Packet loss bursts are shown as blue bars, and the noticeable degradations are represented by a red cross. The first thing to notice is that noticeable degradations appear relatively seldom, about once an hour. They also last for only a few milliseconds, and are therefore only effect the video in fractions of a second at a time. This is in itself an indication that the overall quality and flow of the video was very good. The noticeable degradations can best be characterized as *frame freeze* visible for a very short time.

The low rate of noticeable degradations also shows why it was important that the test lasted for hours. If the test lasted for only seconds or minutes then probably no noticeable degradations would appear and the effect of packet loss would have not been captured. Interestingly, in many of the scenarios most likely to benefit from contribution over the Internet, the videoclips only last for seconds or minutes. Thus, our results show that a news reports, interviews or other content lasting for only a short time will very likely not experience degradations.

In figure 6.5 we can see that two packet loss of 5 ms appeared at the start of the film. Only one person registered this as a visible packet loss, and the losses happened in such a short time interval that it is not possible to say which of them were visible, if not both. In the same rating interval another packet loss burst of 10 ms occurred. This is noticeable to all three participants, and they

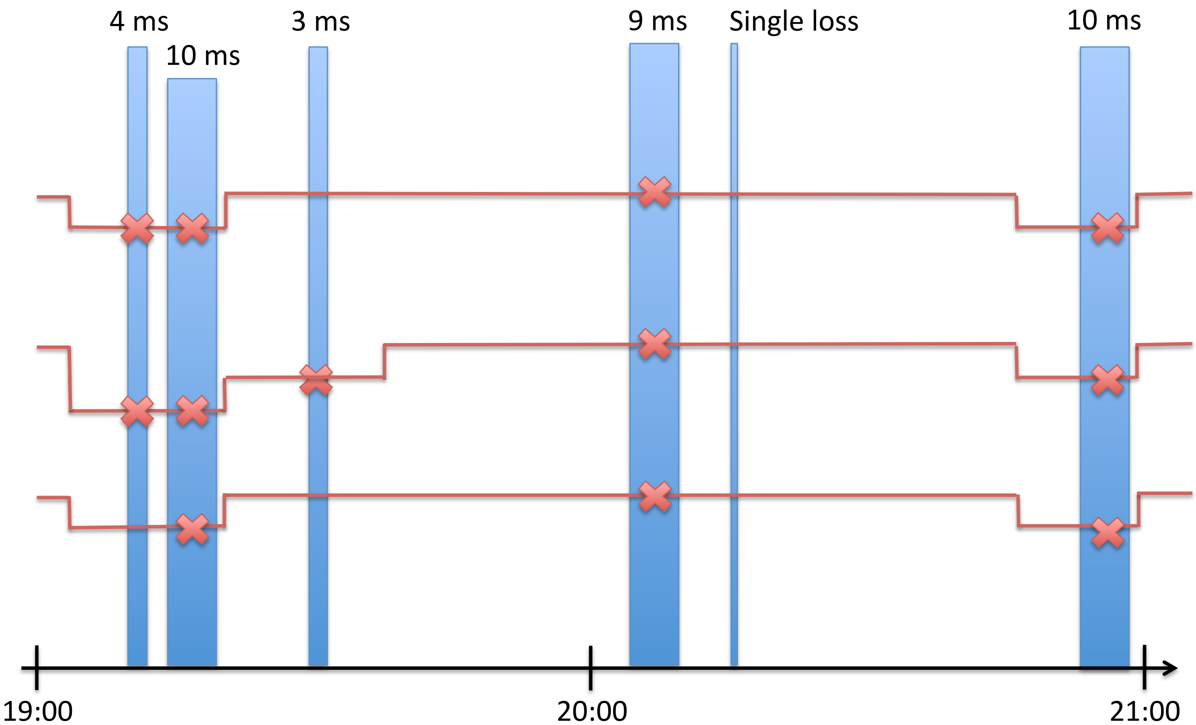


Figure 6.6: Same setup as in figure 6.5, but for "Two Towers". This does not show the periods between 18:00 and 19:00, and 21:00 and 22:00, as no packet loss occurred and no degradations were recorded in these periods. It can be seen that one participant also recorded an interval to be "3 - Slightly annoying".

rated the interval "4 - Perceptible, but not annoying". Further, a single loss, not visible to anybody, and a 5 ms long burst, visible to all three, occurred. Two of the participants chose not to degrade the interval where the visible 5 ms burst happened. This suggests that the degradation was of little significance.

The trend from figure 6.5 also continues in 6.6: Packet loss mostly appeared in bursts, and no packet loss burst lasted for more than 10 ms. All packet loss bursts of around 10 ms gave impairments that were noticed by all participants. Still, the impairment was of such little significance that the participants rated these periods "4 - Perceptible, but not annoying". The lowest rating, "3 - Slightly annoying", was given by one participant in one interval were two such noticeable degradations struck.

Furthermore, the figures show that packet loss burst of length 4 ms or less may or may not be visible to the viewer. What exactly happens when a visible packet loss occurs is complicated and might depend on several things, but it seems fair to assume that it is related to the length of the burst. The data from the POC test suggests that the lower limit for perceptible packet loss is somewhere around 4 ms. It is reasonable to assume that this depends on the fact that only intra coding, and not inter coding was used. This will be further discussed in chapter 7.

In summary, the two most important observations made in the POC test is the following:

1. Noticeable degradations appear relatively seldom have limited impact on the subjective quality. Therefore, the overall subjective quality is very good.

2. There seems to be a lower limit around 4 ms for length of packet loss bursts that lead to noticeable degradations.

6.3 Subjective test

We learned a lot from the POC test, not only with respect to the quality, but also test methodology. This newly acquired wisdom was very useful when we designed the subjective test. To secure maximum validity and the integrity of the results, we developed a research protocol in advance of the test.

In many fields of research, subjective quality assessment included, it is common to use research protocols. This is a document that in full describes the background and reasoning behind the testing, the test setup and how the results will be presented. It also describes the impact of the results and how the researchers intend to publish them. A research protocol is built around one single hypothesis.

The research protocol is not to be changed after the testing starts. Therefore, the original reasoning behind the test will always be clearly stated in this document. It makes the researcher think through every aspect of the test in advance as a aid in developing a test of high validity, i.e. a test that actually measures what it is supposed to.

Our research protocol can be found in Appendix H. It describes how we intend to prove or disprove the following hypothesis:

Hypothesis: Contribution over the public Internet gives an equally good end user experience as contribution over a private IP-network.

It is worth noticing that we want to measure the quality of the experience the system gives to the *end user*. By end user we mean a person that is watching, and possibly paying for, broadcasted TV as a product from the contribution, production and distribution phases before reaching the end users's TV. Because we are only interested in what happens in the contribution phase, the other phases are omitted in our test. This is equivalent to a broadcast system with a lossless production and distribution phase, where loss in the contribution phase will propagate.

6.3.1 Test setup

The technical setup was the same as in the POC test[2]. The main difference between the two tests is how the subjective quality assessment was done. Essentially, the test panel was much larger and the subjective evaluation was done in retrospect using questionnaires. Thus, the Android application VisioRate was not used in this test.

The test had 24 participants and one expert[3]. Their background and previous experience is discussed in section 6.3.2. The maximum capacity of the room we used was seven persons, so for practical reasons, the testing was divided into 4 subtests with six new participants in each subtest. We will refer to each subtest with the weekday the test happen, e.g. "Monday test". The movie was the same every day: *American Beauty*, lasting for 2 hours and 1 minute.

[2]With exception of the video material, which in this test was the movie *American Beauty* (2 hours 1 minute). The quality of the video material was as in the POC test, i.e. very good and 720p resolution.

[3]As in the POC test, we, the authors, acted as both experimenters and experts.

The experts task was to record the number of noticeable degradations. Of course, the network conditions varied from day to day, giving various packet loss counts on each day. The POC test showed that the effect of a packet loss burst cannot always be predicted by analyzing QoS metric measurements - for example, a packet loss burst lasting for 4 ms may or may not be noticeable. Therefore it was useful to note if a packet loss was noticed by the experts. This way, we were able to see the results from the survey in light of the number of degradations that were actually noticeable.

Figure 6.7: The test setup with participants watching the video on the screen.

6.3.2 Test panel

The 24 participants had a similar background: Everybody was students and acquaintances of the experimenters. Their age ranged from 23 to 28.

Figure 6.8 and 6.9 shows that all of the participants watch TV on a regular basis, and that most of them have a monthly subscription in the price range 50 NOK to 450 NOK. This shows that they had sufficient experience with TV to do a meaningful comparison.

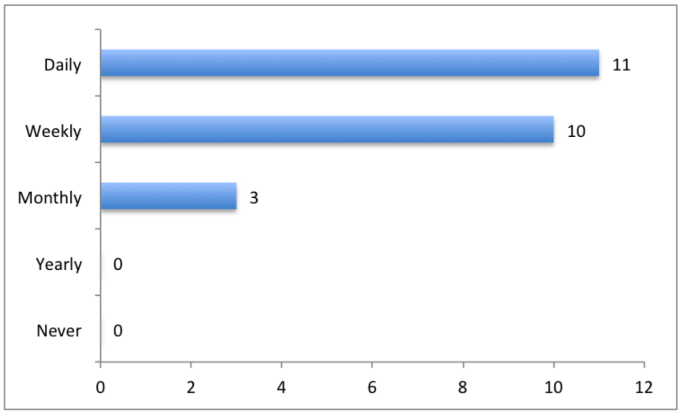


Figure 6.8: **Question 2:** How often do you watch regular TV?

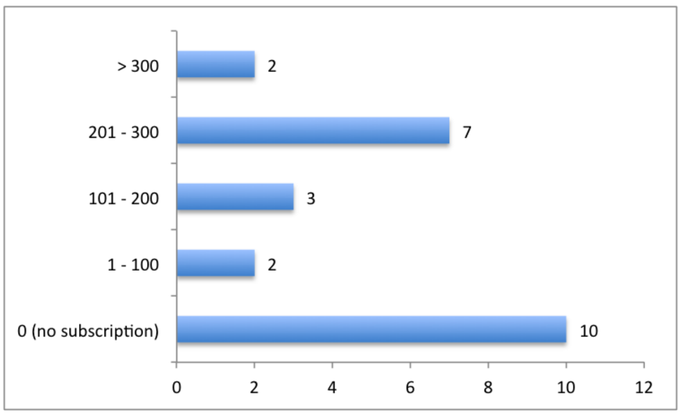


Figure 6.9: **Question 3:** How much is your monthly TV-subscription?

6.3.3 Developing the questionnaire

Three principles were important when developing the questionnaire. First of all, it was important to ask questions related to the video quality. This was done in question 5: "How would you best describe the quality of the video you just saw?"[4], and question 7: "Was the quality better or worse than expected?"[5]. Of course, questions related to what they expected depends on what they are told to expect. To ensure that everybody had the same expectations, we made an instruction sheet that everybody read before the test started. This can be found in the Appendix F. The sheet basically instruct the participants to expect the same quality as they would when watching regular cable-TV. We tried to keep additional information to a minimum, answering only questions to correct basic misunderstandings. This way, we avoided to interfere with the instructions on the sheet.

As a second principle, we included questions about economics because this is a central factor in QoE. It is important for broadcasters to find the balance between user satisfaction and economic aspects like subscription cost [49]. Therefore, it is highly relevant to ask how the participants would react to this level of quality. This was done in question 4: "How much would you be willing to pay to get

[4] rated on a scale from "Bad", "Poor", "Fair", "Good" to "Excellent".

[5] With the alternatives "Much worse", "Slightly worse", "Exactly as expected", "Slightly better" and "Much better"

this quality compared to regular TV quality?"[6] and question 8: "Which action would you take if you experienced this kind of quality on regular TV?"[7]

Third, we deliberately asked questions that not everybody were qualified to answer. For example question 10: "How much better do you think the quality would be if we used HD?" reveal the level of insight a person has in the topic, and sometimes if he is even paying attention to what he answering. In situations when we are able to identify an outlier, the information from these kinds of questions might provide a useful context.

When we present the results from the survey, we will first focus on the most general trends, and thereafter comment on specific outliers. But first of all we present the expert logs and some more detailed network data.

6.3.4 Expert logs and detailed network results

QoS metric measurements from this test is included in table 6.1. It is also useful to give a more detailed breakdown of the packet loss burst lengths, and compare this to the number of noticeable degradations. This is done in table 6.2.

	Monday	**Tuesday**	**Wednesday**	**Thursday**
Single packet loss	5	14	2	1
Bursts of length $<$ 5 ms	36	17	2	2
Bursts of length 5-20 ms	0	6	5	5
Bursts of length 20-200 ms	0	0	1	0
Bursts of length $>$200 ms	0	0	0	0
Noticeable degradations	0	2	2	3

Table 6.2: Expert logs and detailed network data for the subjective test.

From table 6.2 we can make the same two observation as in the POC test:

1. The rate of noticeable degradations is low, a noticeable degradation appears about once every hour.
2. Not every packet loss leads to a noticeable degradation. On Monday, none of the packet loss bursts bellow, all below 5 ms, led to a noticeable degradation.

Thus, the expert and network results in the subjective test are definitely in line with the results and conclusion from the POC test.

It is worth noticing that, in general, the experts were more capable to notice, and payed more attention to, packet loss compared to the participants. Therefore, packet loss noticed by the experts were not necessarily noticed by the participants. This is explained more in detail later.

[6] The scale ranging from 0% through 25%, 50% and 75% to 100%.

[7] Alternatives: "Cancel your subscription (without any economical consequences)", "Call the TV provider to complaint", "Turn off the TV or switch channel", "Point out the poor quality to the ones you are watching TV with" and "Nothing, the impairments does not bother me".

6.3.5 Survey results

In section 6.3.3 we outlined the three principles that were the basis of the questionnaire: Video quality, economics and others. In this section we will present the most important results from the video quality and economics related questions. The other questions will be useful when we discuss the validity and outliers of our results in section 6.3.6.

Video quality

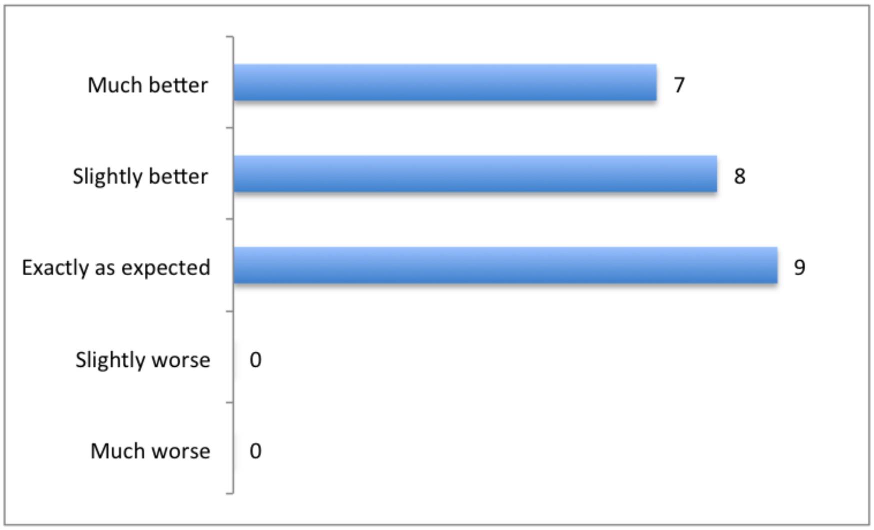
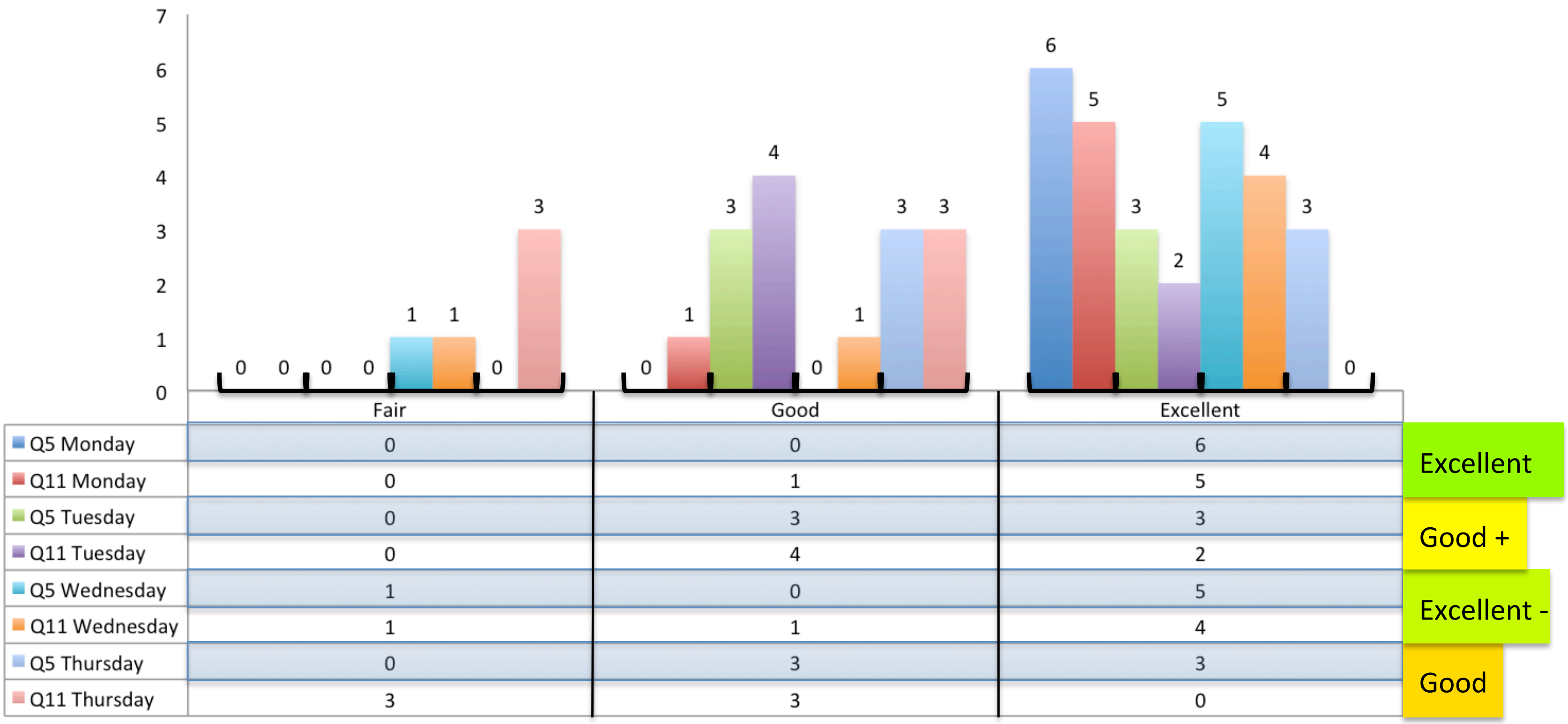


Figure 6.10: **Question 7:** Was the video quality better or worse than you had expected?

As seen in figure 6.10, all the participants rated the video quality *exactly as-*, *slightly better-* or *much better* than expected. *This is a clear indication that the overall quality was very good.* This trend is also shown in the more comprehensive figure 6.11. Although the participants were instructed to expect TV quality, it is not correct to state that the quality was indeed better than regular TV even though some participants answered this. Some participants maybe expected quality lower than TV quality because this was a research setting involving the internet. Maybe some participants expected the internet to have lower performance.

Figure 6.11 shows the results from question 5 and question 11. Although both questions concern video quality, they are different in an important way: Question 5 prompts for the quality of the whole video, i.e. the overall quality, whereas question 11 prompts for quality rating when packet loss was actually visible. And even though question 11 is bound to rate worse than question 5, the combination actually yields a clear result: The overall quality of the video is very good, even in periods with noticeable degradations.

When looking at figure 6.11 in relation to the expert and network results, we see a clear correlation between subjective feedback and noticeable packet loss. On Monday, no noticeable degradations were recorded by the experts. Figure 6.11 shows that only one person on Monday rated the video anything less than excellent, he rated it *good when packet loss occurred*. On Tuesday and Wednesday, when



	Fair	Good	Excellent
Q5 Monday	0	0	6
Q11 Monday	0	1	5
Q5 Tuesday	0	3	3
Q11 Tuesday	0	4	2
Q5 Wednesday	1	0	5
Q11 Wednesday	1	1	4
Q5 Thursday	0	3	3
Q11 Thursday	3	3	0

Figure 6.11: **Question 5:** How would you best describe the quality of the video you just saw? **Question 11:** How would you describe the quality of the video when packet loss occurred? The alternatives "Bad" and "Poor" are not shown in the chart since nobody choose these.

2 noticeable degradations were recorded by the experts, the survey results are more spread but still the overall quality was *good to excellent*. Thursday, the day with the most noticeable degradations (3), was in overall rated *good* which was the worst result among the tests. Here, three persons rated the quality as low as *fair* in degraded periods. Hence, there is a clear correlation between noticeable degradations and the survey results which was expected. This shows that our subjective test using questionnaires was successful in terms of validity.

Despite this, we will still say that the the results from the video quality related questions points in the direction of very high QoE.

Economics

The second design principle we followed when we developed the questionnaire related to economics. Figure 6.12 and figure 6.13 shows two questions in this category.

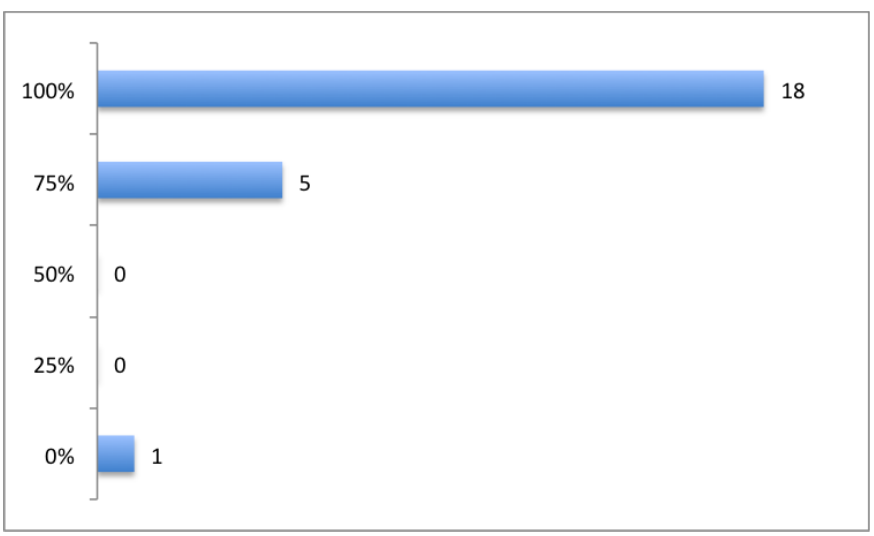


Figure 6.12: **Question 4:** How much would you be willing to pay to get this quality compared to regular TV quality?

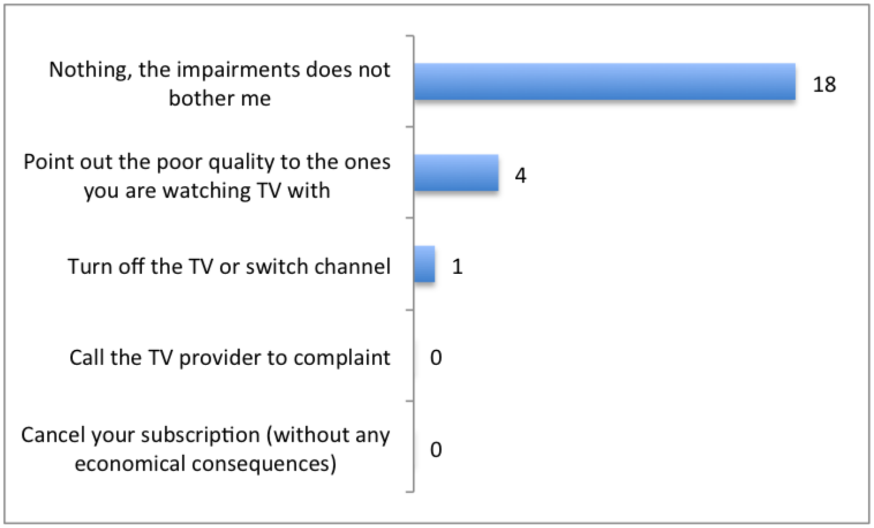


Figure 6.13: **Question 8:** Which action would you take if you experienced this kind of quality on regular TV?

We can clearly see the same trend here as we saw in the video quality results: Contribution over the Internet scores in the high end of the QoE scale. But, although the majority of the answers unambiguously points in the direction of the video being of very high quality, a couple of outliers can be seen. These will be discussed in the next section.

6.3.6 Survey results: Outliers

We ended the previous section by pointing out the fact that a couple of the answers in question 4 and question 8 were deviating from the large majority. Participants giving deviating answers are called outliers. Specifically, in question 4, one person reported that he would not pay more than 0% for this quality compared to regular TV. This person was one of the many who reported to have no TV-subscription at the moment, and we suspect that his reasoning was that 100% of 0 is 0, and therefore, it did not matter what he answered on this question.

The second outlier was found in question 8. One person from the Monday test answered that he would "Turn off the TV or switch channel" if he experienced the same quality while watching regular TV. We already know from the network and expert data that the quality was very good on Monday, no noticeable degradations were recorded. In fact, this person said the quality was "Much better" than expected in question 7, and "Excellent" in question 5 and 11. We therefore find it strange and improbable that he would actually go as far as turning off the TV.

Also, a less obvious outlier was found in figure 6.11. On Wednesday, the same person answers "Fair" on both questions (question 5 and question 11). He is also the outlier in question 10 shown in figure 6.14. Since this person is consistently providing answers that deviates from the general opinion, we have not put to much weight on these in our analysis and conclusions.

In question 1, we asked "Did you notice any degradations in the video?" with the alternatives "Yes"

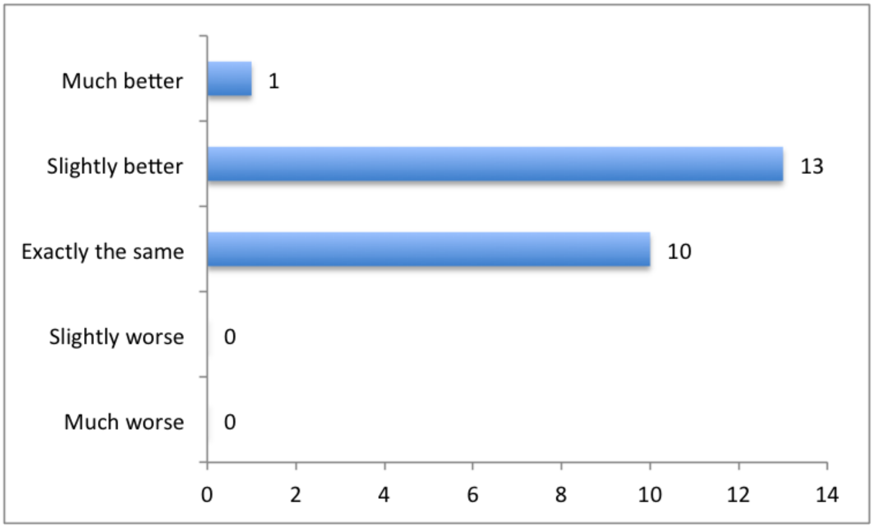


Figure 6.14: **Question 10:** How much better do you think the quality would be if we used HD?

and "No". Because the results were somewhat mysterious we have waited until the outlier section to present it. Figure 6.15 shows how these were distributed on the different days.

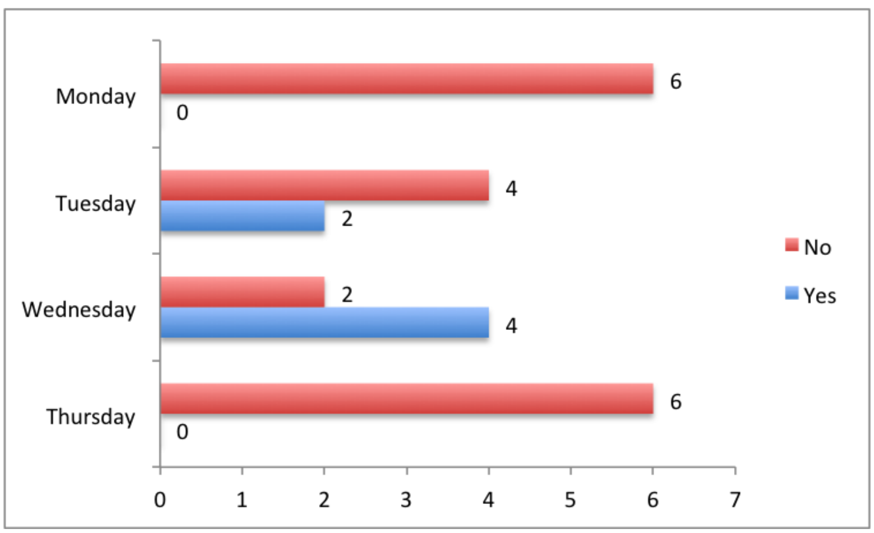


Figure 6.15: **Question 1:** Did you notice any degradations in the video?

Only 6 of the 24 responded that they noticed degradations in the video. We very strongly suspect that more than six perceived what we call "noticeable degradations". Possibly the term "degradation" caused confusion. This is due to the fact that the Thursday test gave invalid results where the participants answered that they did not notice degradations, yet the overall score this day was rated "Good" as illustrated in figure 6.11.

In chapter 8 we discuss different things that influence the validity of a survey. There, question 1 is mentioned and discussed particularly as an example of a question with low validity - a question that is misunderstood or does not measure what it is supposed to. In addition to the questions and results above, we asked questions related to watching Tippeliga football matches, which gave ambiguous results because of varying interest in football among the participants. We also asked some questions related to the test in general. The full set of results can be seen in Appendix G.

6.3.7 Summary

With the outliers identified and discussed, we can look on the results from the subjective test as a whole. We have already discussed clear trend in the survey results, and the correlation between the expert and network results and the survey results. In sum, it all points in one direction: The quality is very good. It actually gives us the necessary basis to say that our hypothesis, that contribution over the Internet gives an equally good end user experience as contribution over a private IP-network, is indeed confirmed.

The test also confirmed our findings from the POC test, thus increasing the reliability of these results. In chapter 7 we will further discuss the POC test and the subjective test in relation to the network tests conducted in chapter 5.

In the introduction, we said that the purpose of this chapter was to provide an answer to the question "*Does contribution over the Internet meet TV viewers expectations in the distribution phase?*". Having already concluded that contribution over Internet gives the same quality as contribution over IP contribution networks, it is reasonable to say that we also have the foundation to claim that "*Yes, contribution over the Internet actually does meet TV viewers expectations in the distribution phase*.

Chapter 7

Discussion

With all the results and conclusions from the network tests, POC- and subjective test in place, we will now briefly discuss some of the implications they have when seen collectively.

7.1 Noticeable degradations and error correction strategies

One of the important findings in the POC test and subjective test, was that packet loss bursts shorter than 4 ms did not generally lead to noticeable degradations. Compared to the measured QoS metrics shown in section 5.5, a lot of the packet loss bursts are below this length. For example, 82% is less than 5 ms long for the tests on 70 Mbit/s (Test 1) as illustrated in figure 7.1.

The figure also shows the error correction strategies we discussed in section 5.5[1]. FEC is designed to correct short packet loss bursts and single packet loss below 4 ms, and therefore FEC is not able to correct for noticeable degradations as a result from packet loss bursts above 4 ms.

It is important to emphasize that the lower limit for noticeable degradations was outline by use of intra-only coding. With inter coding, the error can propagate to other frames, and the result would probably be worse.

The two other strategies discussed in section 5.5, retransmission and delay backup stream, are not limited to 4 ms as packet loss bursts up to 200 ms can be corrected. Therefore, these are better suited to prevent noticeable degradations. Those who consider implementing error correction, must weight the possible gain in QoE against the increased complexity, longer delay and lower bandwidth that often follows with an error correction strategies. And, since the QoE is already very high, it might not be worth it in some cases.

[1]The range of FEC is determined in terms of number of packets, e.g. the maximum number of packets it is abel to correct is 20. Therefore, higher bitrate means more packets per millisecond, and therefore fewer milliseconds of FEC correction. For 70 Mbit/s, the max is approximately 4 ms.

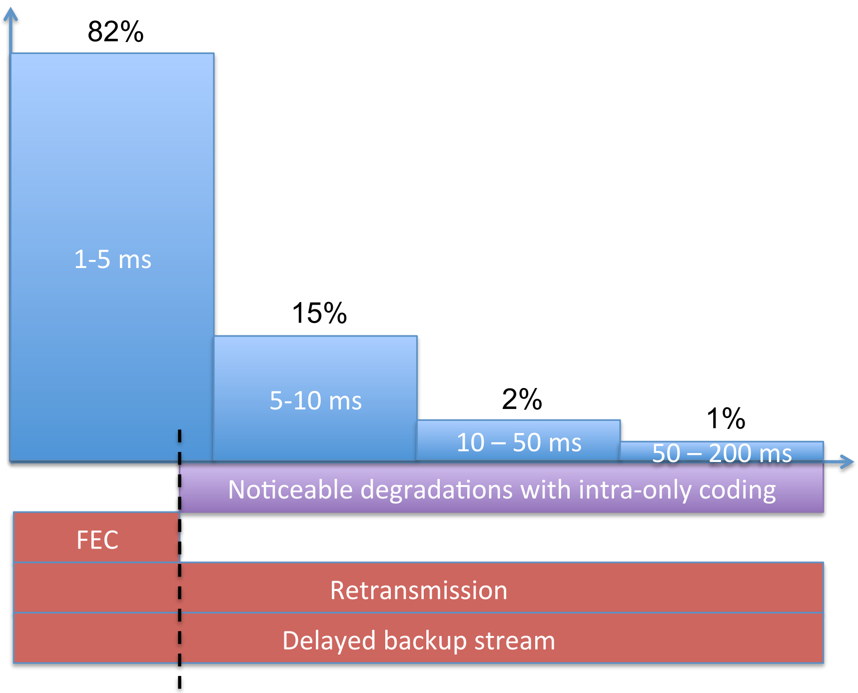


Figure 7.1: Compilation of results from the network tests (blue), POC and subjective test (purple) and some known and suggested error correction strategies (red).

7.2 Limitations of results

The results from chapter 5 and 6 is limited by the connection we chose. That is, the recorded levels of QoS and QoE was a direct result of the networks and routers the packets traversed. So can the results apply for other connections in Norway? This is difficult to definitely say. What we *can* say is that the results applies for connections where the bottleneck[2] is any of the networks or nodes the packet flow in these tests traversed. Namely, Telenor core network, Canal Digital access network and NIX (Oslo). As these network components are serving a vast set of Norwegian Internet users, the results here probably applies elsewhere. Conversely, if a broadcaster use an access network with less performance (bandwidth and reliability) as used here, the same results cannot be expected. As a general recommendation, the broadcaster should chose the best available access network as this can be a bottleneck in the connection.

7.3 Choice of bandwidth

We made a decision on testing the best-effort Internet at high bandwidth, i.e. in the upper scale (70 Mbit/s) of what is offered on the commercial marked (100 Mbit/s[3]). For only a couple of years ago, 70 Mbit/s access link for commercial users was way beyond availability. Hence, our intention was

[2]The network node, or network, with the poorest performance in the connection.

[3]Some ISP offer even more bandwidth. However, 100 Mbit/s is a common limit [41].

that proving successful contribution on such a high bandwidth would be an eye-opener statement.

For occasional contribution over the Internet, probably a better alternative is a MPEG-4 AVC based system, since MPEG-4 AVC is ideal for low bandwidth. This was also our initial standpoint; Why use intra coding at 70 Mbit/s bandwidth when most scenarios is probably best served by a MPEG-4 AVC system which is less bandwidth consuming and provides sufficient video quality? But, the results and our experience changed our standpoint. The choice of high bandwidth and intra coding went from a eye-opener statement to the core success factor in our subjective test results, based on the facts mentioned above. We cannot say anything about the performance of a MPEG-4 AVC system - as we have not tested this - but what we can say is that *given a sufficiently high level of bandwidth on the contribution location, we see only advantages using high bandwidth and intra coding*.

7.4 Guaranteed performance with QoS in Tier-2 networks

In our tests, we showed that the level of reliability over the best-effort network is sufficient. However, best-effort implies that reliability cannot be guaranteed. For guaranteed reliability, the solution is to buy QoS from a Tier-2 network as mentioned in chapter 4. When discussing whether contribution is possible in the Internet with Telenor network core engineers and technical engineers in NRK, all parties agreed on that contribution is indeed possible and should be done by buying QoS from a Tier-2 network.

In addition to guaranteed reliability, this solution has the benefit of protection agains security attacks and link failures. Security attack, such as a Denial-of-Service (DOS) attack, on a network router would cause immediate overflow and failure. Some broadcasters cannot take this risk. Buying a VPN solution or a dedicated link from a Tier-2 network, one is guaranteed that the contribution traffic is handles before any DOS attack - providing protection. When link failure occurs, chapter 4 show, as an example, that the Telenor Tier-2 core network is sufficiently redundant such that QoS-marked traffic can be rerouted over other links. As a result, the best-effort traffic is most affected, while QoS-marked traffic has priority in the network.

Chapter 8

Method critique

8.1 The Internet - the worlds largest IP network

This section was written with the intent to introduce recent Internet performance trends. A Internet performance description, however, is impossible to keep it general as the Internet is a collection of networks with different performance. General performance statements about the Internet will quite possibly be both correct and wrong - depending on the eye of the beholder. For example, over-provisioning Tier-2 networks is common in Norway - but this might not be the case in another country.

Therefore, we chose to keep the Internet performance discussion specific. First of all because specific information is correct. Also, it is our belief that specific information gives the reader more hands-on knowledge. Therefore, some lack of generality might be present.

Most of the information, applies for Norway in particular. In fact, most of the information was given by Telenor and may therefore not apply for *all* ISPs in Norway. Telenor, however, is the largest and most powerful ISP in Norway. Even though we could have talked to other ISPs, Telenor was a natural first choice for discussion Internet performance.

We also chose to not describe Internet philosophy trends. Debates like net neutrality versus QoS are left out. It is indeed important to discuss the future of the Internet, but in this thesis we chose the hard facts.

8.2 Internet performance measurement

As far as method critique, the most difficult element in this test was to capture general Internet characteristics. This is because the Internet is a collection networks with different performance. *It is not generally possible to describe performance and reliability for all connections in the Internet*. Hence, when measuring Internet performance, test behavior is uncontrolled and often cannot be explained because details and status of routers along the path is unavailable. The alternative would be to describe the Internet with a model, and then do controlled tests with this model. However, modeling the Internet is difficult and therefore the results may be unrealistic. Therefore, we chose realistic measurements over Internet.

Another difficulty is choosing the connection, i.e. the placement of the client and server. Possibly, a vast set of different location in Norway should be tested. Due to practical reasons, this is difficult to accomplish as we only have access to a handful of different access points. [32] When discussing this matter with Telenor network core architects, we were advised to only test one connection extensively to capture life-cycle behavior. Another host location *with the same Tier-2 ISP* should not, in principle, yield different results. Also, even though the access points were the same, the traffic patterns in the peering point, NiX1 in Oslo, will vary. Thus, the NIX element add the necessary variance to the testing without actually varying the host placement.

A third difficulty is the quality of the data. When degradations were found, e.g. jitter or loss, we had no knowledge about the component that caused it. We cannot rule out that the application in some cases was the degrading factor, or bottleneck. As mentioned, the software Wireshark was responsible for logging every packet header arriving at the network interface. At 70 Mbit/s, roughly 5800 packets arrive every second. With a small drop in computer performance, e.g. process blocking by the processor scheduler, some packets may have gone unlogged. Possibly, a real-time operating system[1] would yield more stable performance. To check if this was a significant problem, we tested the system for an hour at rates around 100 Mbit/s in our lab. We did not observe any failures in that test, and therefore, it is our belief that application failure did not significantly contaminate our results.

[1] In a real-time operating system, prioritized processes will *always* run as scheduled.

8.3 Testing video contribution in the Internet

8.3.1 Subjective Quality assessment

Most subjective video quality assessment methods evaluates the video for a given setting, for example a given bitrate or coding technique. The parameters are predetermined, and the stimuli period is normally around 10 seconds for methods with retrospective rating and up to 30 minutes for methods with continuos rating [26]. In our case, all the parameters could not be predetermined, given the nondeterministic behavior of the network, and the stimulus period had to be long term, in the order of hours. Thus, we turned to the wider concept Quality of Experience.

The problem with QoE, is that it still lacks an undisputed definition and standardized methods. In the advent of standardized methods to subjectively or objectively assess QoE, we choose tailor a method to meet our needs. There is a clear drawback with choosing a custom made method over a standardized: The strengths and weaknesses of a standardized method is much studied and well understood, a property we loose when we choose to go our own way.

In both the POC test and the subjective test, it was necessary for the experimenters to also take the role as the expert. This will generally weaken the integrity of the results, as it is reasonable to assume that the experimenters have presumptions about the results. This will be of less concern when the experts are to provide quantitative data like number of noticeable degradations than qualitative data like ratings.

In the subjective test, we use *regular cable TV* as a reference. The participants are asked to imagine the experience they usually get from watching TV, and compare this to what they experienced in the test. Ideally, all the factors that influence QoE, listed in section 3.4, except for the network related system System IFs should be the same in the reference and test scenario. For example, the participants should be used to watching TV in HD resolution, since this is what we are using in the test. We had no way of ensuring that they had the right background, but tried to map their experience through some of the questions.

8.3.2 Survey methology

From the definition of QoE presented in section 3.4, it is obvious that we need some way to measure "the degree of delight or annoyance of the user". This is a common case in social studies like psychology, and the preferred method is often surveys. The *reliability* of a survey say something about the generality of the results; will repeated measurements yield the same results? This is not the same as the *validity* of a survey; does it actually measure what it is supposed to? [50]

In a survey, a *sample*, a set of persons, chosen from a *population* answers a *questionnaire*. The sample should be a representative selection of the population. This is done by randomly choosing a sample of large enough size from the population one wish to say something general about [51, p. 280]. For practical reasons, we were not able to get a representative sample of Norwegians, which is the *population* in our case. All of the participants were students, and may for example be less willing to pay for a service than a person that has a higher income. The age of the participants only spanned from 23 to 28, which might also be an influencing factor. Also, a sample size of 24, as in our experiment, is not large enough to get very high reliability. If one person has an unrepresentative view or way of answering the questionnaire, this will be less obvious and more influential when the sample size is small. Sometimes we are able to track more than one *outlier* to the same participant.

The reactive effects of the experimental setup is also a source of error [51]. The participants are influenced by their perception of what results the experimenters wish or expect to get. In our case, all of the participants were acquaintances of us, and some of them had heard about our project and what we were trying to prove. Thus, it is likely that they were somewhat biased because of this.

Another thing that influence the validity is the language used in the question and alternatives. Some questions might appear leading and words may be misunderstood or have different meaning to different persons. For example, in question 1 we ask: Did you notice any degradations in the video? The alternatives are "Yes" and "No". In section 6.3.6 we cast doubt about the validity of the answers we got. A possible explanation can be that the participants did not consider visible packet loss to be a degradation, they might not fully understand what "degradation" means. Another explanation can be that they interpreted the question in a qualitative-, not quantitative, way. That is, "How many times did you notice packet loss? Zero times? One time or more?" might have been interpreted as "Was the video good or bad?". In social science, analysis of question semantics is an own discipline that is yet to be fully understood [50].

A strategy for getting more contextual information to enlighten the key results with was to also ask questions that only some of the participants were competent enough to answer, questions that only gave meaning to the ones noticing degradations and questions that required the answerer to be concentrated and think it through. Unfortunately, this led some of the participants to not provide an answer on some of the questions, although instructed otherwise.

Chapter 9

Conclusion

The purpose of this thesis was to answer the question: "Can contribution be realized in the Internet at a reliability and bandwidth level conforming to QoS and QoE contribution requirements? ". This was further broken down into three more specific questions, presented in the sections bellow:

What is the state of Internet performance in terms of bandwidth and reliability?
In the last decades, Internet has evolved from a small research network designed for document transfer, to a world spanning network with a wide range of applications. Internet is now a vital communication and entertainment platform for end users. The key to Internet success is the flexibility of the IP protocol, able to carry any user service. Moreover, TCP has been successful in creating a scalable Internet. Finally, Internet survival and expansion is purely due to development of successful business models: Customers demand, and are willing to pay for, more bandwidth consuming services - which in turn demands, and finance, bandwidth expansion and higher reliability by the ISPs.

Bandwidth consuming Internet video traffic (to both PC and TV) is now the dominating traffic type in the Internet accounting for more than 50% of all Internet traffic. Access link bandwidth keeps pace with this rapid development; in Norway, access link bandwidth annually grew with 40% from 2004 to 2011. The average first-mile bandwidth is now 12 Mbit/s and expected to be 40 Mbit/s by 2015. In some areas, ISPs offer 100 Mbit/s access link bandwidth at a reasonable price. As far as contribution bandwidth requirements are concerned, commercial ISPs now offer bandwidth well within contribution bandwidth class D. Bandwidth is therefore not a limiting factor for contribution over the Internet.

The best-effort principle is still dominating in the Internet. Guaranteeing QoS across peering points is generally not a practice. Reliability is ensured by proper network provisioning and smart traffic distribution. Broadcasters can expect contribution over the best-effort Internet traversing only a limited set of networks and peering points to be delivered with an adequate level of QoS.

Tier-2 ISPs, like Telenor, are focused on designing core networks with a high level of performance. This allows more services to be implemented, which ultimately means more profit. Telenor core network, BRUT 2.0, is a specific example. This network is designed to never be more than 70% loaded, avoiding network impairments such as packet loss and jitter. Real-time services can be delivered with QoS guarantees inside Tier-2 network, however, some services are seamlessly delivered as best-effort traffic because of proper network planning and design. Contribution as best-effort traffic within this Tier-2 network can therefore expect reliability and performance.

QoS guarantees can be bought within a Tier-2 network. This is arguably the best solution for contribution over the Internet, as QoS is guaranteed. Broadcasters can buy VPN solutions where the traffic is QoS marked, and also dedicated fiber lines where capacity is devoted to the broadcaster. Not only guaranteeing QoS, this solution protects against DoS attacks, network link failure (the QoS traffic has first priority also in the case of rerouting) and congestion at access nodes.

Can a contribution packet stream conform to contribution requirements?
Extensive network testing showed that the Internet conforms to professional IP contribution network requirements in measures of all QoS metrics except the rate of packet loss bursts (PLBR). We used ITU-T network requirements, where the three profiles A, B and C are defined. IP contribution network must conform to profile A, and best-effort networks must conform to profile C. In measures of packet loss, jitter and reordering, our testing showed that Internet conformed to profile A, but in measures of packet loss burst rate, the Internet conformed to profile B and C.

Analysis of packet loss burst patterns showed that no bursts lasted for more than 200 ms - and no two bursts were separated by less than 2 seconds. This allows effective error control strategies to be implemented with low tradeoff in delay and bandwidth. A retransmission strategy or a parallel delayed backup stream can solve the problem with high PLBR given a delay budget in the range of 200-250 ms and 10-30% bandwidth overhead. For broadcasters tolerating this tradeoff, packet loss bursts in the contribution packet stream can be corrected or masked.

Does contribution over the Internet meet TV viewers expectations?
Our tests show that contribution over the Internet does indeed meet TV viewers expectations. This is an important result because, in the end, the broadcasters main concern in terms of quality, is that their customers expectations are met. We showed a movie, conveyed over the Internet using professional IP gateways, to a test panel of 24 participants. After the movie, the participants filled out a questionnaire. By analyzing the answers they provided, we were able to measure the provided QoE and compare it to expected QoE for regular cable TV.

Furthermore, we were able to outline a lower limit to the length of visible packet loss bursts for intra coded video. Packet loss bursts shorter than 4 ms were generally not noticed by the participants in our tests. This has important implications on how packet loss is handled: If short bursts have no effect on the QoE, techniques to correct them might be redundant. This favors retransmission and parallel delayed backup stream over Forward Error Correction (FEC).

The fact that visible packet loss bursts occurred relatively seldom, on average once an hour, also indicates that the overall quality was very good. For short broadcasts, visible packet loss will actually be unlikely to occur at all. For example, an on-site news coverage lasting a few minutes will very probably be perfectly conveyed over the Internet.

So, to the main question: **Can contribution be realized in the Internet at a reliability and bandwidth level conforming to QoS and QoE contribution requirements?**
Yes, the Internet is definitely able to provide both the bandwidth and reliability required for contribution. It performs as well as IP contribution networks in terms of network QoS parameters and end users QoE. It is therefore safe to say that broadcasters can use Internet for contribution whenever this is a favorable option, without having to worry about degraded quality.

Appendix A

Pictures of Contribution equipment

This appendix presents pictures of various contribution equipment. Some of the pictures are taken by the authors, and some are found in the world wide web.

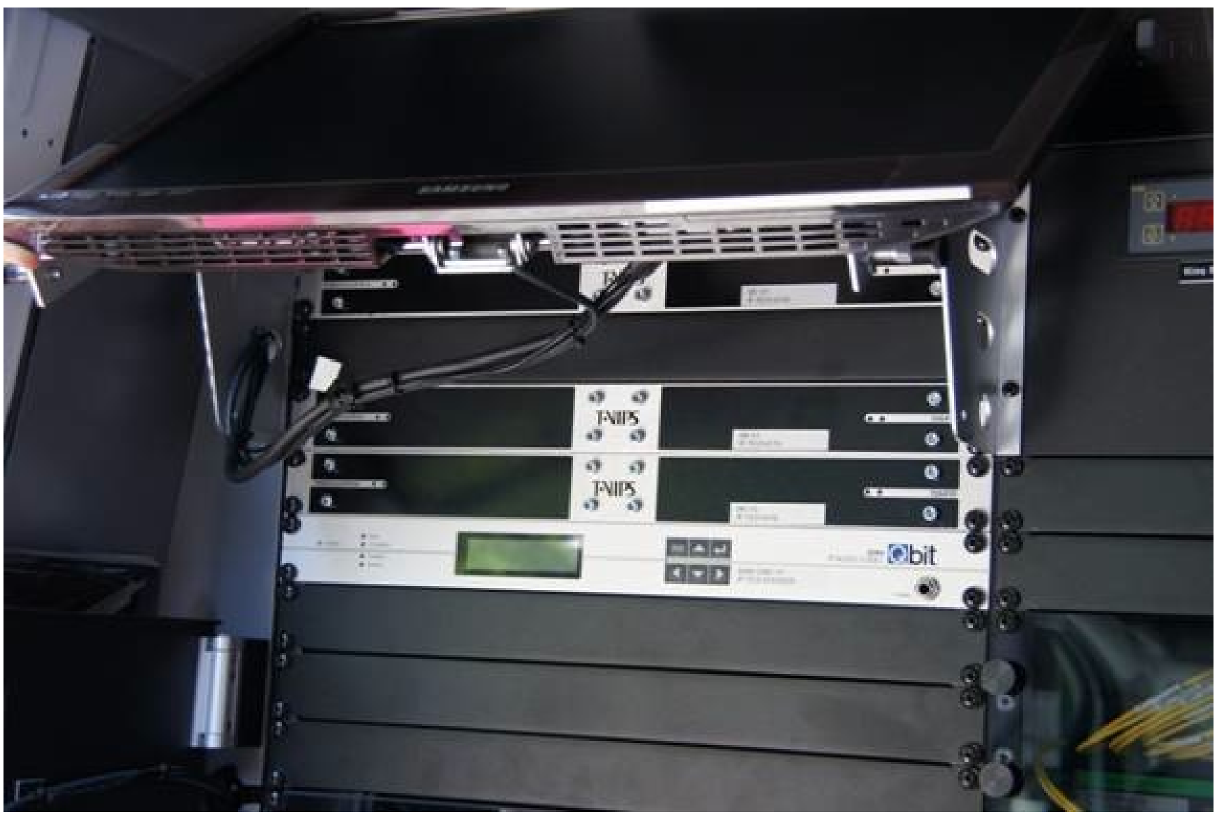


Figure A.1: Photo of two IP gateways using JPEG2000 for compression. An IP gateway produces IP packets from an raw audiovisuall input stream. One gateway is used for transmission, and one for reception. Photo taken from http://pipelinecomm.wordpress.com/2011/09/01/t-vips-provides-jpeg2000-video-transport-solution-for-a1-telekom-austria%E2%80%99s-roving-ob-units

Figure A.2: Photo showing a SNG-truck to the left, and an OB-satellite bus on the right. The OB bus is combined production centre and satellite uplink station. Photo found at wikipedia.org

Figure A.3: Photo of a BGAN device. The white "plate" is the satellite transmitter/receiver. Inside the suitcase, there is a laptop which can be connected to the BGAN device. This equipment is easily carried by one person. Photo taken at NRK

Figure A.4: The microwave solution STRATA used for DENG. The spout is directed towards a receiving antenna up to 70 kilometers away. Photo taken at NRK

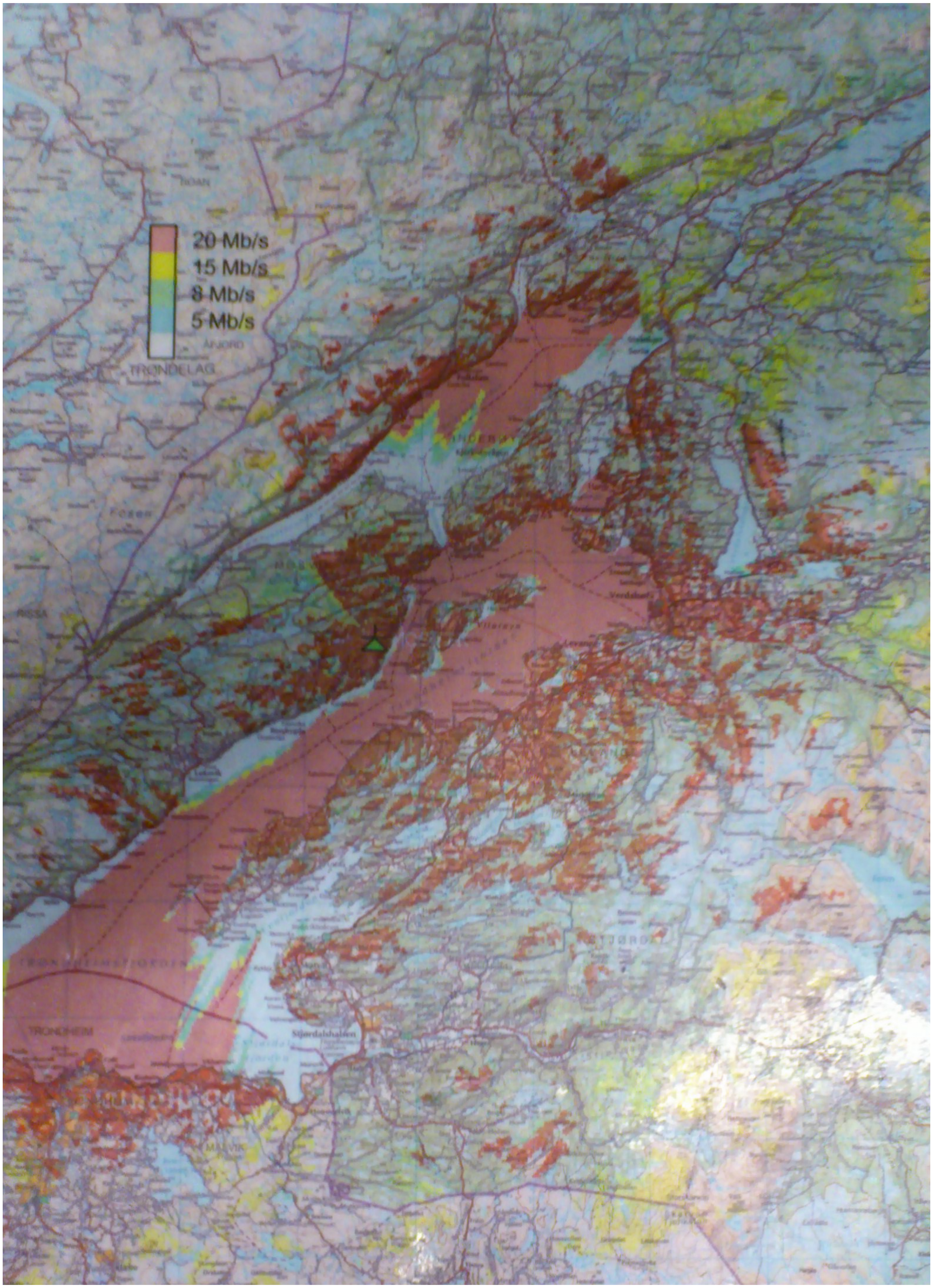


Figure A.5: A coverage map of Trondheim, showing where the STRATA can be used, and at what maximal bitrate. Photo taken at NRK

Figure A.6: Example of contribution over 3G network. Here, a simple app on the handheld device is used for both video capturing and contribution back to a nearby news studio. Photo found at www.streambox.com/live

Appendix B

Internet performance measurements test results

This appendix display test result from all Internet performance test in detail. For each test, statistics are presented in a table and a graph showing packet loss and throughput is displayed.

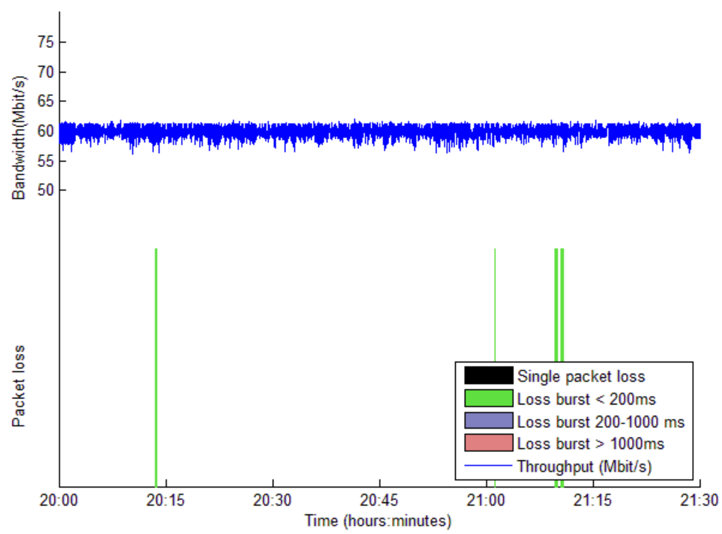


Figure B.1: Graph showing packet loss and throughput

Figure B.1 is presented here as an example. The graph is interpreted as follows. At the bottom in the graph, packet loss bars show when packet loss occurred. The width of each bar illustrates how long the packet loss bursts lasted. However, the width of each bar is not necessarily correct with respect to the time axis. That is, a packet loss burst of one second will have a wider bar than one second on the x-axis. This is to make it easier to interpret for the reader. Note that packet loss bars have different colors (see legend).

The blue graph show throughput at the server in Mbit/s. Incoming data are averaged over 100ms, and each such average is one point on the graph. Hence, the variations are not necessarily packet loss, but variation in arrival time (jitter).

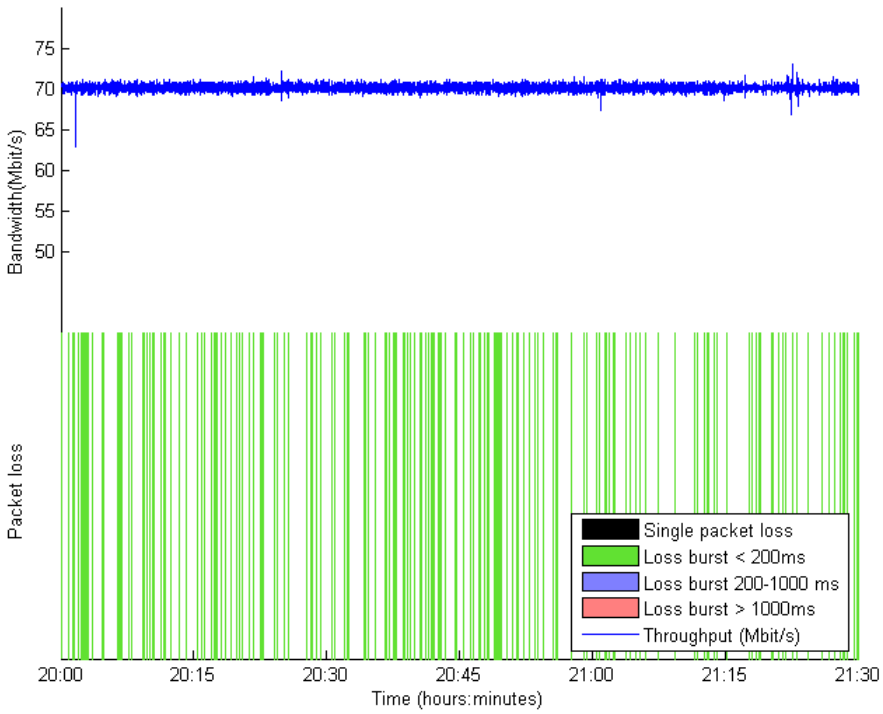
For each test, the results are compared to the contribution profiles form section 3.3.1. The requirements for profile A, B and C are listed and compared to test results. The last column specifies what profile the test result conforms to. The table below show an example.

QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test x**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	23.6	A
Packet loss	percent (%)	<0.05	<2	<20	0.005	A
Max packet loss burst	ms	n/a	<200	<10 000	89	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.12	C
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

B.1 Test 1

This test 05/03/2012 between 20:00 and 21:30. The UDP sending rate was **70 Mbit/s**.

Resulting QoS metrics			**Packet loss distribution**			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
2701 (0.0084%)	62.1	3 ($\tilde{0}$%)	12	169	0	0



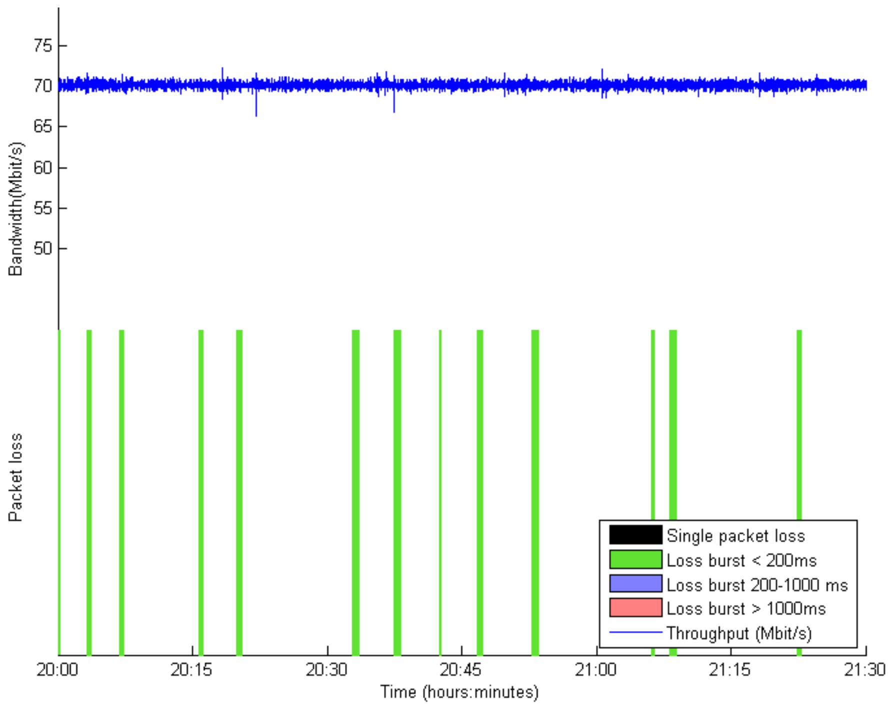
QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test 4**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	62.1	A
Packet loss	percent (%)	<0.05	<2	<20	0.0084	A
Max packet loss burst	ms	n/a	<200	<10 000	7	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0312	C
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

In this test, a lot of small (less than 50 packets) burst losses occurred. However, the total packet loss ration was low. The table show that this test confirms to profile C due to requirements of consecutive packet loss bursts.

B.2 Test 2

This test 06/03/2012 between 20:00 and 21:30. The UDP sending rate was **70 Mbit/s**.

Resulting QoS metrics			**Packet loss distribution**			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
2318 (0.0072%)	43.0	2 (0̃%)	5	77	0	0



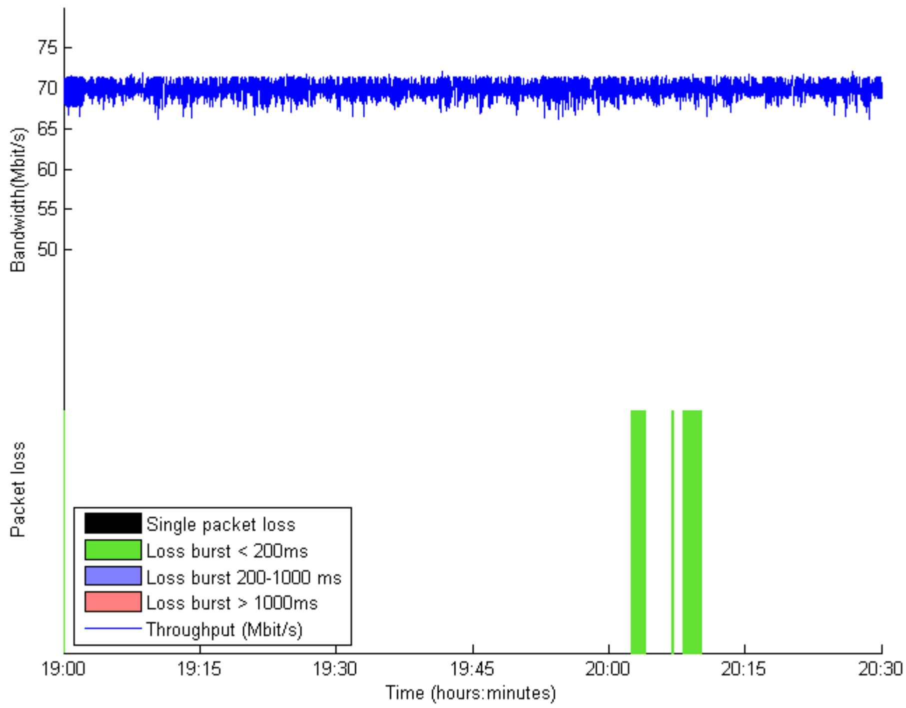
QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test 4**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	43.0	A
Packet loss	percent (%)	<0.05	<2	<20	0.0072	A
Max packet loss burst	ms	n/a	<200	<10 000	10	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0142	C
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

In this test, small bursts (10-60 packets) appear in clusters. However, the total packet loss ration was low. The table show that this test confirms to profile C due to requirements of consecutive packet loss bursts.

B.3 Test 3

This test 22/03/2012 between 19:00 and 20:30. The UDP sending rate was **70 Mbit/s**.

Resulting QoS metrics			**Packet loss distribution**			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
1690 (0.0028%)	35.1	2 ($\tilde{0}$%)	5	6	0	0



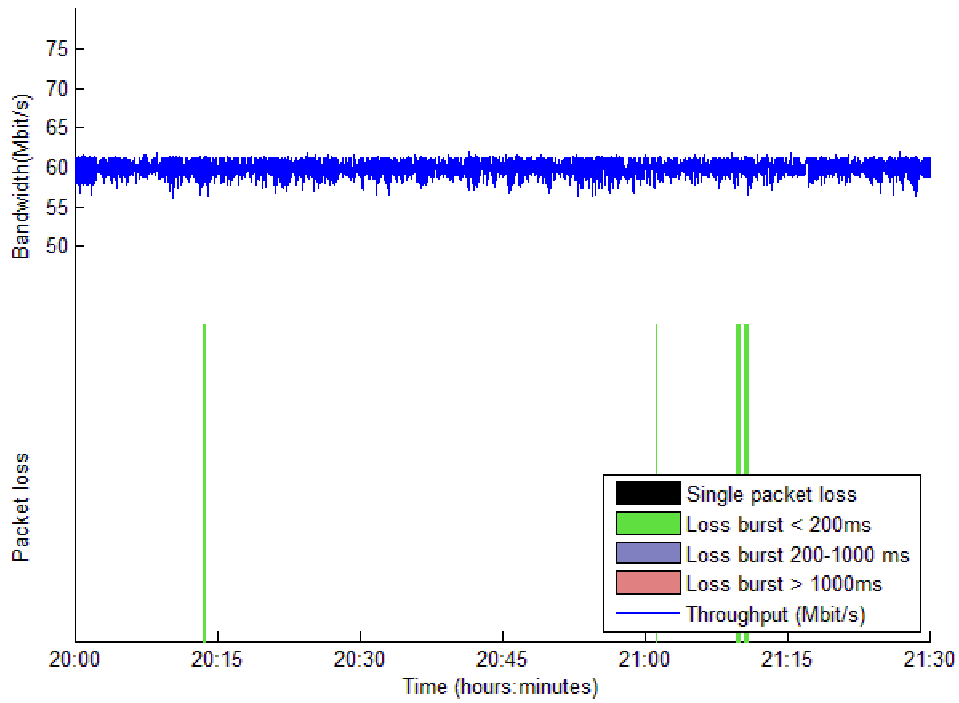
QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test 4**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	35.1	A
Packet loss	percent (%)	<0.05	<2	<20	0.0028	A
Max packet loss burst	ms	n/a	<200	<10 000	156	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0009	B
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

In this test, larger packet bursts occurred - but not above 1000 ms (maximum here was 156 ms). The total packet loss ration was low. The table show that this test confirms to profile C due to requirements of consecutive packet loss bursts.

B.4 Test 4

This test 08/03/2012 between 20:00 and 21:30. The UDP sending rate was **60 Mbit/s**.

Resulting QoS metrics			**Packet loss distribution**			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
1047 (0.0038%)	23.6	2 ($\tilde{0}$%)	0	5	0	0



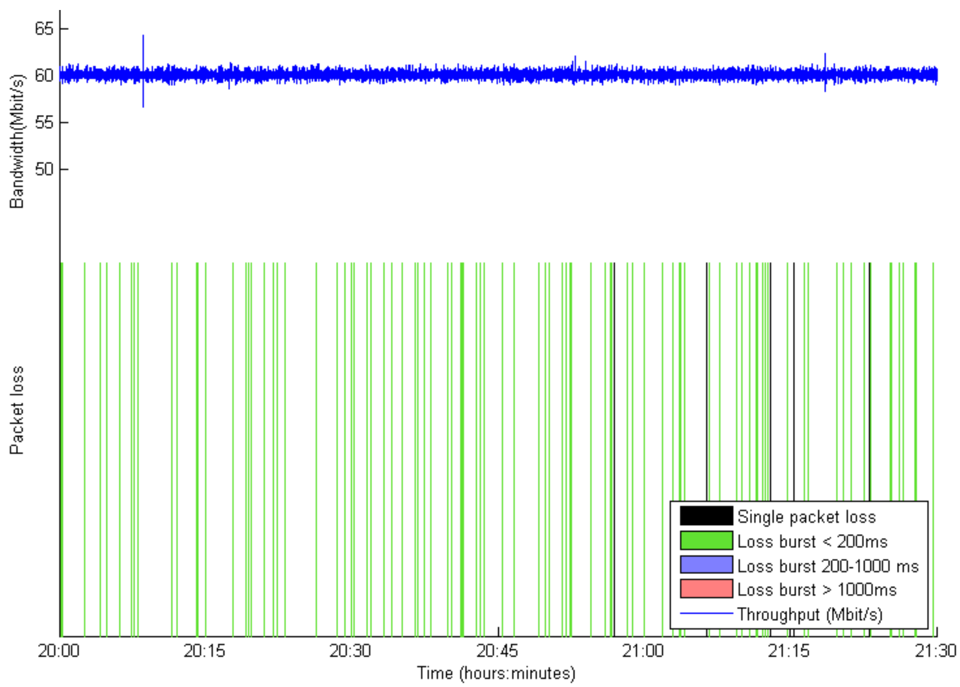
QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test 4**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	23.6	A
Packet loss	percent (%)	<0.05	<2	<20	0.0038	A
Max packet loss burst	ms	n/a	<200	<10 000	89	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0009	B
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

The table show that this test confirms to profile C due to requirements of consecutive packet loss bursts - here the minimum distance was 55.75 sec. No loss burst was above 200 ms.

B.5 Test 5

This test 12/03/2012 between 20:00 and 21:30. The UDP sending rate was **60 Mbit/s**.

Resulting QoS metrics			**Packet loss distribution**			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
1459 (0.0053%)	43.9	2 (0̃%)	7	91	0	0



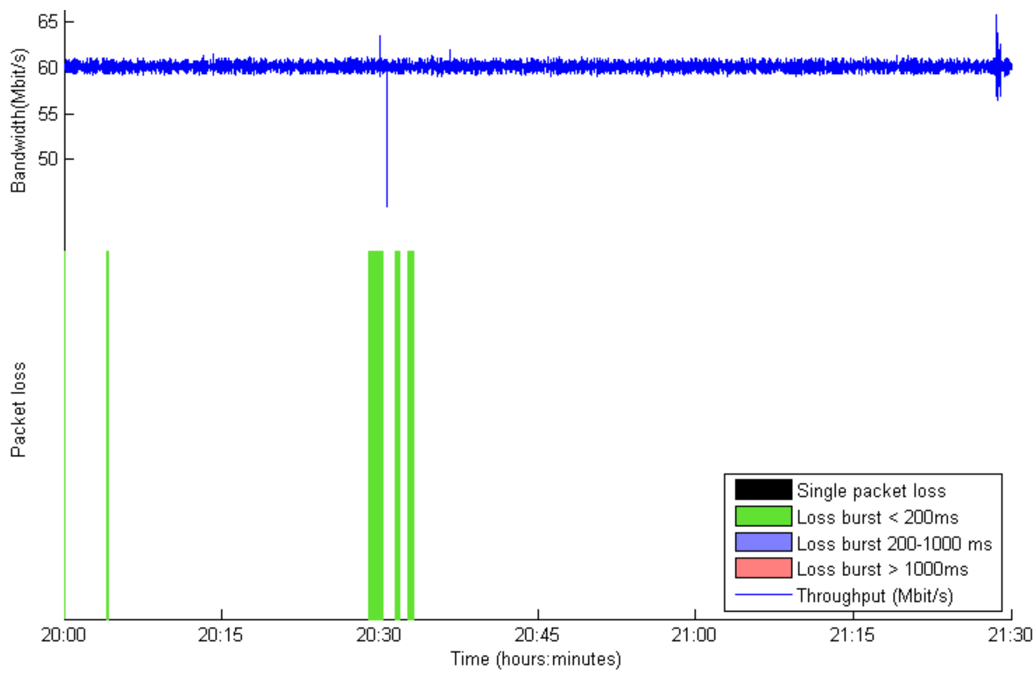
QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test 4**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	43.9	A
Packet loss	percent (%)	<0.05	<2	<20	0.0053	A
Max packet loss burst	ms	n/a	<200	<10 000	48	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0168	C
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

Only small (less than 30 packets) burst losses occurred. The total packet loss ration was low. The table show that this test confirms to profile C due to requirements of consecutive packet loss bursts.

B.6 Test 6

This test 13/03/2012 between 20:00 and 21:30. The UDP sending rate was **60 Mbit/s**.

Resulting QoS metrics			**Packet loss distribution**			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
1663 (0.006%)	53.9	2 (0̃%)	1	4	0	0



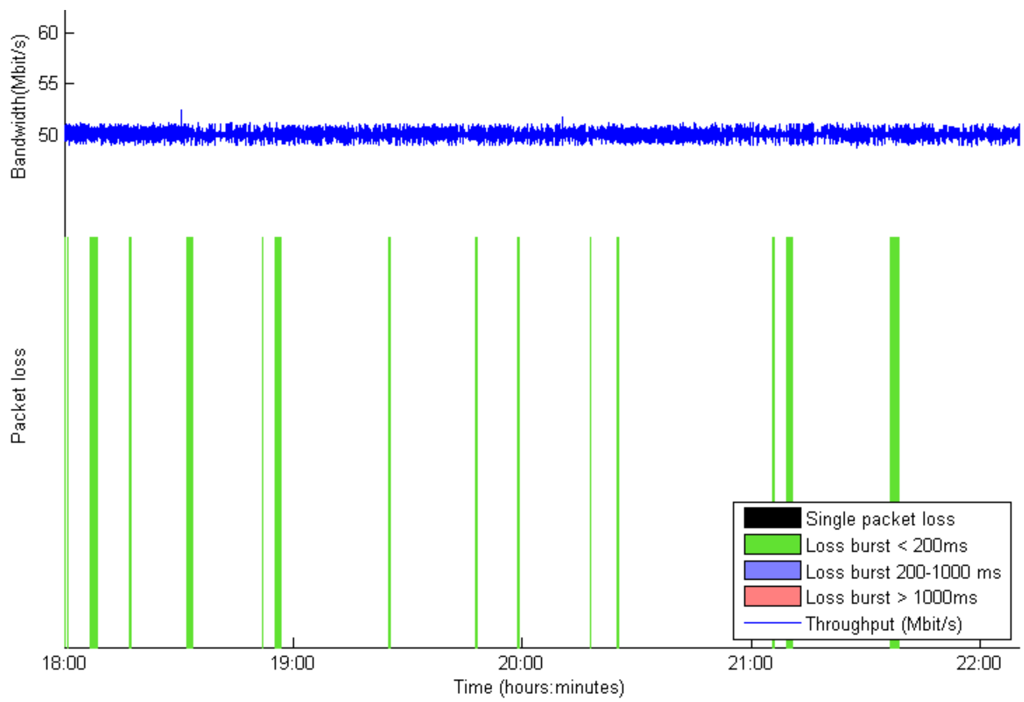
QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test 4**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	53.9	A
Packet loss	percent (%)	<0.05	<2	<20	0.006	A
Max packet loss burst	ms	n/a	<200	<10 000	130	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0007	B
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

Here, fewer but larger bursts occurred - yet, below the 200 ms requirement. The total packet loss ration was low. The table show that this test confirms to profile C due to requirements of consecutive packet loss bursts.

B.7 Test 7

This test 14/03/2012 between 18:00 and 22:00. The UDP sending rate was **50 Mbit/s**.

Resulting QoS metrics			**Packet loss distribution**			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
1679 (0.0027%)	25.3	2 (0̃%)	6	13	0	0



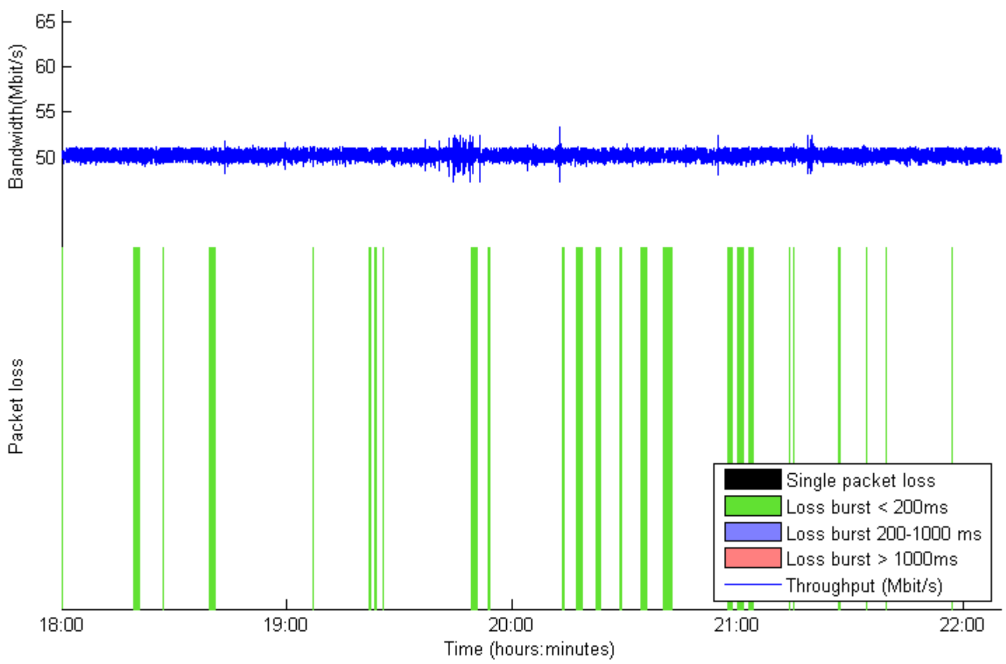
QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test 4**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	53.9	A
Packet loss	percent (%)	<0.05	<2	<20	0.0027	A
Max packet loss burst	ms	n/a	<200	<10 000	59	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0009	B
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

Note that this test lasted for 4 hours. Some larger bursts loss occurred, but all under the 200 ms criteria. The packet loss percent was low, and similar to loss percentage using 70 Mbit/s. The table show that this test confirms to profile C due to requirements of consecutive packet loss bursts.

B.8 Test 8

This test 15/03/2012 between 18:00 and 22:00. The UDP sending rate was **50 Mbit/s**.

Resulting QoS metrics			**Packet loss distribution**			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
1917 (0.0031%)	21.4	2 ($\tilde{0}$%)	12	19	0	0



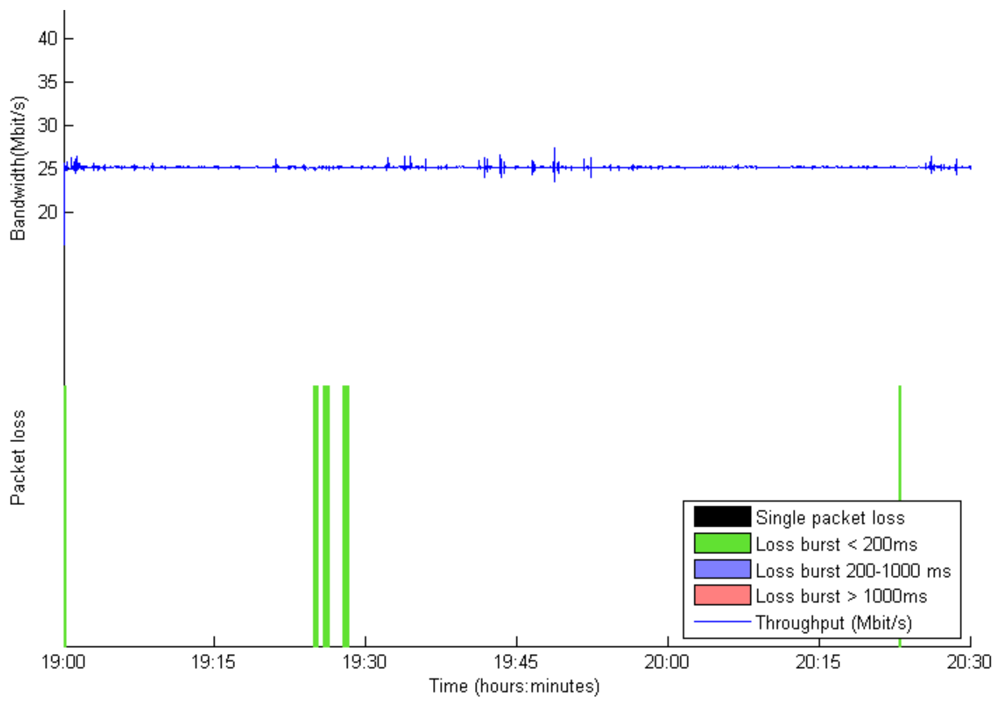
QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test 4**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	21.4	A
Packet loss	percent (%)	<0.05	<2	<20	0.0031	A
Max packet loss burst	ms	n/a	<200	<10 000	51	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0013	C
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

The results here very similar to test 7. The total packet loss ration was low. The table show that this test confirms to profile C due to requirements of consecutive packet loss bursts.

B.9 Test 9

This test 19/03/2012 between 19:00 and 20:30. The UDP sending rate was **25 Mbit/s**.

Resulting QoS metrics			Packet loss distribution			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
207 (0.0017%)	25.0	2 ($\tilde{0}$%)	0	5	0	0



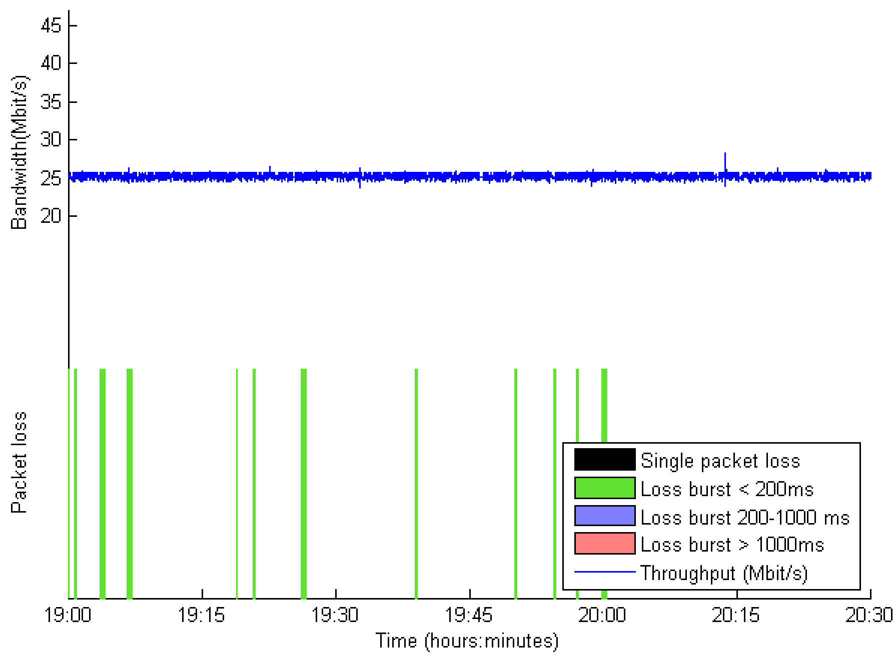
QoS metric	**Unit**	**Profile A**	**Profile B**	**Profile C**	**Test 4**	**Conformity**
Jitter (IPDV)	ms	<50	<150	<500	25.0	A
Packet loss	percent (%)	<0.05	<2	<20	0.0017	A
Packet loss burst	ms	n/a	<200	<10 000	56	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0009	B
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

Here, 4 large bursts occured - yet, below the 200 ms requirement. The total packet loss ration was low. The table show that this test confirms to profile C due to requirements of consecutive packet loss bursts.

B.10 Test 10

This test 20/03/2012 between 19:00 and 20:30. The UDP sending rate was **25 Mbit/s**.

Resulting QoS metrics			Packet loss distribution			
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	*Bursts*		
				< 200ms	200-1000ms	>1000ms
397 (0.0034%)	31.8	2 (0̃%)	3	10	0	0

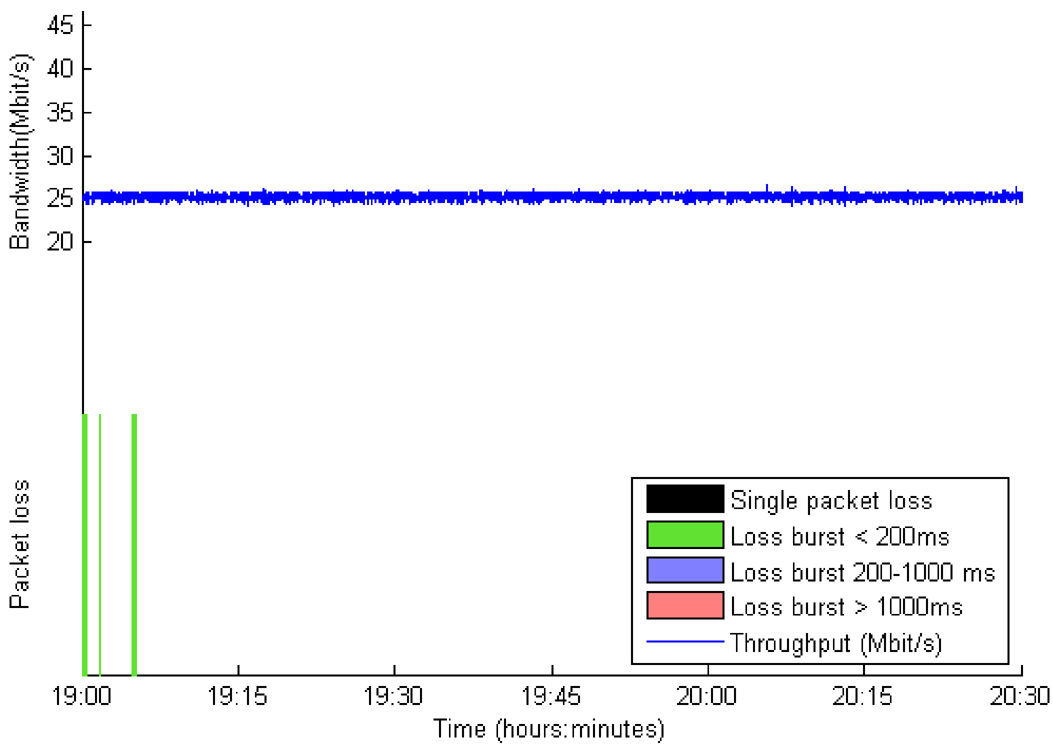


QoS metric	Unit	Profile A	Profile B	Profile C	Test 4	Conformity
Jitter (IPDV)	ms	<50	<150	<500	31.8	A
Packet loss	percent (%)	<0.05	<2	<20	0.0034	A
Max packet loss burst	ms	n/a	<200	<10 000	86	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0018	C
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

B.11 Test 11

This test 21/03/2012 between 19:00 and 20:30. The UDP sending rate was **25 Mbit/s**. Totally, 11489299 packages were sent.

Resulting QoS metrics			Packet loss distribution			
				Bursts		
Packets lost	*IPDV (ms)*	*Reordering*	*Singles*	< 200ms	200-1000ms	>1000ms
297 (0.0025%)	19.9	2 ($\tilde{0}$%)	1	4	0	0



QoS metric	Unit	Profile A	Profile B	Profile C	Test 4	Conformity
Jitter (IPDV)	ms	<50	<150	<500	19.9	A
Packet loss	percent (%)	<0.05	<2	<20	0.0043	A
Max packet loss burst	ms	n/a	<200	<10 000	91	B
Loss bursts rate	1/s	n/a	0.001	0.1	0.0007	B
Reordering	percent (%)	<0.001	<0.01	<0.1	>0	A

Appendix C

Buffer refilling problem

This problem here is to find the sending rate and the buffer refilling rate such that the buffer can be refilled with content following a retransmission when the buffer is emptied. The sum of these rates must be below the first-mile bandwidth.

Let *B* be the availble bandwidth in Megabits/second and *bs* be the buffer size is ms. *D* is the availible build up time in ms, that is, the target time to refill the buffer. Finally, *P* is the availible media sending bandwidth (equal Playback rate), and *F* is the availible bandwidth to Fill the buffer. The buffer refilling problem is then defined as:

Problem: Given *B*, what is the resulting values of *R* and *F* such that the empty buffer is completely filled with content equal *bs* in a time interval *D*.

Matemathically, the problem is

$$R\,bs = F\,D \tag{C.1}$$

Given that first-mile bandwidth is fully utilized, but not exceeded

$$B = R + F \tag{C.2}$$

C pasted into C gives

$$R\,bs = (B - R)\,D$$

After some rearrangments

$$R = \frac{B\,D}{bs\;+\;D}, \qquad F = B - R \tag{C.3}$$

For example,

$$B = 70Mbit/s, D = 2s, bs = 250ms \tag{C.4}$$

Gives,

$$R = 62.7Mbit/s \wedge\; F = 7.3Mbit/s$$

Appendix D

Uninett core network

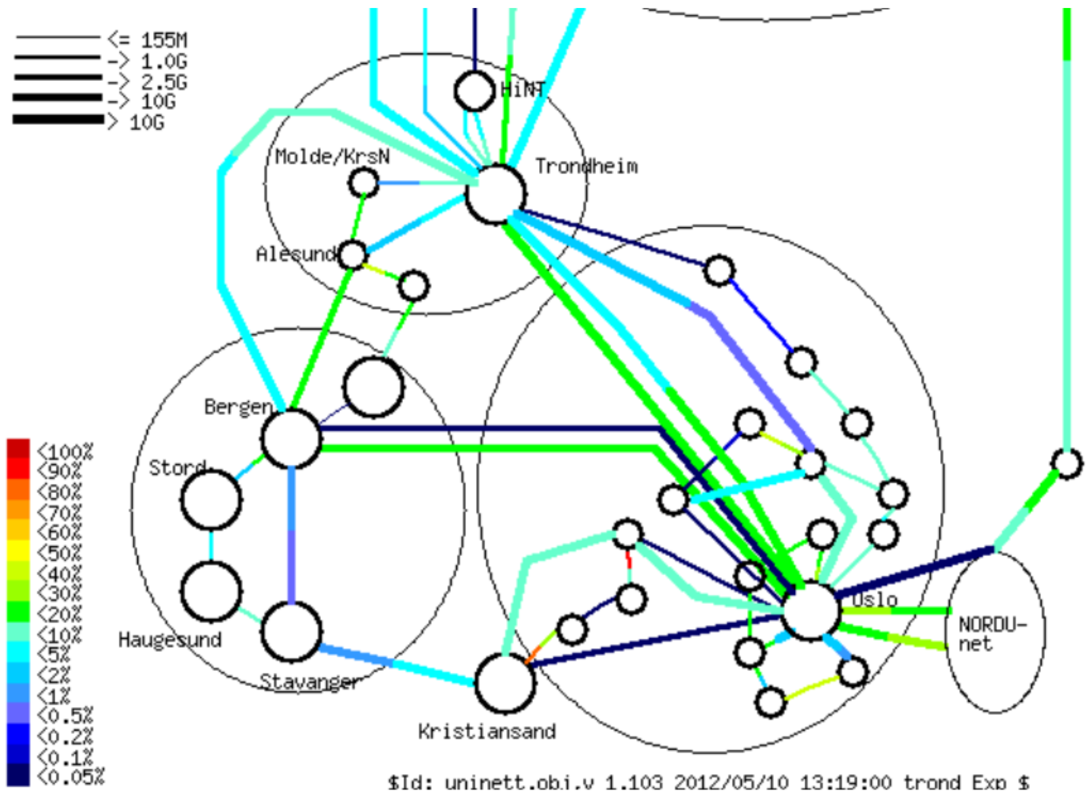


Figure D.1: Figur showing network load in Uninett core network (only showing the southern part of Norway) at mid-day on 18/05/2012. The link between Trondheim and Oslo, used for testing in this thesis, is a 10Gbit/s fiber line with load level below 20%. Hence, this is not a bottleneck in the connection.

Appendix E

Forward Error Correction (FEC)

As recommended by the Pro-MPEG forum, a XOR-FEC strategy is common in professional contribution. The FEC stream is sent on a parallel UDP port to the media stream, allowing correction of lost or corrupted RTP/UDP packets at the receiver.

The generation of FEC packets is done by calculating an error correcting checksum for a matrix of media packets. A set of *L* x *D* consecutive RTP/UDP packets carrying the media content is delayed at the sender, allowing checksum calculation before these are sent. Figure E.1 illustrate this. Here, the matrix of media packets (M_y) is delayed to produce the purple FEC packets (FEC_x).

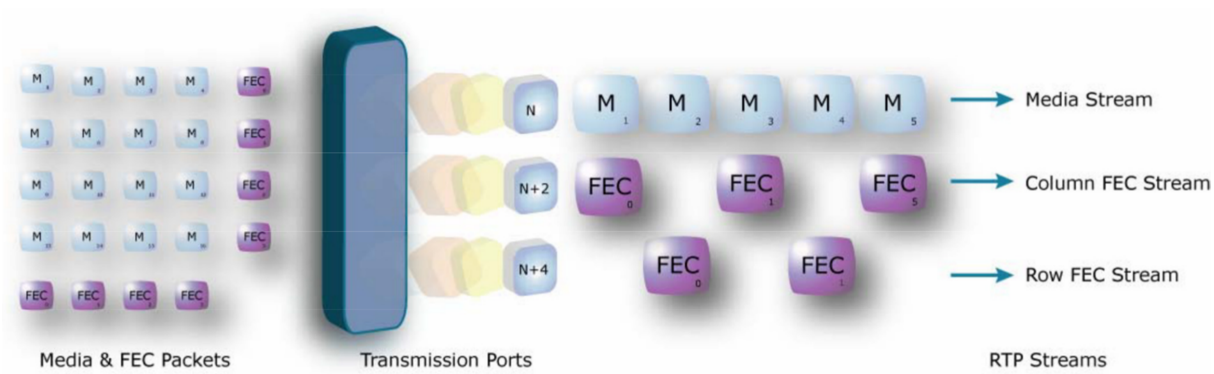


Figure E.1: Illustration of Forward Error Correction (FEC). Illustration found at www.tandberg.com .

The FEC stream can be produced by only producing FEC packets for the columns in the media matrix. For more robust protection, FEC packets can be produced for the rows in the media matrix as well. In general, column-FEC has the nice property that, for every LxD sized block of packets, consecutive erasure errors up to length L is recoverable. Thus, this strategy is ideal for correcting bursts errors. Row-FEC is ideal for correcting non-consecutive loss, i.e. random loss.

The principle is quite simple. In column FEC, every column of media packets in the matrix produces one FEC packet. Let FEC_x be FEC packet *x* and M_y be the media packet at time instant *y*. Then, as an example given *L*=*D*=4 as in figure E.1;

$$FEC_0 = M_1 \oplus M_5 \oplus M_5 \oplus M_9 \oplus M_{13}$$

Thus, if any of the packets in column 1, say M_5, is lost during the transmission, then this packet can be recovered:

$$M_5 = M_1 \oplus FEC_0 \oplus M_5 \oplus M_9 \oplus M_{13}$$

Although the bandwidth of the media stream is not directly affected, the FEC stream introduces a bandwidth overhead in the network according to the size of matrix. E.2 presents the bandwidth overhead with RTP transmission. Clearly, 2D FEC has a much higher cost than 1D FEC. Therefore, the use of 2D FEC may only be justified in networks with high error rate.

Matrix (L,D)	Column (1D)	Column & Row (2D)
(4,4)	32%	58%
(10,10)	16%	26%
(5,20)	20%	32%
(20,5)	26%	32%

Figure E.2: Bandwidth overhead introduced by column-FEC (1D) and a combination of column-FEC and row-FEC. Illustration found at www.tandberg.com .

Only a limited set of consecutive errors can be corrected using FEC. Figure E.2 shows that when L=20 and D=5, which means that a maximum burst of 20 packets can be corrected, a overhead of 26% is introduced. FEC is therefore suitable for IP network where only random errors or very short bursts occur. This is the case for professional contribution networks, and therefore FEC is a very nice strategy in such networks.

In addition to transmission overhead, latency at the decoder side is unavoidable. This latency is a result of the decoder wait for the LxD block of media packets, before FEC calculations for that can be performed. This latency is roughly proportional to matrix size. Since the latency is directly tied to the level of overhead, there is a trade off between these parameters.

Appendix F

Introduction sheet to subjective test

About the test

This test has two parts:

1. You will watch a movie for about 2 hours
2. You will fill out a form which takes roughly 10 minutes

We want to measure your experience using this setup. Basically, this setup is supposed to be as good as cable-TV. Therefore, pretend you are watching this movie via cable-TV. We will be asking you about your experience with this setup versus cable-TV after the test.

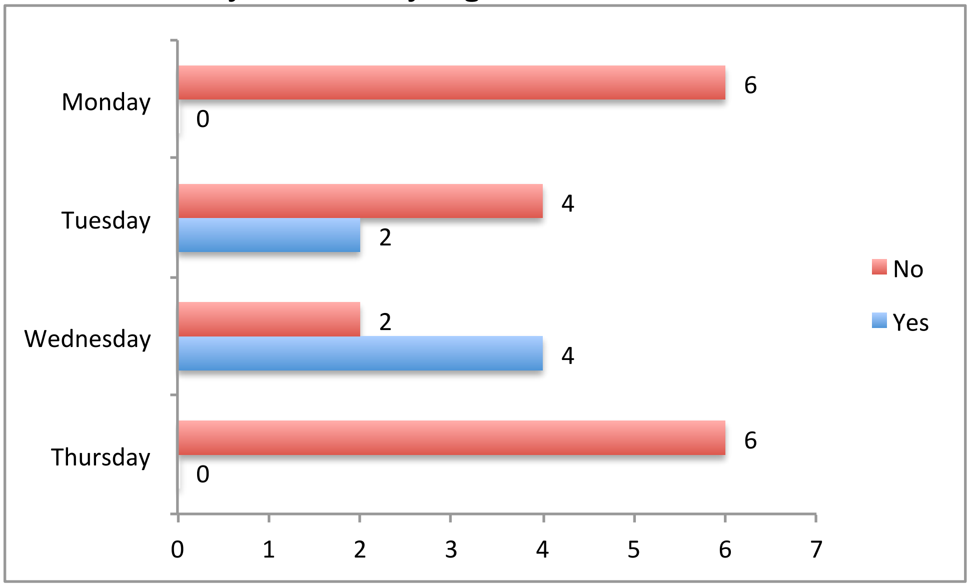
The movie itself should have sufficiently high picture quality for your comfort. We are NOT measuring what you think of the picture quality. We have not intentionally done anything with the video; for all we know, it might be perfect all the time.

We are measuring how comfortable you think the movie is to watch, in terms of being fluent and without artifacts (norsk: måler hvor flytende/hakkfri du synes filmen er å se på). Do not concentrate hard on finding any errors, try to relax and watch the movie, as you would when watching cable TV.

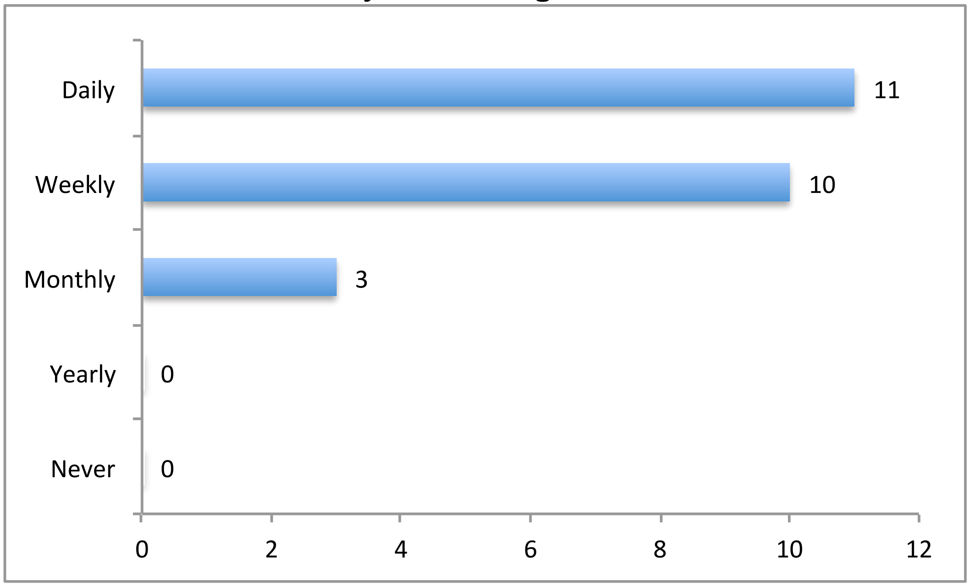
Appendix G

Survey results from subjective test

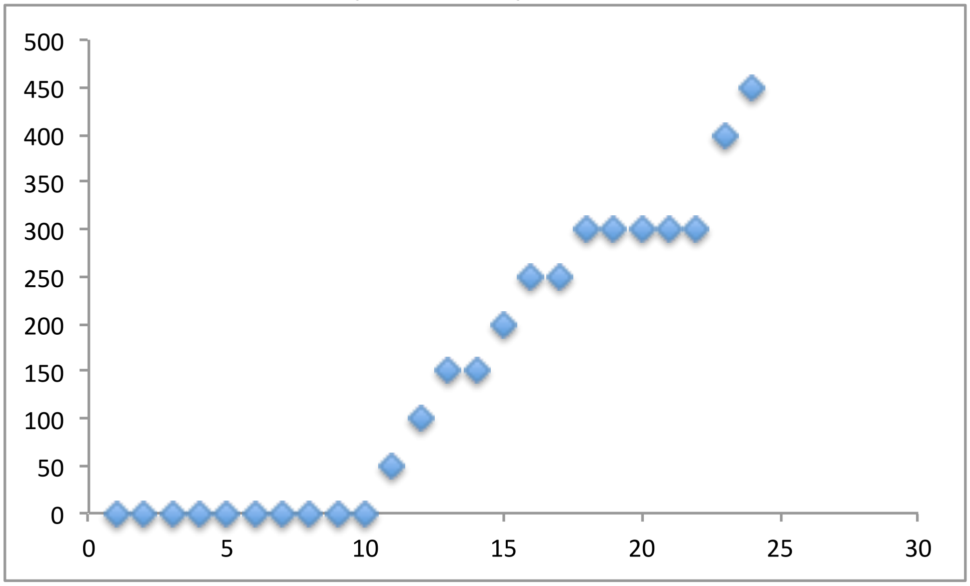
Question 1: Did you notice any degradations in the video?



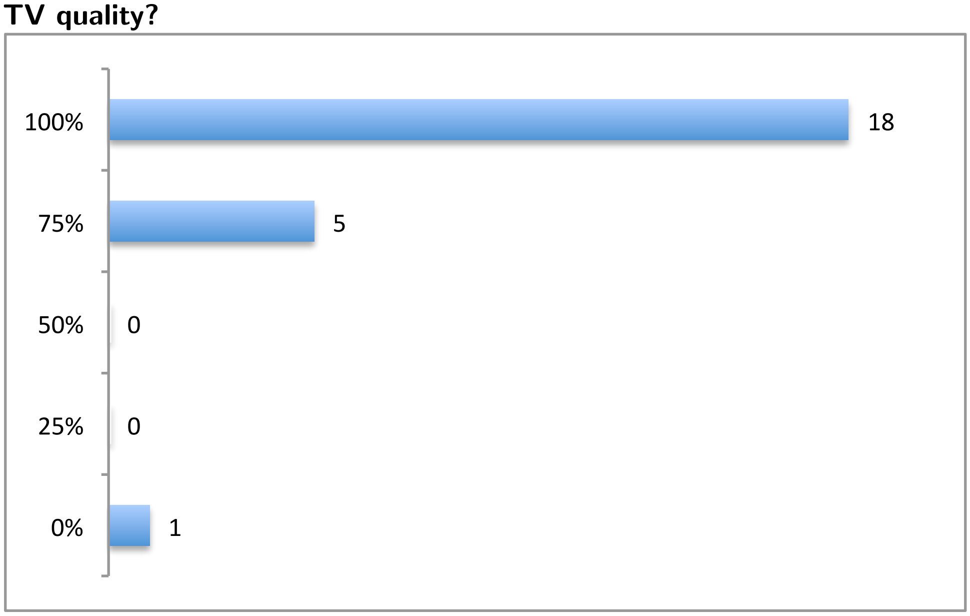
Question 2: How often do you watch regular TV?



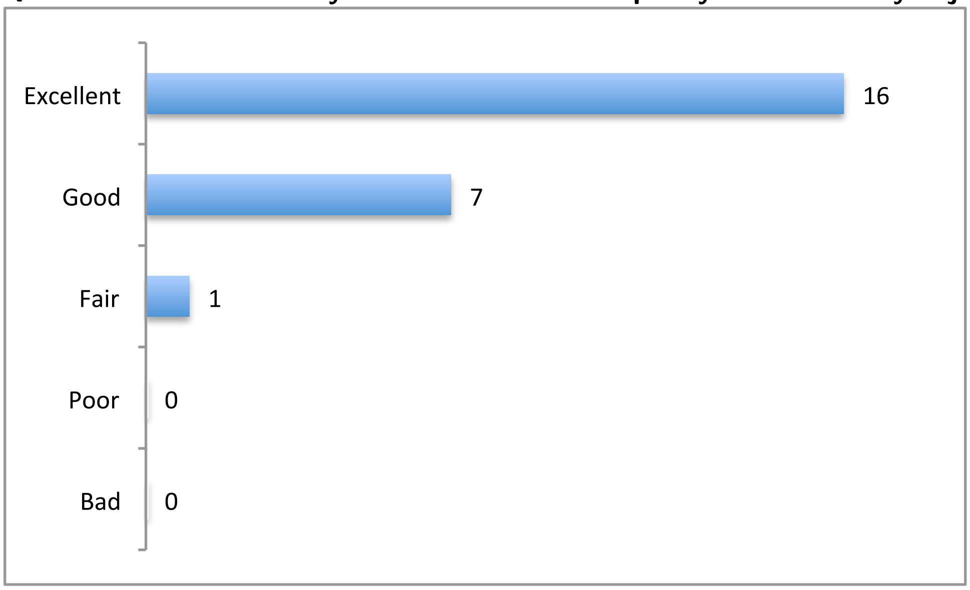
Question 3: How much is your monthly TV-subscription now?



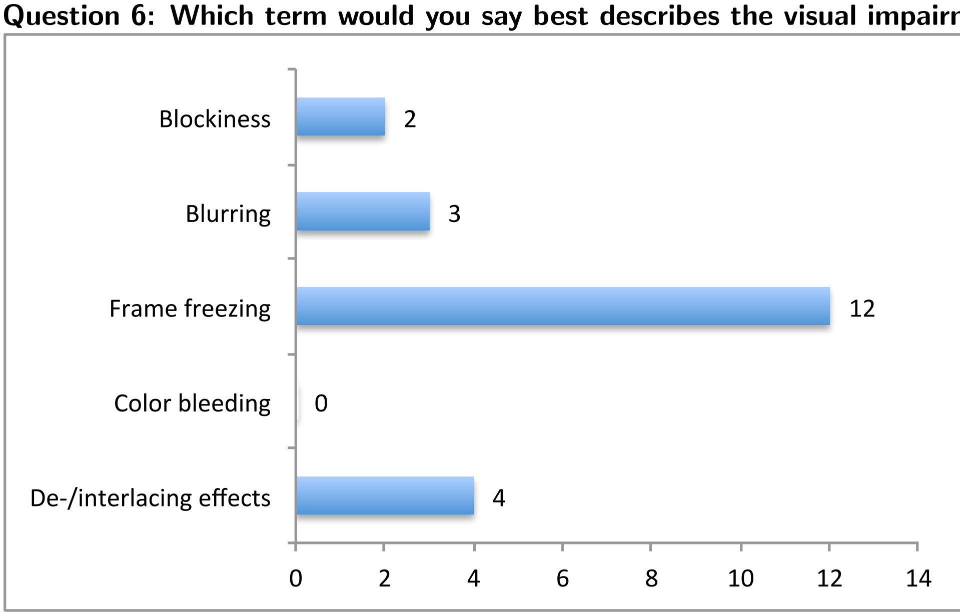
Question 4: How much would you be willing to pay to get this quality compared to regular TV quality?



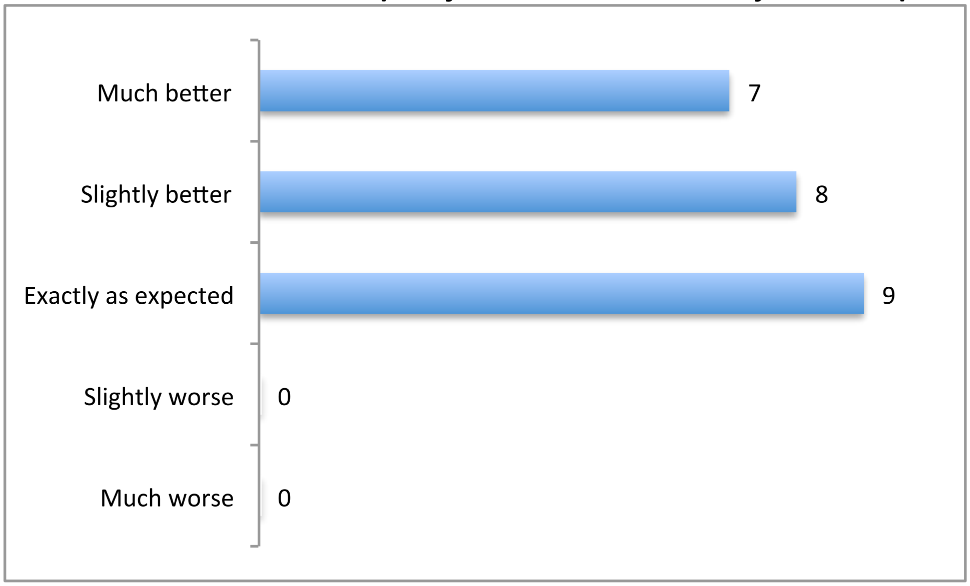
Question 5: How would you best describe the quality of the video you just saw?



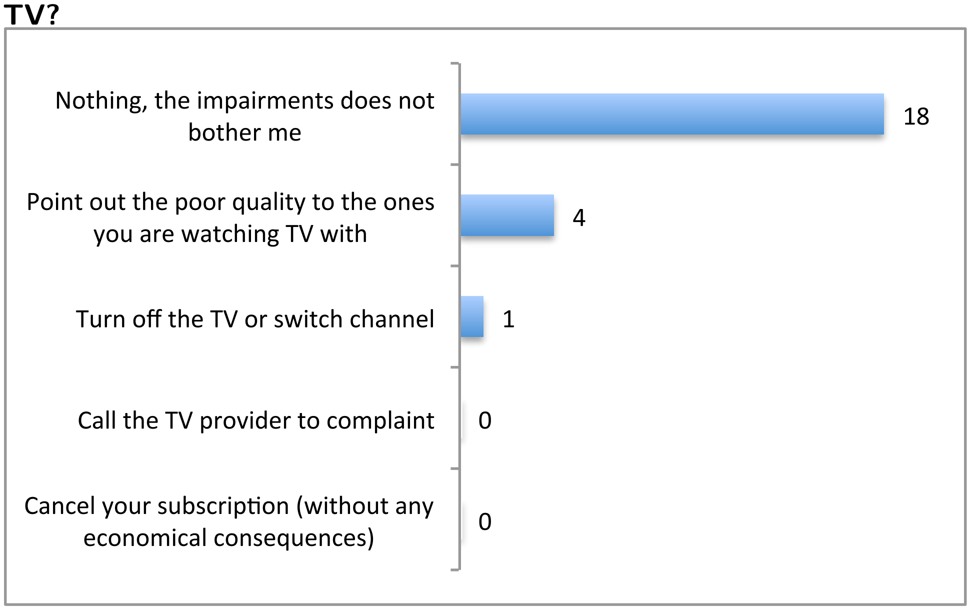
Question 6: Which term would you say best describes the visual impairments in the video?



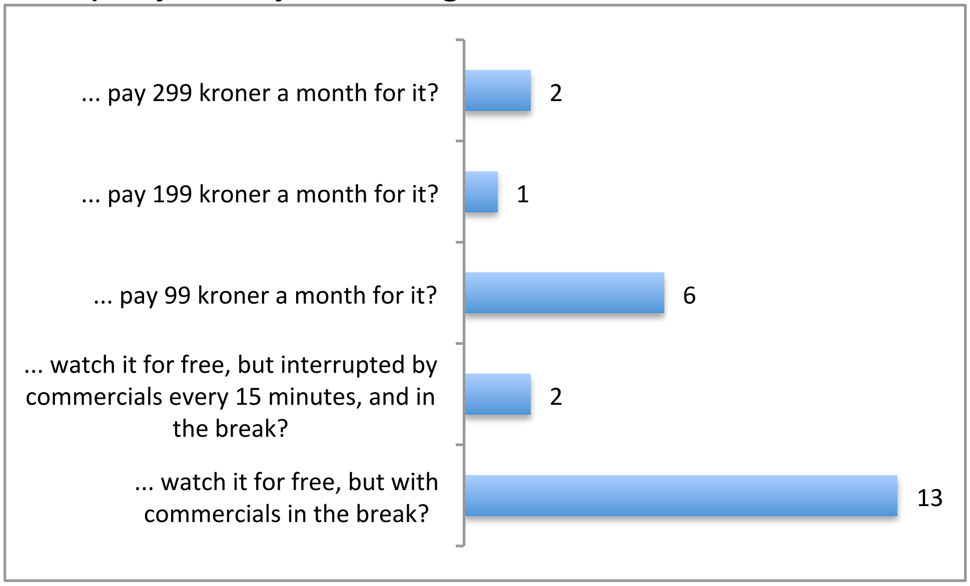
Question 7: Was the video quality better or worse than you had expected?



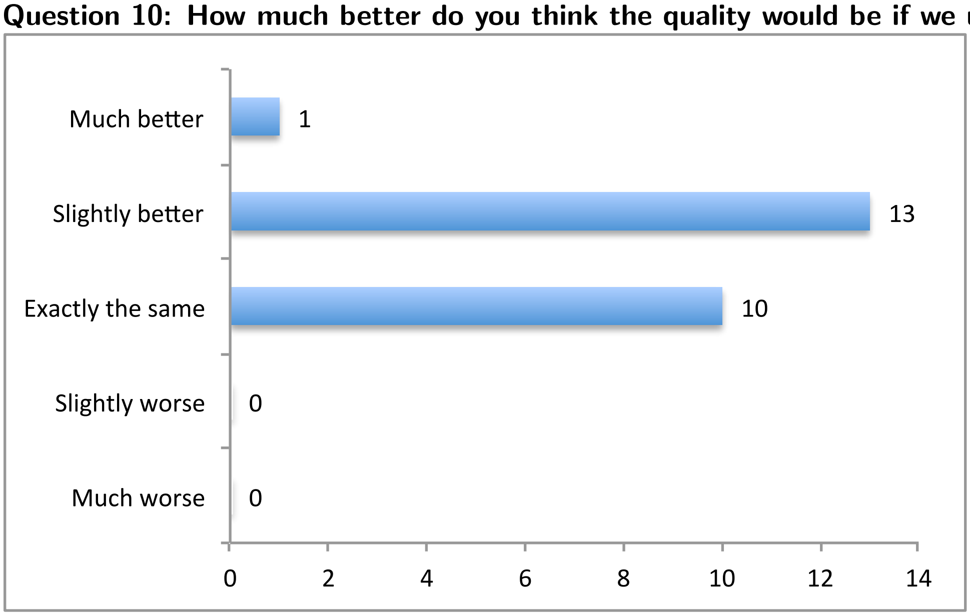
Question 8: Which action would you take if you experienced this kind of quality on regular TV?



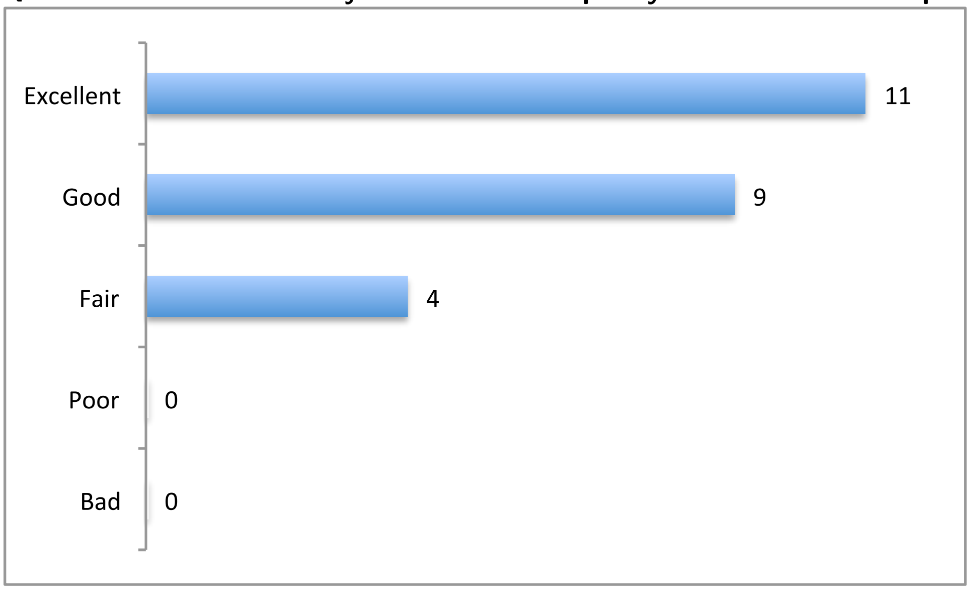
Question 9: If you were abel to watch all Tippeliga football matches for a year with this video quality, would you be willing to ...



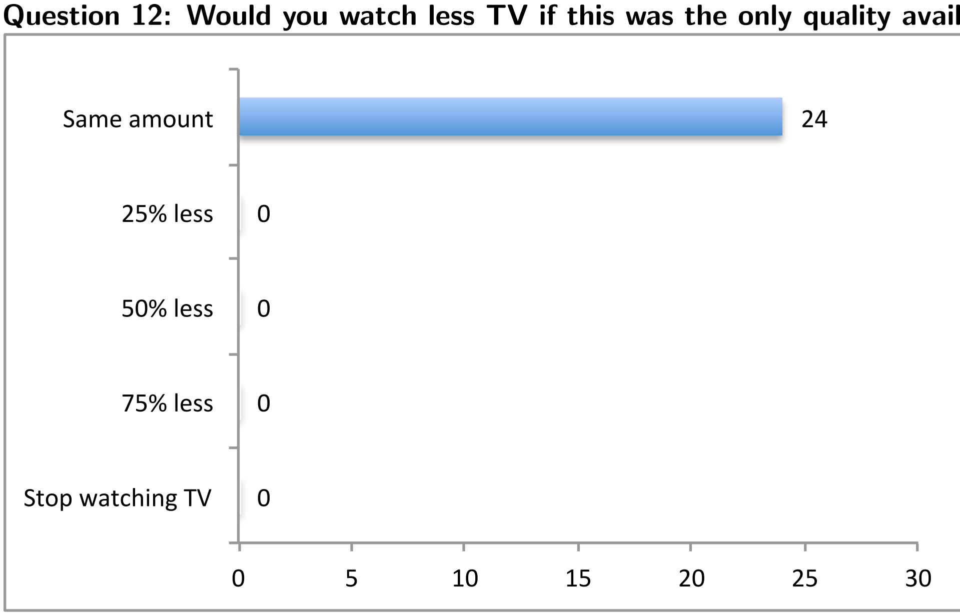
Question 10: How much better do you think the quality would be if we used HD?



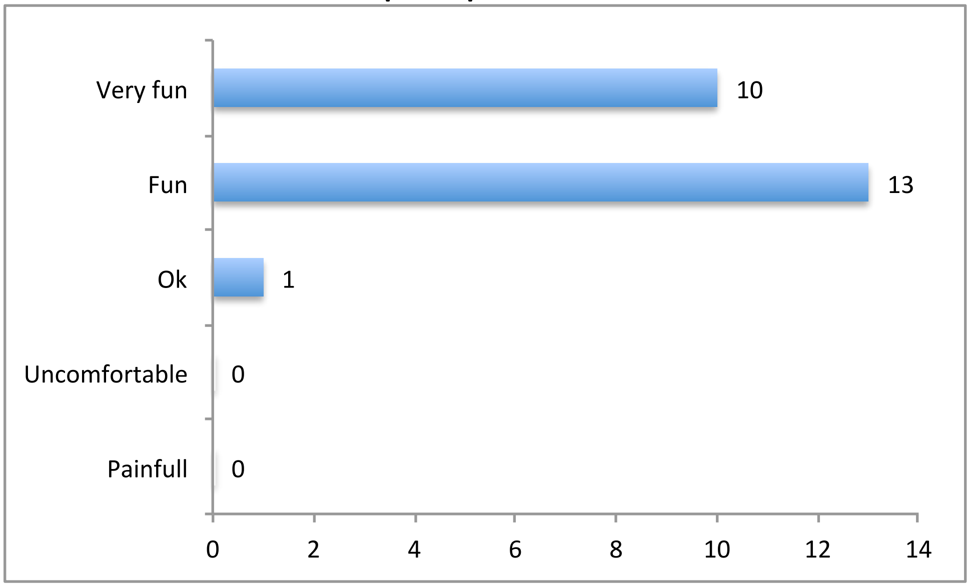
Question 11: How would you describe the quality of the video when packet loss occured?



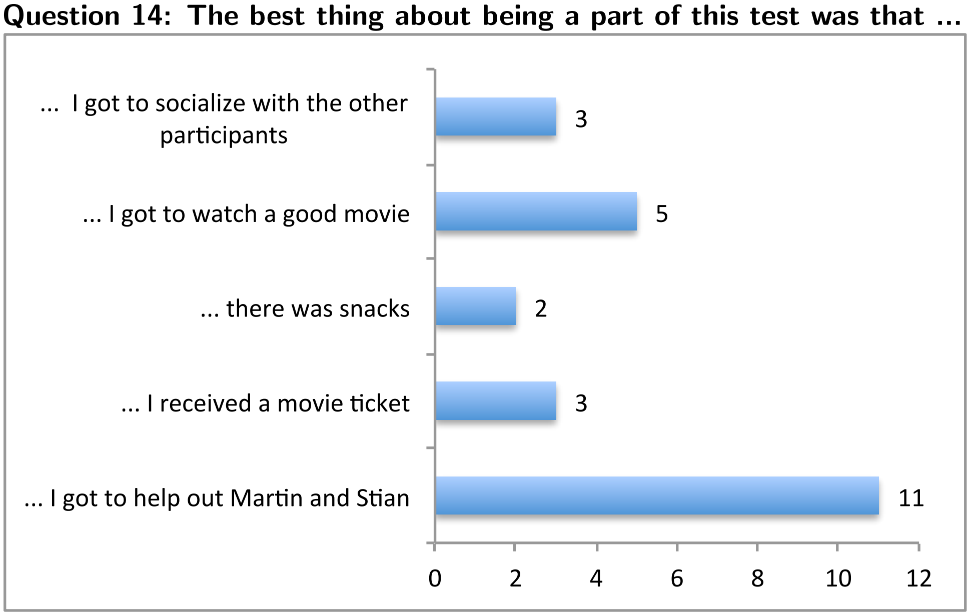
Question 12: Would you watch less TV if this was the only quality available?



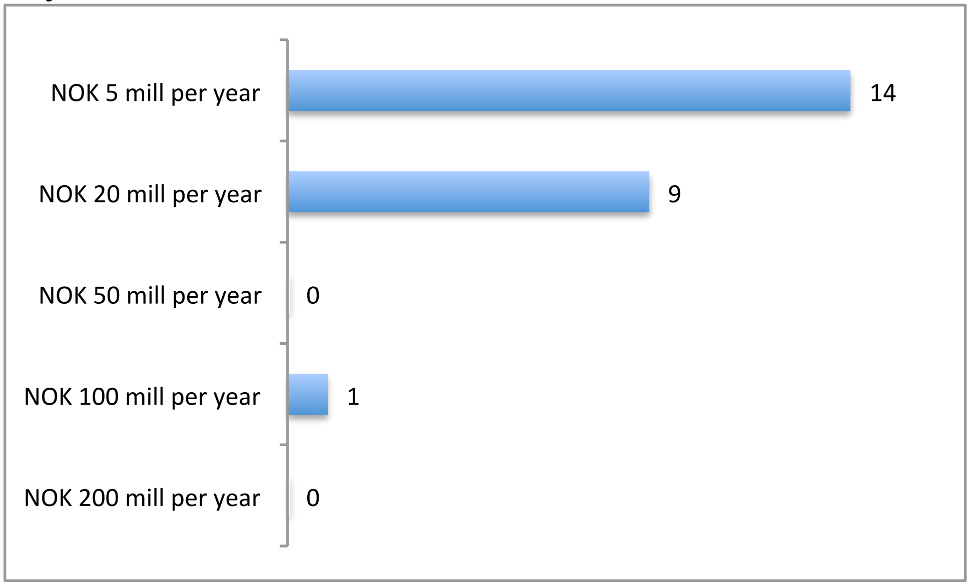
Question 13: How was it to participate in this test?



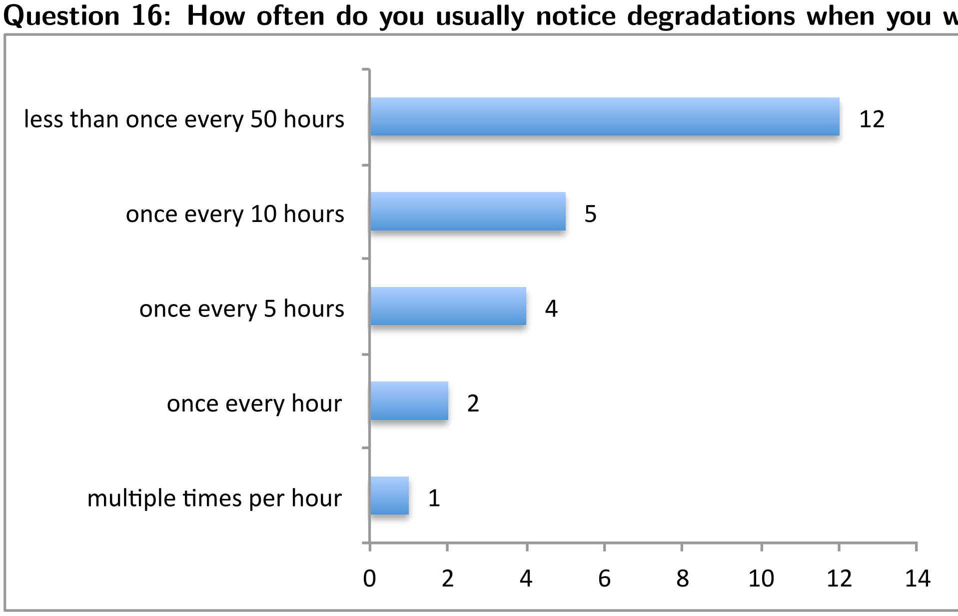
Question 14: The best thing about being a part of this test was that ...



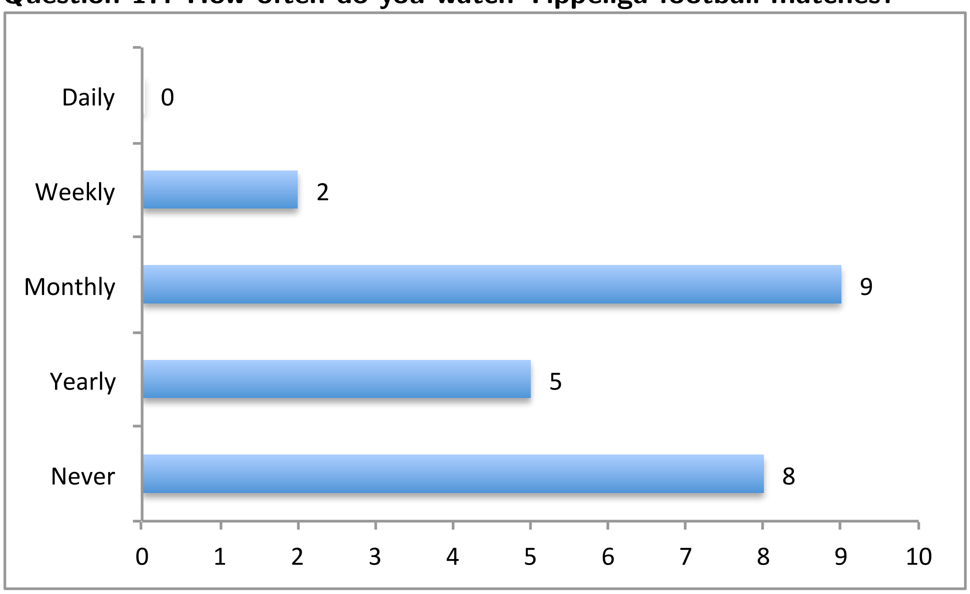
Question 15: Would you recommend that NRK start using this technology if it ment that they could save more than ...



Question 16: How often do you usually notice degradations when you watch cable-TV?



Question 17: How often do you watch Tippeliga football matches?



Appendix H

Research protocol

Research protocol: Subjective test of contribution over Internet

Stian Tokheim and Martin Markman

March 19. 2012

Synopsis

This protocol shows how we would like to research the visual impairments on video introduced by contribution over Internet. The research will tell us whether or not this setup gives a satisfying experience to an end user. It will be conducted using the Internet, giving us a realistic network scenario.

We will have 24 participants watch a full movie that is sent over the Internet between two video gateways made for professional contribution. The participants will describe the experience in a questionnaire we have developed. We will also log the network activity, and from this we can find out when packet loss occurred. When visible impairments due to packet loss appear, this will be registered by an expert. A combination of all this data will give us a clear description of what happened, and we will be able to discuss what the participants experienced in the context of this. For practical reasons the test is done in four parts, with 6 participants in each.

The results will give us a notion of how well contribution over Internet works. This will be an important part in a bigger study that aims to explain if and how contribution over Internet can be done.

1 Introduction and background

1.1 Introduction to the topic of interest

Video contribution is used by professional broadcasters. Live video is conveyed from a recording site, for example a football stadium, to a production center. The term *contribution* is limited to this part of the production chain. The video is then edited and distributed to TV viewers. Contribution over IP is usually done over a private IP-network. The broadcasters are able to set up the routers and control the traffic as they want, giving them high and reliable performance. The Internet is a public IP-network that can be used by anybody. It is relatively inexpensive compared to a private IP-network, but can not generally guarantee reliable performance.

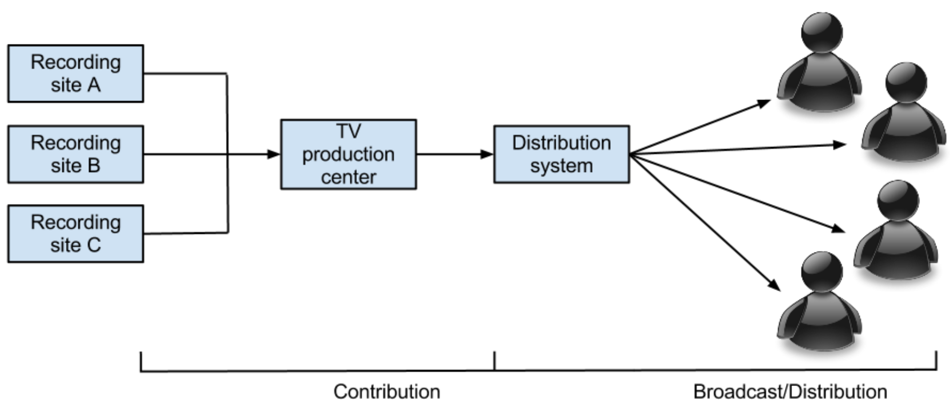


Figure 1: This illustrates how contribution is used, and how it is different from distribution.

1.2 What is already known and what is missing

Contribution over IP-networks with guaranteed QoS is a well known and well studied technology, and is used by professional broadcasters around the world. There is little or no packet loss in these networks, [Helge Stephansen, 2007] shows that 52 packets was lost over 45 days on a good link used by NRK (Norwegian Broadcasting Corporation). The video consequently suffers from little or no degradation.

The worlds largest IP-network, the Internet, has long been considered too unreliable and unable to provide enough bandwidth to be used for contribution. The key difference between a private IP-network and the public Internet, is that Internet has no guaranteed Quality of Service, it is a so-called best-effort network. We have found no record or reports of trials with contribution over Internet, although it probably has been tried. Our goal is to find out and document if it works and how well it works.

We would like to do this from an *end user* point of view: A person watching television that is conveyed to a production center using contribution over Internet, and broadcasted to his or her television. To only test the effects of the contribution stage, we shortcut the broadcast stage by connecting a TV directly to the receiving video gateway. In Figure 1 this would be like using only one recording site, replacing the TV production center with a TV screen and voiding the distribution. This is what we mean when we use the term *end user* in this protocol.

In preparations for this test, we have done network tests by sending dummy data, and a "proof of concept" (POC) test where we set up the system and confirmed that it worked. These tests showed that:

- Packet loss and bursts appear, but not very often (typically a few times per hour). They are sometimes visible.
- In the POC test, the video arrived in good condition most of the time. Sometimes we were able to notice packet loss. The test lasted for 7 hours, and nobody left because of degradations in the video, which indicates that the video was not painful to watch. Some of the smaller bursts were not registered as visible.
- The jitter-buffers in the gateways are 100 ms, and at no time have we observed delay larger than 100 ms. Therefore, the only QoS network parameter we have to concern ourself with is packet loss.

1.3 Scope of this test

We limit our testing to one connection over a relatively short distance (a few kilometers) with only a few router hops. This is close to a scenario with a local sports event being covered by a local broadcaster. The study is further limited to video, not sound.

Many techniques have been developed to handle different network scenarios:

- Delay and jitter: A short buffer on the receiving end can compensate for jitter and delays of less than the buffer size.
- Packet reordering: The RTP protocol is used on top of UDP to handle reordering.
- Packet loss: A protocol that does retransmission can avoid packet loss. This requires larger buffers on both ends. Forward error correction (FEC) is also sometimes used to compensate for small bursts.

Our test will use two video gateways, borrowed from the company T-VIPS, made for professional contribution over IP-networks. The gateways has a jitter buffer (up to 100ms) and uses RTP over UDP to transport the packets, thus jitter and packet reordering will not be a problem. They do not use any form of retransmission, so packet loss will cause problems. Packet loss is handled by freezing the frame, and will often be visible to the viewer.

It is most interesting for us to test this in a realistic network scenario, and to try to "push it to the limit". Therefore, we sent the video on a constant bitrate of 70 MBit/s, the maximum bandwidth offered by the ISP, on an afternoon, traditionally a peak period for Internet traffic. Although an available option in the gateways, FEC will not be used in our test. On the basis of the packet loss data from the network logs, we are able to analyze which of the bursts could be completely compensated by FEC. Thus, using FEC would yield no additional information.

1.4 Problem statement

The aim of this test is to give a reasonable description of experienced video quality when no measures are taken to prevent packet loss. The results will serve as basis for a discussion about the use of Internet for contribution; whether it can be used, and how packet loss should be handled.

To get such a description, we set up the system, use it to show a movie (about 3 hours long) to 24 participants and get them to evaluate it through a questionnaire. For practical reasons, the test will be done in four parts, in groups of 6. The tests will be done in the afternoon to get similar network conditions.

1.5 Impact of solution

It is much cheaper to use the Internet than to use dedicated networks. In that perspective, it is preferable for broadcasters to start using Internet for contribution. Big broadcasters, like NRK, have high demands to the quality of the video they broadcast: after all, it is watched by many people all over the country, and sometimes the world. But, in recent years, streaming over Internet has made it possible to do low cost broadcasting at a smaller scale. An example is local newspapers embedding videos on their web pages. These broadcasters and their customers might be willing to accept lower quality at a lower cost.

There are many broadcasters between the extremities main national broadcaster and local newspaper, and some of these might also be satisfied with the reliability that contribution over Internet gives. In addition to existing broadcasters, lower costs will make way for new businesses and business models. We aim to establish a notion of how well contribution over Internet works. With this notion in place, businesses that might profit from this technology will have a reason to consider applying it to their specific needs.

2 Hypothesis

Contribution over the public Internet gives an equally good end user experience as contribution over a private IP-network.

3 Methodology and design

In short, the test will be done as following:

- We will set up the two video gateways to send video over Internet. The sending gateway is connected to a computer that provides the video, and the receiving gateway is connected to a TV that shows the video.
- 24 participants will watch a movie and answer a questionnaire. They are divided in groups of 6 over 4 different days. At least one expert will be watching the movie at the same time, and recording any visible packet loss.

The test will give us three kinds of data:

1. A set of answers from a questionnaire. This is provided by the participants after they have watched the movie.
2. A log of visible packet loss, registered by the expert.
3. A log of the network activity. Information about the packets are logged on the sender and on the receiver side.

3.1 Questionnaire

From the hypothesis it is obvious that we want to find a way to measure the quality of what the end user experience. [Fiedler et al., 2010] says about the metric QoE that *"It links user perception and expectations on one side and technical Quality of Service parameters, management, pricing schemes, etc., on the other side. Such links are needed in order to balance user satisfaction and economic aspects of service provisioning"*. With this description in mind we developed a questionnaire that the participants will answer after they have watched the movie (attached). The questions aim to measure user satisfaction through economical aspects, as described in the above definition.

Most of the questions uses regular cable TV as a reference point, as regular TV is generally perceived to be of very good quality and is something that the participants are used to watching and paying for. This eliminates the need to show a reference video, because the participants already knows what to expect.

3.2 Log of visible packet loss

The logging of visible packet loss is done straight forward: At least one expert will be watching the video and manually recording the time whenever he is able to see any degradations due to packet loss. He will not be answering the questionnaire.

3.3 Technical setup

The sending end is set up at the media lab Cafe Media at NTNU (Norwegian University of Science and Technology), and the receiving end is a private residence a little more than a kilometer away. Here, a room able to fit 8 persons will be furnished as a living room with a TV and speakers. We aim to make the test as close to a normal "movie night" as possible. A long HDMI and sound cable will go into another room where the technical equipment for the receiving side of the system will be.

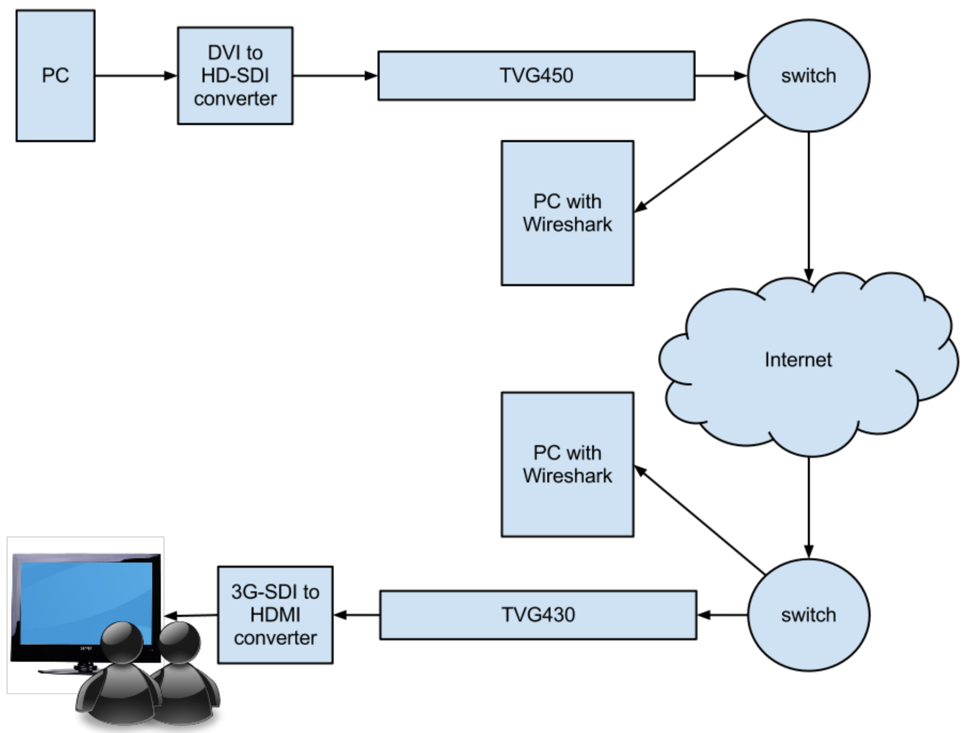


Figure 2: This figure shows how the test was set up over the Internet.

Figure 2 shows the technical setup for the test, and contains the following components:

- **PC with video material:** A computer capable of mounting a secondary screen and playing a BlueRay ripped video. We will be showing the same movie, *American Beauty*, four times. They are H.264 encoded at about 30 MBit/s with no noticeable encoding artifacts.
- **DVI to HD-SDI converter:** A *Gefen DVI to HD-SDI Scaler* box is mounted as a second screen on the computer. The box take in a 720p,

60 fps video signal on the DVI interface and output a video signal with the same specifications on the HD-SDI output.

- **Sending video gateway, TVG450:** One of the TVG450 video gateways produced by T-VIPS. This gateway is set to send at a constant bitrate of 70 MBit/s. This is high enough to not produce any noticeable encoding artifacts. Forward Error Correction is disabled.

- **Two switches:** The sending video gateway is connected to the Internet through a switch, as is the receiving gateway. The two switches is of the type *DLINK 16-Port Layer2 EasySmart Switch*. These switches are able to do *port mirroring*, which means that the packets going to a specific interface can be sent to another interface as well. This makes it possible for a computer running Wireshark to monitor and log the network traffic going in and out of the gateways. This is illustrated in figure 3.

- **Two PCs with Wireshark:** A regular computer with the network protocol analyzer Wireshark running. Wireshark is set up to filter out packets to and from the ip address and port of interest, so that only the packets containing video is tracked. The PC on the receiving end will also be used to playback the sound. This is synced up manually as soon as the video starts.

- **Internet:** A connection between the media lab Cafe Media at NTNU and a residential address a little over 1 kilometers away. The residential address has a 70 MBit/s (download, 10 MBit/s upload) Internet connection through the ISP Canal Digital. There are a little more than 10 routers on the way.

- **Receiving video gateway, TVG430:** One of the TVG430 video gateways produced by T-VIPS, compatible with the TVG450 gateway.

- **3G-SDI to HDMI converter:** A *Mini Converter SDI to HDMI* produced by BlackMagicDesign.

- **TV screen:** A Samsung plasma TV capable of displaying a 720p, 60 fps video.

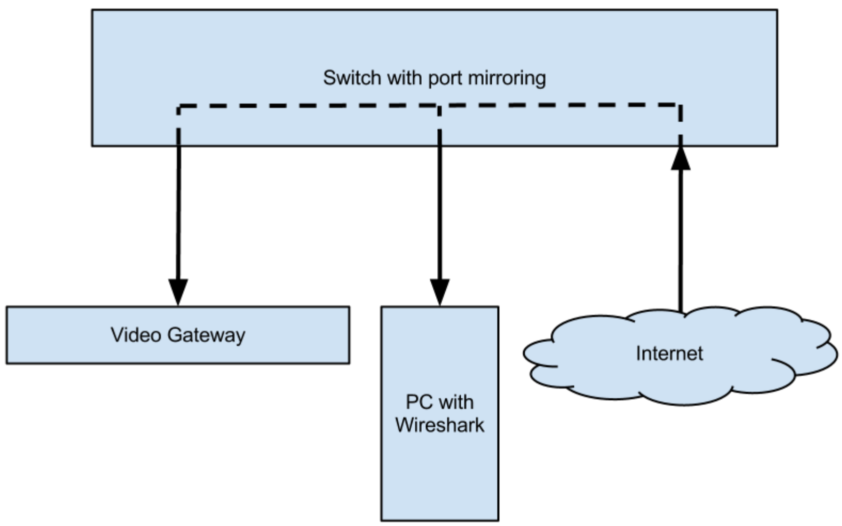


Figure 3: Port mirroring on the receiver side. The switch outputs the same packets from Internet on two interfaces.

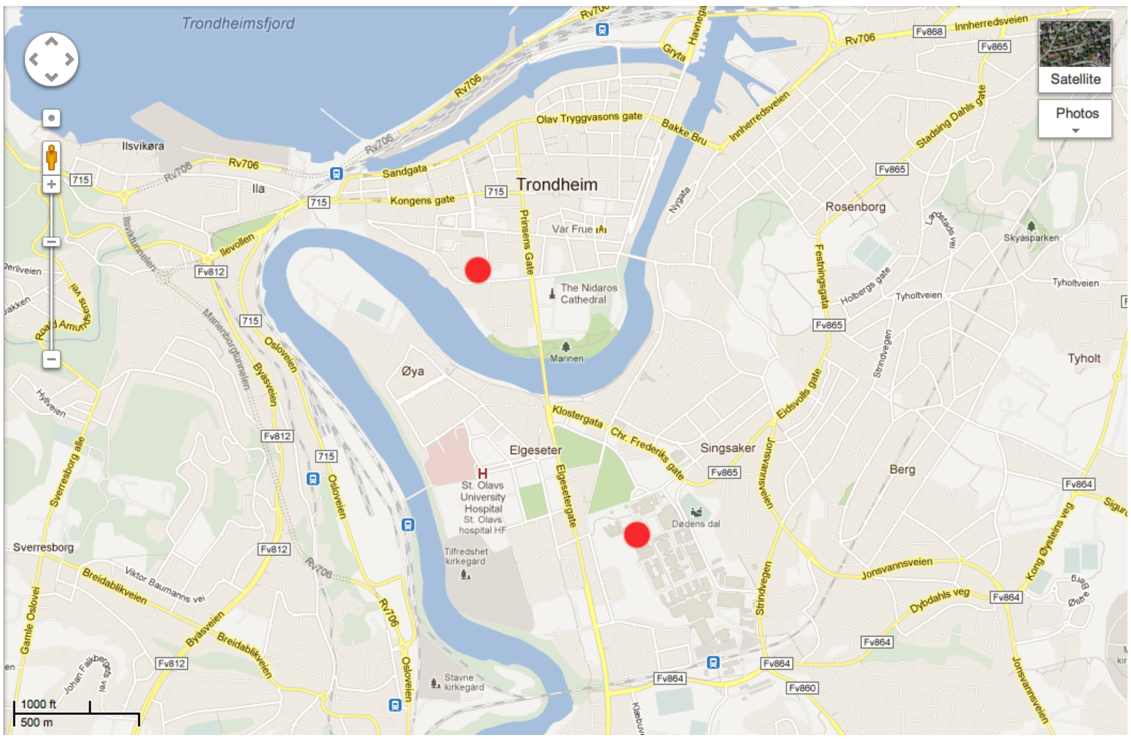


Figure 4: A map showing the position of the gateways.

4 Results, analysis and discussion

This test is conducted as part of a master thesis. The results, analysis and discussion will be presented in the thesis. A method critique will also be presented there. As this test gives a description of the experienced quality without packet loss protection, it will be a good starting point for a discussion about packet loss-protection and -handling techniques.

We use a program we have developed to analyze the network logs. The output from this program, and data from the logs of visible packet loss will be displayed in a table, exemplified in Table 4.

QoS metric	**Test 1**	**Test 2**	**Test 3**	**Test 4**
Test start	Sunday, 18:30	Monday, 18:30	Tuesday, 18:30	Thursday, 18:30
Test duration	3h 01m	3h 50m	3h 50m	3h 50m
Packets sent	83186470	106294500	83186470	83186470
Packets lost	198 (0.0002%)	340 (0.0003%)	198 (0.0002%)	198 (0.0002%)
Max packet loss burst	97 (10ms)	100 (10ms)	97 (10ms)	97 (10ms)
Avg delay (RTT)	40.7 ms	41.4 ms	40.7 ms	40.7 ms
IPDV (Jitter)	23.4 ms	30.7 ms	23.4 ms	23.4 ms
Single packet loss	3	3	3	3
Bursts of size 2-30	4	4	4	4
Bursts of size 30-100	4	4	4	4
Bursts of size >100	4	4	4	4
Visible packet loss	4	4	4	4

Table 1: This is how we plan to display the data from the network- and visible packet loss logs. The data showed in this example is mostly fictional.

This will give a good overview of the different network conditions for each test. This is important information that will be used to discuss the integrity of our results.

The questionnaire consists of 16 questions. They can be classified in four categories:

1. **VQ**: Question 5, 7 and 11 seeks to measure how they experienced video quality.
2. **WTP**: Question 3, 4, 8, 9, 12 and 15 are tied to economy, and indicates

the participants willingness to pay.

3. **XP**: Question 2, 3 and 16 gives us an indication about the participants previous experience with regular TV.

4. **Other**: Question 1, 6, 10, 13, 14 and 17 mainly serves as relevant control questions.

The data from the VQ and WTP questions will be aggregated and shown in a bar chart for each question. If some questions have unanimous answers, they will only be described by text. The answers to the XP and Other questions will be used to discuss the integrity of the VQ and WTP answers. We will check for correlations and comment anything we find interesting.

5 Priority and timetable

The experiments will be a carried out on four afternoons. Afternoons are peek hours for network traffic, which is good for our test. Each part will last as long as the movie (about 3 hours) plus 10 minutes for questioning. We, as in us two conducting the test, will serve as the experts. We will also get hold of 24 participants, and divide them into groups of 6, which is a practical upper limit to our test. The participants should be rewarded one movie ticket for their effort. Except for this, we have all the equipment we need.

References

[Fiedler et al., 2010] Fiedler, M., Hlavacs, H., Hackbarth, K. and Arlos, P. (2010). Quality of experience. Annals of Telecommunications *65*, 1–2.

[Helge Stephansen, 2007] Helge Stephansen, T. E. K. (2007). Experience with SDI Contribution Experience with SDI Contribution Experience with SDI Contribution over IP Network. SMPTE Mot. Imag J. *116*, 24–28.

Bibliography

[1] Interview with Steinar Bjørlykke. Chief of technology, NRK.

[2] Kjell Brunnström, Katrien De Moor, Ann Dooms, Sebastian Egger, Marie-Neige Garcia, Tobias Hoßfeld, Satu Jumisko-Pyykkö, Christian Keimel, Chaker Larabi, Bob Lawlor, Patrick Le Callet, Sebastian Möller, Fernando Pereira, Manuela Pereira, Andrew Perkis, António Pinheiro, Alexander Raake, Peter Reichl, Ulrich Reiter, Raimund Schatz, Peter Schelkens, Lea Skorin-Kapov, Dominik Strohmeier, Christian Timmerer, Martin Varela, Junyong You, and Andrej Zgank. Qualinet white paper on definitions of quality of experience. Input version for the Dagstuhl seminar 12181, May 2012.

[3] ITU G.1050. "series g: Transmission systems and media, digital systems and networks: Quality of service and performance - generic and user related aspects. 2007.

[4] Helge Stephansen. Quality of service in ip networks for video contribution and distribution. http://www.t-vips.com/sites/default/files/whitepapers/Whitepaper_QoS.pdf. Accessed 07/05/2012.

[5] M. Clark. *Data networks, IP and the Internet*. Wiley, 2003.

[6] T-VIPS. Case studies. http://www.t-vips.com/case-studies. Accessed 05/06/2012.

[7] Interview with Lasse Wiik. Chief of technology, media netwerk A/S.

[8] Interview with Øyvind Vikanes. Engineer at t-vips, norway.

[9] Martin Markman. Introductional paper on reliable broadcast contribution over the public internet. Department of Electronics and Telecommunications, Norwegian University of Science and Technology (NTNU) 2011.

[10] Cisco. The connected life - bejing olympics. http://www.cisco.com/en/US/solutions/ns341/ns525/ns537/ns705/ Accessed 25/05/2012.

[11] Cisco. Cisco powers nbc's coverage of vancouver olympic winter games. http://www.cisco.com/en/US/solutions/ns341/ns898/nbc_2010_olympic_winter_games.html. Accessed 25/05/2012.

[12] Interview with Steinar Bjørlykke. Contribution expert.

[13] SSB.no. Ikt i husholdningene 2. kvartal 2011. http://www.ssb.no/ikthus/. Accessed 07.05.2012.

[14] Paul Read and Mark-Paul Meyer. *Restoration of Motion Picture Film*. Butterworth-Heinemann Ltd, 1st edition, 2000.

[15] Gary J. Sullivan and Thomas Wiegand. Video compression – from concepts to the h.264/avc standard. In *PROCEEDINGS OF THE IEEE*, pages 18–31, 2005.

[16] J.L. Mitchell W.B. Pennebaker. The jpeg still image data compression standard. *Van Nostrand Reinhold*, 1993.

[17] Wikipedia. Inter frame. http://en.wikipedia.org/wiki/Inter_frame.

[18] Teknisk Ukeblad. Slik fraktes tv-bildene hjem i stua. http://www.tu.no/forbruker/2012/04/29/slik-fraktes-tv-bildene-hjem-i-stua . Accessed 05/06/2012.

[19] International Organization for Standardization. Iso/iec 14496-10:2010. http://www.iso.org/iso/iso_catalogue/catalogue_ics/catalogue_detail_ics.htm?csnumber=56538 . Accessed 05/06/2012.

[20] D. Marpe, V. George, H.L. Cycon, and K.-U. Barthel. Performance evaluation of motion-jpeg2000 in comparison with h.264/avc operated in pure intra coding mode. 2004.

[21] Digital Cinema Initiatives. Dci website. http://www.dcimovies.com/ . Accessed 05/06/2012.

[22] A. Bilgin and M.W. Marcellin. Jpeg2000 for digital cinema. In *Circuits and Systems, 2006. ISCAS 2006. Proceedings. 2006 IEEE International Symposium on*, pages 4 pp. –3881, may 2006.

[23] Dvd - mpeg differences. http://dvd.sourceforge.net/dvdinfo/dvdmpeg.html . Accessed 05/06/2012.

[24] Barry G. Haskell, Atul Puri, and Arun N. Netravali. *Digital Video: An Introduction to Mpeg-2*. Springer, 1997.

[25] Wikipedia. Sdi article. http://en.wikipedia.org/wiki/Serial_digital_interface . Accessed 05/06/2012.

[26] Mathias Gjerstad Lervold. Measuring perceptual quality in internet television. Master's thesis, Norwegian University of Science and Technology, June 2009.

[27] Introduction to the OSI model. http://www.washington.edu/lst/help/computing_fundamentals/networking/osi. Accessed 06/06/2012.

[28] Dapeng Wu, Yiwei Thomas Hou, and Ya-Qin Zhang. Transporting real-time video over the internet: Challenges and approaches. *Proceedings of the IEEE*, 88, Issue 12:1855–1877, 2000.

[29] DVB CM-AVC (Audio Visual Codecs). High level contribution codec commercial requirements. 2011.

[30] F. Boavida and E. Cerqueira. Benchmarking the quality of experience of video streaming and multimedia search services: the content network of excellence. *Telecommunication Review and Telecommunication News Journal, Technical Journals and Books Publisher SIGMA NOT Ltd*, December 2008.

[31] Tom Leighton. Improving performance on the internet. *Communications of the ACM*, 52, 2009.

[32] Interview with Ove Tøien. Ip engineer, Telenor.

[33] UiO. Norwegian internet exchange (nix) website. http://www.uio.no/tjenester/it/nett/fastnett/nix/. Accessed 10/05/12.

[34] Caida.org. Passive internet monitoring (usa). http://www.caida.org/data/realtime/passive/. Accessed 10/05/2012.

[35] Interview with Haakon Smeplass. Sit it middleware and content services, Telenor.

[36] Cisco. Cisco visual networking index: Forecast and methodology, 2010 - 2015. 2011. Whitepaper.

[37] Sandvine. Global internet phenomena report. Spring 2011.

[38] Fiber Optics LAN Section of the Telecommunications Industry Association (TIA). Fiber in the horizontal: The better way to carry information, 2000.

[39] SSB.no. Ikt i husholdningene 2. kvartal 2009. http://www.ssb.no/emner/10/03/ikthus/arkiv/tab-2009-09-24-02.html. Accessed 10/05/2012.

[40] Post og Teletilsynet. Bredt nok - kapasitetsbehov og utviklingstrender innen bredbåndskommunikasjon. 2011.

[41] teknofil.no. Bredbandsguiden. http://speedometer.teknofil.no/providers.php. Accessed 10/05/2012.

[42] Cisco. Cisco visual networking index: Forecast and methodology, 2007 - 2012.

[43] David Olshefski Weibin Zhao and Henning Schulzrinne. Internet quality of service: an overview. *Columbia University*, 2003.

[44] Vern Paxson. End-end internet packet dynamics. *Network Research Group*, University of California, Berkeley, 1999.

[45] Hung Nguyen and Matthew Roughan. On the correlation of internet packet losses. *School of Mathematical Sciences, Adelaide University, Australia*, 2002.

[46] Pro-MPEG Forum. Code of practice #3. Release 2, 2004.

[47] Wireshark.org. http://www.wireshark.org/ . Accessed 06/06/2012.

[48] Magnus Jeffs Tovsliv. Jpeg2000 quality scalability in an ip networking scenario. Master's thesis, Norwegian University of Science and Technology (NTNU), 2012.

[49] Markus Fiedler, Helmut Hlavacs, Klaus Hackbarth, and Patrik Arlos. Quality of experience. *Annals of Telecommunications*, 65:1–2, 2010. 10.1007/s12243-009-0158-2.

[50] K.R. Larsen, D. Nevo, and E. Rich. Exploring the semantic validity of questionnaire scales. In *Hawaii International Conference on System Sciences, Proceedings of the 41st Annual*, page 440, jan. 2008.

[51] Steinar Ilstad. *Generell Psykologi*. Tapir akademiske forlag, 7 edition, 2007.