Master Thesis
Electrical Engineering
Thesis no: 100015
December 2014

Video Quality of Experience through Emulated Mobile Channels

NARAYANA KAILASH AMUJALA
JOHN KENNEDY SANKI

Dept. of Communication Systems
Blekinge Institute of Technology
37179 Karlskrona
Sweden

This thesis is submitted to the Dept. of Communication Systems at Blekinge Institute of Technology in partial fulfilment of the requirements for the degree of Master of Science in Electrical Engineering. The thesis is equivalent to 20 weeks of full time studies.

This Master Thesis is typeset using LaTeX

Contact Information

Author(s):

Narayana Kailash Amujala
Address: Hyderabad, India
E-mail: narayana.kailash@gmail.com

John Kennedy Sanki
Address: Karlskrona, Sweden
E-mail: jasa11@student.bth.se

University advisor(s):

Prof. Dr.-Ing. Markus Fiedler

Dept. of Communication Systems
Blekinge Institute of Technology
371 79 KARLSKRONA SWEDEN

Internet: www.bth.se/diko
Phone: +46 455 385000
SWEDEN

Abstract

Over the past few years, Internet traffic took a ramp increase. Of which, most of the traffic is video traffic. With the latest Cisco forecast it is estimated that, by 2017 online video will be highly adopted service with large customer base. As the networks are being increasingly ubiquitous, applications are turning equally intelligent. A typical video communication chain involves transmission of encoded raw video frames with subsequent decoding at the receiver side. One such intelligent codec that is gaining large research attention is H.264/SVC, which can adapt dynamically to the end device configurations and network conditions.

With such a bandwidth hungry, video communications running over lossy mobile networks, its extremely important to quantify the end user acceptability. This work primarily investigates the problems at player user interface level compared to the physical layer disturbances. We have chosen Inter frame time at the Application layer level to quantify the user experience (player UI) for varying lower layer metrics like noise and link power with nice demonstrator telling cases.

The results show that extreme noise and low link level settings have adverse effect on user experience in temporal dimension. The video are effected with frequent jumps and freezes.

Keywords: H.264/SVC, AWGN noise, Downlink power, QoE problems, User Interface.

Acknowledgements

Studying in this Double Degree Masters Program has been wonderful! We got an opportunity to study in two different Universities i.e. Jawaharlal Nehru Technological University, Hyderabad, India, Blekinge Institute of Technology, Karlskrona, Sweden. It has been an international experience; initial semester was studied at JNTUH, India and the remaining semesters were at BTH, Sweden. Finally the thesis defense took place at BTH-Sweden. Throughout this journey, we had received unmatched support of our Professors at India and Sweden without whom this Masters and Thesis would not have been possible.

We are indebted to many people for making this experience an unforgettable one! First and Foremost, we would like to express our sincere gratitude to Prof. Dr. Ing. Markus Fiedler who motivated and supported us from the outset to the final stage. He patiently cleared all our doubts raised during our work and encouraged us in the times of difficulties. We have learned a lot under his expert guidance in the scientific field.

Furthermore, we gratefully acknowledge Dr. Patrik Arlos, Dr. Imran Iqbal, Mr. Tahir Nawaz and Mr. Selim Ickin for their continuous advice and encouragement throughout the course and thesis. Our numerous scientific discussions helped us in fetching knowledge.

Also, we owe our special appreciations to Friends and Dear Ones for their support and all the fun we have had during our stay in Sweden.

No words can express our gratitude towards our Parents, Brothers and Sister for their continuous support and strength they gave us throughout our life and guided towards the right trajectory.

Narayana Kailash Amujala
John Kennedy Sanki
Karlskrona

Dedicated to our families and friends

Contents

List of Figures

List of Tables

Acronyms

3GPP 3rd Generation Partnership Project

ACLR Adjacent Channel Leakage Ratio

ACLR Adjacent Channel Leakage Power Ratio

AMD Advanced Micro Devices

AMPS Advanced Mobile Phone System

ASF Advanced Systems Format

AVC Advanced Video Coding

AVI Audio Video Interleaved

BLER Block Error Rate

BTH Blekinge Tekniska Högskolan

CCDF Complementary Cumulative Distribution Function

CDMA Code Division Multiple Access

CPU Central Processing Unit

DDR Double Data Rate

DVB Digital Video Broadcasting

DVD Digital Versatile Disc

EDGE Enhanced Data for Global Evolution

FDD Frequency-Division Duplexing

GPRS General Packet Radio Service

GSM Global System for Mobile Communications

GSM-R Global System for Mobile Communications-Railway

HD High Definition

HDTV High Definition Television

HSDPA High-Speed Downlink Packet Access

HSPA High Speed Packet Access

HTTP Hypertext Transfer Protocol

ICS Internet Connection Sharing

IEEE Institute of Electrical and Electronics Engineers

IP Internet Protocol

IPTV Internet Protocol Television

ITU-T International Telecommunication Union-Telecommunication

JNTUH Jawaharlal Nehru Technological University Hyderabad

MOS Mean Opinion Score

MPEG Moving Picture Experts Group

NAT Network Address Translations

OBW Occupied Bandwidth

PEVQ Perceptual Evaluation of Video Quality

QoE Quality of Experience

QoS Quality of Service

RF Radio Frequency

RTP Real-time Transport Protocol

RTSP Real Time Streaming Protocol

RX Receiver

SDRAM Synchronous Dynamic Random-Access Memory

SNR Signal to Noise Ratio

SVC Scalable Video Coding

SVCD Super Video Compact Disc

TCP Transmission Control Protocol

TX Transmitter

UDP User Datagram Protocol

UE User Equipment

UI User Interface

URCT Universal Radio Communication Tester

VOB Video Object

VoIP Voice over Internet Protocol

WCDMA Wideband Code Division Multiple Access

WMA Windows Media Audio

WMV Windows Media Video

Chapter 1

Introduction

In Next Generation Networks, Video streaming has a pivotal role. According to the latest statistics, nearly 90% of internet traffic [1] will be dominated by the video streaming applications. In recent years, there has been tremendous growth in the fields of video streaming - development and research areas. So it has gained lot of interest by the public in various fields like communicating, telecasting, surfing, video conferencing etc. Latest developments in compression technology, computing technology and high-speed networks have made it viable to provide real-time multimedia services over the Internet. Real-time transport of stored video and live video is the principal part of real-time multimedia services. The combination of various types of media content like still images, video, audio, animation, text and interactivity form a multimedia. The list of multimedia applications includes Internet Protocol Television (IPTV), video conferencing, Voice over Internet Protocol (VoIP), video streaming, video broadcasting etc. Real-time Transport Protocol (RTP), User Datagram Protocol (UDP), Transmission Control Protocol (TCP) and Real Time Streaming Protocol (RTSP) are widely used protocols which run on top of IP Networks for streaming of visual media over the Internet.

Providing access of video services across wireless communication remains a challenging task due to high error rate, time-varying bandwidth, and limited resources on mobile hosts. The decoding failure at the receiver end can be caused by transmission errors. More prominently, the transmission errors presented in one video frame will broadcast to its subsequent frames along the motion prediction path and significantly degrade video presentation quality.

In our scenario the main problem relates to video transmission or communication over mobile networks. The poor quality of video transmission is caused due to the network congestion which leads to delay, delay variations or packet loss [2]. In this study, the point of discussion is the quality of the video degraded due to the freezes appeared in the video [3].

The research is carried especially on these interesting cases with the use of demonstrator. The demonstrator consist of a Tester (i.e. CMU 200), Server and Client machine. To produce the radio-level disturbances and to know the impact it creates on video streaming, we use a Tester which mimics the real network. Through this demonstrator setup, we can analyze the disturbed videos and develop the instructive demo cases for demonstration. This demonstrator can be used for educational purpose, researchers, students, visitors and to make a meaningful progress towards the research.

1.1 Objectives

The aim is to analyze and demonstrate the effect of SNR on frame rates and user perception of streaming video when passed through a mobile network.

To provide an overview of published results and demonstrations regarding disturbances in real network and their impact on video.

To perform analysis on frame timing information at the buffer of the player.

1.2 Research Questions

RQ 1: How does the radio-level disturbances affects the instantaneous frame rates of an SVC video stream ?

RQ 2: How do irregularities in inter-frame times affect the QoE ?

RQ 3: Which are the telling use cases for demonstrating the effect of link-level disturbances on QoE of SVC-coded video ?

1.3 Research Methodology

Firstly, a literature study is made on disturbances in the network and its effect on frame rates, video quality and QoE, as well as on related demonstration experiments.

SVC coded videos are to be produced and selected with different codec parameters.

Noise levels are introduced into the network with the Universal Radio Communication Tester (URCT) CMU 200.

Once the videos are streamed through disturbances in the network, the time stamps of network traffic and displayed frames need to be collected.

The timestamps used to be analyzed and relation between frame rates, noise levels and signal strength used to be established.

1.4 Thesis Outline

This report is organized as follows:

Chapter 1 provides introduction to the thesis work.

Chapter 2 describes background work and research related to this thesis work.

Chapter 3 describes the experimental setup.

Chapter 4 contains results and discussion.

Chapter 5 describes conclusion and future work.

Chapter 2

Background

In this chapter, we discuss the key concepts and the related research work. Real-time transport of stored video and live video is the major part of real-time multimedia services.

The 3rd Generation Partnership Project (3GPP) defines the multimedia streaming as the ability of an application to play synchronized media streams like audio and video in a continuous way, while those streams are being transmitted to the client over the networks [4].

In our thesis, we are concerned with video streaming, which refers to real-time transmission of live and stored video. There are two approaches for transmission of video stored over the Internet, namely the streaming mode and the download mode (i.e., video streaming). In the download mode, a customer downloads the unbroken video file and then plays back the video file. However, full file transmission in the download mode frequently suffers long and possibly intolerable transfer time. In contrast, in the streaming mode, the video content need not be downloaded in full, but is being played out while parts of the content are being received and decoded. Due to its real-time environment, typically video streaming has delay, loss requirements and bandwidth. Nevertheless, the modern best-effort Internet does not compromise any quality of service (QoS) assurances to streaming video over the Internet such as minimum packet loss rate, bandwidth and delay which are dangerous to many multimedia applications.

2.1 Video Streaming System - Overview

In video streaming applications over internet, there is a need of continuous stream of video data sent from the source to the receiver [5] [6]. While the parts of video are being transmitted, the receiving part of the video file can be played at the end user. A typical architecture of video streaming system over the Internet is shown in Figure 2.1, which consists of a streaming server with videos stored in it, the network in between and end-users devices that

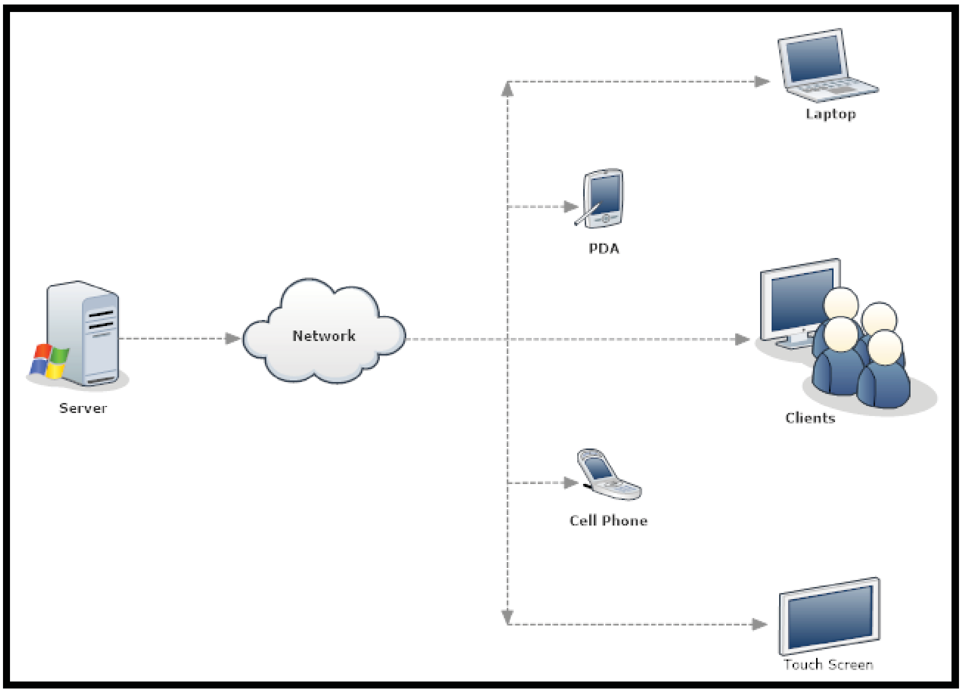

Figure 2.1: A typical architecture of video streaming system

receive the video.

When the client requests for specific video from server, a streaming server allocates resources, retrieves video data from storage devices and then the application-layer QoS control module adapts to the video bit streams according to the network status and QoS requirements. The streaming servers are required to process video data under timing constraints and support interactive control operations such as pause/ resume, fast forward and rewind, the transport protocols packet size the bit streams and send the video packet over the Internet. The streaming server and the relay server are generally responsible for matching the output video stream to the available channel resources and ultimately the clients device capabilities [7].

The arriving packets at the client are de-capsulated into media information and passed to the application for playback. During transmission the video packets may be dropped or experience excessive delay on the network because of network congestions or link failure. Considering the best-effort service of the Internet, the streaming servers together with client devices may use the received video to analyze the network condition and feedback information for adapting QoS requirements.

2.2 Video Streaming Protocols - Overview

Video streaming requires a steady flow of information and delivery of packets by a deadline from the source to destination [6]. To achieve this requirement, streaming protocols provide data transmission, network addressing and a negotiation service between the server and the client. Protocols that are relevant to video streaming can be classified into three categories; network layer protocol, transport protocol and session control protocol.

Network-layer protocol, being served by Internet Protocol (IP), provides basic network service like network address resolution and generally responsible for the transmission of TCP and UDP packets to the end users. Transport layer protocols, for instance User Datagram Protocol (UDP), Transmission Control Protocol (TCP), and Real-time Transport Protocol (RTP), Real-time Transmission control Protocol (RTCP) provide end-to-end transport functions for data transmission [6]. Session control protocols such as Real-time Streaming Protocol (RTSP) create and maintain session between source and destination applications. The session control protocols handles the exchange of information to initiate, keep active, and to restart the sessions that are disrupted or been idle for a long period of time [8].

The Transmission Control Protocol or TCP is a byte stream, connection-oriented and reliable delivery transport layer protocol [9]. It is a byte stream since the application that uses TCP is unaware of the data segmentation performed by TCP. TCP segments the byte stream in order to transmit the appropriate format to the receiver. TCP is reliable since it contains mechanisms such as checksums, duplicate data detection, retransmissions, sequencing and timers.

TCP uses a three way handshaking between the sender and receiver in order to facilitate the feature of a connection-oriented protocol. The delivery of packet between devices is guaranteed by triggering retransmission until the data is correctly and completely received. The overhead of certifying dependable data transfer reduces the complete transmission rate and increases the latency when streaming video. In addition, TCP also includes a congestion-control mechanism that adjusts the transmission rate by limiting each TCP connection to its fair share of network bandwidth when the network is congested between sender and receiver. The congestion control may also have a very harmful effect on real-time video streaming applications [10].

TCP has several limitations for real-time video data transmission; TCP does not guarantee a minimum transmission rate. In particular, the sender is not permitted to transmit at a maximum rate; instead the sending rate is regulated by TCP congestion control, which may force the sender to send at a low average rate [9]. TCP does not provide any delay guarantees. In particular, when a sender transmit data, the data will eventually arrive at the receiver, but TCP guarantees absolutely no limit on how long time the data may take [11].

UDP is a lightweight transport protocol with a minimalist service model that runs on top of IP networks. UDP is commonly used for real-time video streaming due to the datagram service which emphasizes on reduced latency over reliability. Unlike TCP, UDP does not provide reliability. However, UDP transmission is mostly blocked by firewalls or Network Address Translations (NATs). Most of the leading video service providers run their videos over Hyper Text Transfer Protocol (HTTP), which uses Transmission Control Protocol (TCP) [12].

2.3 Video Codes SVC

In modern era, amount of video data which is to be transmitted to communication channel is rapidly increasing. In order to reduce the size of the video data in an efficient way such that the quality is maintained at the receiver end and optimizing the resource allocation resulting a technology known as Video Coding. The inclusion of different video coding technologies evolved with video compression including H.261, H.262, H.263 and Motion Picture Expert Group (MPEG-2) [11]. In heterogeneous network scenario, to meet the demands in Quality of Service (QoS) is very essential for a better compression technique. Special Codec or Tools designed to meet the required Quality of Service (QoS) in order to adopt the robustness of the channel used of transmitting. In order to produce superior video quality, algorithms designed for specific codes have to face challenges with in the network disturbances [13].

H.264/AVC (Advanced Video Coding) is the recent video coding technology that has been approved by International Telecommunication Union-Telecommunications (ITU-T) as the standard video coding standard for transmitting videos over satellite or cable [11]. It is used for most common video applications ranging from mobile services and video conferencing to HDTV, IPTV and HD video storage [14] has been employed recently to optimize the encoding parameters for motion compensation and also deliver acceptable video quality at substantially lower bit rates [15]. It exploits tradeoffs between the cost and quality to achieve a good compression ratio.

The Scalable Video Codec is a recent enhancement of ITU-T [16] extension to the H.264 standard, designed to deliver the benefits described in the anteceding ideal scenario. It is based on the H.264 Advanced Video Codec standard (H.264 AVC) and heavily leverages the tools and conceptions of the original codec [17]. The scalable video coding structure allows the video stream to be split into a combination of spatial, temporal, and quality layers. A base layer encodes the last temporal, spatial, and quality representation of the video streaming. Enhancement layers convert additional information that, using the base layer as a beginning point, can be used to reconstruct higher quality, resolution, or temporal versions of the video during the decoding process. By decoding the base layer and only the resultant enhancement layers required, a decoder can produce a video stream with certain desired characteristics [18] [19].

The values 040, 140, 240 and 241 represent the identifiers for Scalable Video Coding (H264/SVC). 040 represent the Base Layer, 140 represent the Enhancement layer and 241 represents the Enhancement layer with one additional quality layer.

The index follows the DTQ notation, which represents the spatial, temporal and quality identifiers respectively for a specific bit stream. D stands for Dependency ID, T stands for the temporal ID, and Q for Quality ID. 040 contain bit-stream corresponding to spatial, temporal and quality levels equal to 0, 4 and 0 respectively. Similarly, 140 stand for spatial level of 1, temporal level of 4, and quality level of 0 and 240 stands for spatial level of 2, temporal level of 4 and quality level of 0. 241 indicate the highest quality level by having a value of 1 at the quality identifier. The higher the value of the index, (D, T or Q), the more detailed information is present [13].

2.4 Related Work

With the volatile development of video applications over the Internet, many methodologies have been recommended to stream the video successfully over best-effort networks and packet switched [20]. Several use techniques to modify system architectures, or from channel coding and source coding, or implement transport protocols in order to deal with loss, delay, and time-varying environment of the Internet.

The rapid increase in usage of Internet [3], has made many problems based on the network traffic congestion to appear as Real time data traffic is very much sensitive to losses, delays and bandwidth limitations.

Video streaming over internet can have problems from the disturbances in the real network, such as packet loss, delay and delay variations, which affect the quality of a video at the receiver end [3] [2]. If the delay is longer, the packets get queued in the network for longer times. This will result into emptied jitter buffer at the player, causing freezes at the application layer. This affects the end user experience which can lead to user churn [7].

Since the real network has disturbances, it will be of interest to see what impact it has on video streaming. In Reference [21], the authors analyze the effects of noise at physical layer related to the application layer of video streaming for Wi-Fi technology. The ultimate impact of packet loss on steaming video is the decrease of intact playback frames of video [22]. So the study on frame rates and how it is affected in different scenarios will be an interesting topic to be researched upon [23].

The receiver buffer requirement for video streaming over TCP is determined by the network characteristics and desired video quality is explained in [24] [25]. In real-time streaming, if the player buffer gets empty the end user will face freeze. In mobile video streaming, it is more probable to have video freezes, but in real-time streaming with freeze, the content of video can be lost during the frozen interval. It is observed that users are pickier for such kind of artifact as compared to the PEVQ rating. Furthermore, we establish that the location of disturbance of video disturbs the customer assessment [8]. The real-time adaptation of scalable video content on a network has been demonstrated with client-server model in [26]. This would be interesting if we demonstrate the artifacts and analyze for further research.

Each video frame is transmitted in the burst of packets that is potentially queued at the access point. The burstiness of the video is due to frame based nature of the encoded video and exhibits saw-tooth-like delay [26].

Quality of Experience (QoE) is the key criteria for evaluating the Media Services. In recent times, QoE got increasingly much attention from service providers, operators, manufacturers and researchers. As wireless networks have been progressively deployed, the necessity of quality measurement became crucial since network operators want to control their network resources though retaining customer satisfaction. Further significantly, measurement of technical constraints fails to provide an account of the customer experience, what could be named QoE [9]. Describing the conceptual model of QoE in parallel to the classical internet architecture hourglass model is presented in [27]. It discusses the different factors affecting QoE from IP layer to application layer.

Chapter 3

Implementation

3.1 Experimental Setup

The experimental setup is composed of three main parts; a server machine, a client computer and the base station i.e. CMU 200 as shown in Figure 3.1. In this experiment, streaming is done from server machine to client machine. Here, server and client machines are running on Linux operating system.

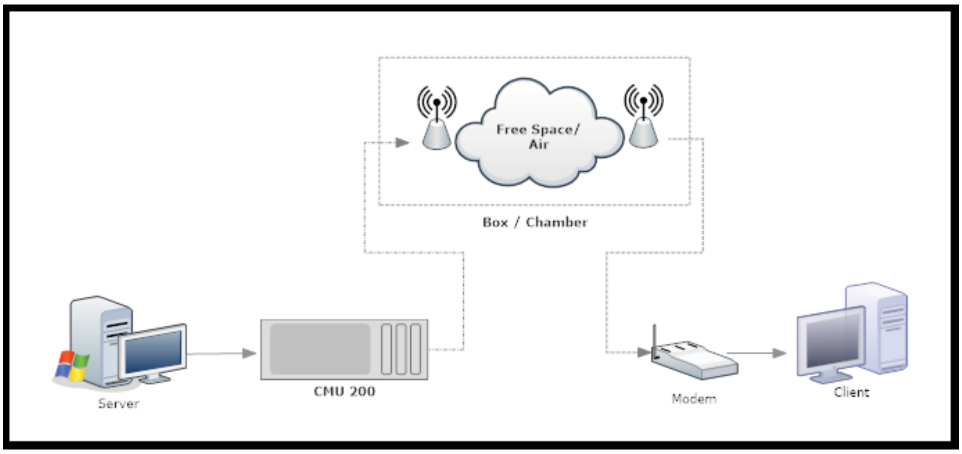

Figure 3.1: Experimental Setup

In this scenario Host A i.e. Server sends packets to Host B i.e. Client via Emulator i.e. CMU 200.

3.1.1 Server

Flumotion is an open source streaming media server. It allows content delivery to the devices like browsers, players, media centers, mobile devices and game stations. It supports all necessary audio and video codecs like SVC, H.264, WMV, etc. The technology used in the Flumotion streaming server gives quality, performance, unprecedented stability and scalability in offering high-quality streaming media delivery. The formats supported for on demand streaming are WebM, MP3, MP4, MOV, 3GPP etc. A Flumotion system consists of several processes working together, with the worker creating process for the components, and the manager telling the worker what to do. The Flumotion user interface connects to the manger which in turns controls the workers and tells it when to start and stop a system. So in order to maintain the connection between the manager and the worker, xml files needs to be placed in the Flumotion directory [28].

In our thesis, the Flumotion 0.8.1 streaming server was used, which support on demand streaming for the codecs like SVC, H.264 and Google codec. The server was a HP desktop with the AMD Phenom 2 CPU Q720 @ 1.60 GHz 1.60 Hz, 4096 MB DDR3 SDRAM, running on a Ubuntu 10.10 with LTS Linux kernel version 2.6.35.4.

3.1.2 Client

The client machine is operating on Linux platform. The client machine receives the commands form the server via CMU 200. These videos are played with MPlayer. MPlayer is a video player that runs on popular operating systems. It is open source software for several UNIX platforms including like Linux, Solaris, MacOS 10 & BSDs. Presently, MPlayer supports all players in Linux platform and almost all video formats including MPEG1 (VCD), MPEG2 (DVD/DVB/SVCD), DivX 3/4/5, XviD, AVI, MPEG/VOB, ASF/WMA/WMV etc. MPlayer has finer stability & fault tolerance comparing to other original players of these similar formats. Supporting wide output devices is an additional advantage of MPlayer [29] [30].

3.1.3 CMU 200

For testing the applications which are used in mobile phones, a public mobile radio network or the simulation of such a network is required. Earlier, radio networks are simulated with the help of complex setups [31]. Now this is simplified by the interesting equipment Rohde & Schwarz (R&S) CMU 200 Universal Radio Communication Tester. The CMU 200 has control buttons such as Radio Frequency Channel downlink and uplink, downlink power, UE power control, band select etc., which can initiate and release different connection types [32]. It can be tuned for application tests with specifications like WCDMA, HSDPA and CDMA2000. The non-signaling mode in CMU 200 is for generating and analyzing WCDMA signals in full frequency range. It provides specific transmission measurements on signals such as ACLR (adjacent channel leakage power ratio), OBW (occupied bandwidth), Modulation, Power (max, min, off) etc. The non-signaling mode allows tests of all the essential RF parameters of the connected user equipment.

The impact of radio-level disturbances on video QoE and user accepted levels of video quality are the main concerns of this research. Currently, the research on radio-level disturbances and frame rates is not adequate. It has to be investigated and our research is carried especially on these interesting cases with the use of demonstrator. The demonstrator consists of a Tester (i.e. CMU 200), Server and Client machine. To produce the radio-level disturbances and to know the impact it creates on video streaming, we use a Tester which mimics the real network. Through this demonstrator setup, we can analyze the disturbed videos and develop the instructive demo cases for demonstration. This demonstrator can be used for educational purpose, researchers, students, visitors and to make a meaningful progress towards the research.

CMU 200 supports the following technologies:

3GPP WCDMA (FDD)

3GPP HSPA

GSM/GPRS/EDGE

GSM-R

AMPS and IS-136

CDMA2000 1xRTT

CDMA2000 1xEV-DO

IEEE 802.15.1 Bluetooth

3.2 3GPP WCDMA

Transmitter Characteristics with respect to WCDMA (Wideband Code Division Multiple Access) technology are studied.

3.2.1 Packet Data - Data rate

Data rate for the packet data connection (initiated from the UE). The R&S CMU supports symmetric 64 kbps connections and faster, asymmetric 384 kbps Downlink / 64 kbps Uplink connections [32].

3.2.2 SNR

Another measure of link quality is the Signal-to-Noise-Ratio (SNR), which is defined as the ratio between the power of a signal and the power of corrupting noise. Because the range of the signals can be very dynamic, it is common to express the SNR in terms of a logarithmic decibel scale.

$$SNR = 10 \times \log 10(P_{signal}/P_{noise})$$

3.3 WCDMA

WCDMA Application Testing (option R&S CMU-K92) allows the R&S CMU200 to be integrated in a TCP/IP network in order to test and monitor IP-based, packet switched data applications that a WCDMA UE services under realistic operating conditions. The option is used together with the WCDMA UE test options of the CMU200 Universal Radio Communication Tester [32].

The software provides a special RX measurement (Receiver Quality RLC BLER) which evaluates the downlink BLER and the data throughput for the tested applications. Moreover, the R&S CMU can perform all TX and RX tests (except RX tests relying on a special RMC) while a packet data application is running [32].

3.4 Procedure

The experimental setup consists of a video streaming server, video player at client, CMU200. The Streaming server namely Flumotion streaming Server is used to send the encoded video sequences to the client. Streamer and player were installed on Linux Ubuntu 10.10 platform and these are connected with Ethernet cables. The client is connected to the base station using a windows machine as gateway. The reason for using the windows machine as gateway is twofold; firstly, there are no exact drivers for the modem in Linux environment. Secondly, the custom compiled svc player is performed in Linux environment. The base station used is a Rohde and Schwarz company CMU200 tuned to perform in application test environment (ATE) mode with WCDMA specifications. The down-link speed is made to 3.6 Mbps and uplink speed is set to 384 Kbps. The gateway is set to make network sharing to the client machine (c.f., ICS enable in windows XP). Thus the whole test-bed is shared with the client machine. IP address of the machines involved: As the IP address is made static in the server machine.

Server IP: 192.168.168.169

CMU IP: 192.168.168.170

Gateway IP: 192.168.0.20

Client IP: 192.168.0.1 assigned by the gateway.

Chapter 4

Results and Analysis

The behaviour of a video to the network disturbances are studied in this chapter. The results are based on the experiments conducted by varying signal, noise powers and observing the videos at two different buffer levels i.e. 32 kB and 320 kB (default value) using the procedure which is explained in chapter 3.

For experiment, we have considered different Ducks videos with the resolution and frame rates of videos are given in below Table 4.1.

Video	Resolution	Frame rate (fps)
040	352×288	30
140	640×368	30
240	640×368	30
241	640×368	30

Table 4.1: Video Specifications

These videos have frame rate of 30 fps and varied resolutions. The videos are played at client machine and M player records the traces at the client machine. These traces are collected for the analysis.

4.1 Varying Signal Power

Disturbances are introduced by varying signal power at a fixed noise level.

The following cases considered are:
Case1: Noise power N = -70 dBm, and signal power S = {-45, -55, -65} dBm.
Case2: Noise power N = -80 dBm, and signal power S = {-45, -55, -65, -75} dBm.
Case3: Noise power N = -90 dBm, and signal power S = {-45, -55, -65, -75, -85} dBm.

Experiments are done at the above conditions and traces are collected for analysis at client for the four videos namely 040, 140, 240, 241 at above three cases (i.e. 12 scenarios). After calculating the inter frame times, at the following conditions, the CCDF graphs are plotted and analyzed.

Why traces are to be analyzed?
A well designed and optimally performing network is required to maintain user satisfaction. Network performance is evaluated through measurements and analysis. Analysis of traces plays a crucial role in understanding and analyzing the network behaviour. But to perform any functionality or calculations, basically the trace files should be user friendly and easily understandable. To analyze the network behaviour, we concentrate on the traces at receiver side.

For analysis, we calculate the Inter-frame time (IFT):
IFT is the time duration between the arrivals of successive frames. Analysis of IFT helps to differentiate between smooth and distributed flow of video traffic. So we have concentrated on this step to observe if there are any disturbances in the network.

If $T_{R(n)}$, $T_{R(n+1)}$ represents the receiver time stamps of n^{th} and $(n+1)^{\text{th}}$ frame, then IFT is given by:

$$IFT = T_{R(n+1)}T_{R(n)}$$

After calculating the inter frame time for the above videos at the following conditions, we plot and analyze using the Complementary Cumulative Distribution Function (CCDF) graph. Obtained CCDF graphs are matched with curves such as exponential, linear, logarithmic, etc. to check for best fit. The curves are best matched with exponential curves and some values of the obtained equation and Coefficient of Determination (R^2) value are tabulated in below Table 4.2.

S[dBm]	N[dBm]	Equation	R^2
45	70	$Y = 0.214778e^{-5.348563x}$	0.907795
55	70	$Y = 0.166252e^{-4.475908x}$	0.952970
65	70	$Y = 0.293201e^{-6.891760x}$	0.989185
45	80	$Y = 0.179534e^{-4.612561x}$	0.96791
55	80	$Y = 0.191188e^{-4.582816x}$	0.955602
65	80	$Y = 0.198541e^{-4.627708x}$	0.951919
75	80	$Y = 0.305495e^{-6.693311x}$	0.984357

Table 4.2: Exponential Curve & R^2 values

By observing, the equations fit into a functional form $Y = ae^{-bx}$, where, pre-factor value a is either greater than or less than one and b is the reciprocal time.

Below is the Table 4.3 where a, b, average $-1/b(s)$ and R^2 values for IFT 240 at constant noise and varied signal power are tabulated and the CCDF graphs follows.

S[dBm]	N[dBm]	a	b	$-1/b$(s)	R^2
-45	-70	0.214778	-5.348563	0.186966107	0.907795
-55	-70	0.166252	-4.475908	0.223418354	0.952970
-65	-70	0.293201	-6.891760	0.145100816	0.989185

Table 4.3: Approximation coefficients for video 240 in case of different signal and noise strengths

The below graph (Figure 4.1) is a representation of an estimated complementary cumulative distribution function of Inter-frame times at different signal powers (s) at a constant noise power of the video - 240. The blue, red, yellow line represents the signal powers such as -45 dBm, -55 dBm, -65 dBm. It can be inferred from the graph that at threshold x=100 ms, $\Pr\{IFT > x\}$ = 13%, 10%, 14% for signal power = -45 dBm, -55 dBm, -65 dBm. There is a significant change in curves as the time is increased by 150 ms, i.e. at around 250 ms, the curves meet and then we can observe a shift in curves (i.e. -45 dBm, -65 dBm) from there on and overlaps as they proceed. Furthermore, a curve that is higher than another means that values higher than x appear more frequently. Which implies that longer freezes appear more frequently.

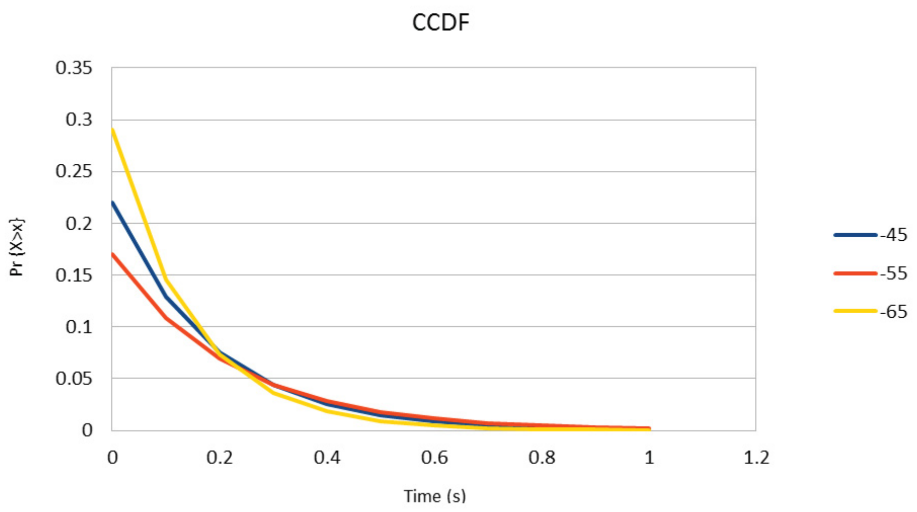

Figure 4.1: CCDF plot for the data in table 4.3 Inter frame time (in s) on X-axis and $\Pr\{X > x\}$ on Y-axis

The yellow line which is higher than the others indicates that it has more freezes and the higher risk (i.e. percentage) of inter frames times exceeding the normal value.

S[dBm]	N[dBm]	a	b	$-1/b$(s)	R^2
-45	-80	0.179534	-4.612561	0.216799301	0.96791
-55	-80	0.191188	-4.582816	0.218206448	0.955602
-65	-80	0.198541	-4.627708	0.216089693	0.951919
-75	-80	0.305495	-6.693311	0.149402889	0.984357

Table 4.4: Approximation coefficients for video 240 in case of different signal and noise strengths

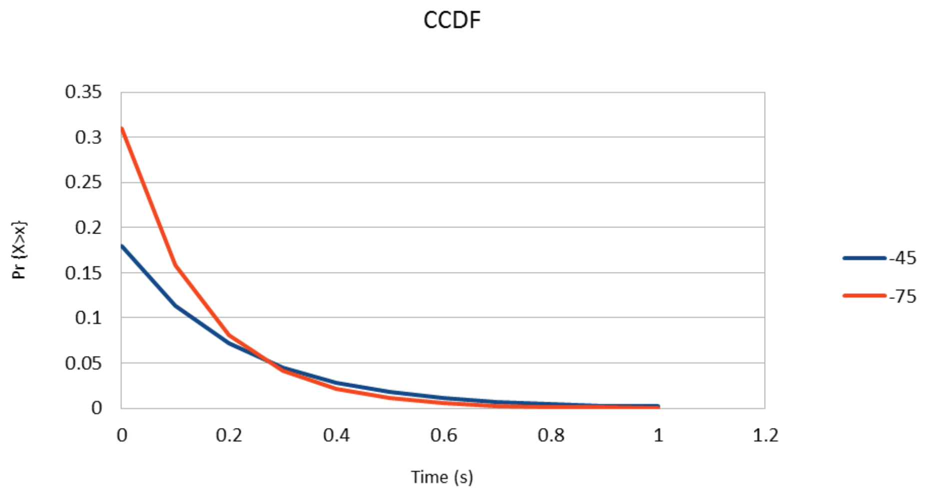

Figure 4.2: CCDF graph for the data in table 4.4 Inter frame time (in s) on X-axis and Pr$\{X > x\}$ on Y-axis

It can be inferred from the graph (Figure 4.2) that at threshold x=100 ms, Pr$\{IFT > x\}$ = 16%, 11.5% for signal power = -45 dBm, -75 dBm. These curves intersect at around 280 ms and then a shift in curves (i.e. -45 dBm, -75 dBm) from there on and overlap as they proceed.

The red curve which has a steep increase when compared to blue as the x values decreases from 100 ms to 0 ms. This results in a large gap between two curves, which indicates high inter frame times and freezes at S = -75 dBm and N= -80 dBm.

It can be inferred from the graph (Figure 4.3) that at threshold x=100 ms, Pr$\{IFT > x\}$= 12%, 14% for signal power = $\{$-55, -75$\}$dBm. These curves intersect at around 280 ms and then a shift in curves (i.e. -45 dBm, -75 dBm) from there on and overlap as they proceed.

S[dBm]	N[dBm]	a	b	$-1/b$(s)	R^2
-45	-90	N/A	N/A	N/A	N/A
-55	-90	0.194761	-4.86530	0.205537171	0.96854
-65	-90	0.200117	-5.02054	0.199181761	0.89887
-75	-90	0.243599	-5.45604	0.183283114	0.97362

Table 4.5: Approximation coefficients for video 240 in case of different signal and noise strengths

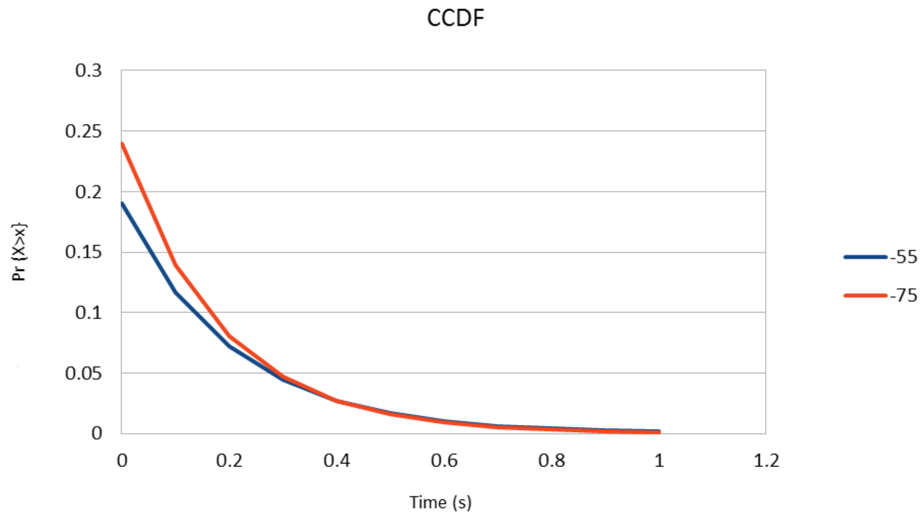

Figure 4.3: CCDF graph for the data in table 4.5 Inter frame time (in s) on X-axis and Pr$\{X > x\}$ on Y-axis

As shown in the graph (Figure 4.4) that blue, red, yellow curves represents the signal power {-45, -55, -65} dBm at a constant noise power of -70 dBm. It can be inferred from the graphs that, at threshold x=100 ms, Pr$\{IFT > x\}$ = {30%, 22%, 40%} for {-45, -55, -65} dBm. That means the percentage of inter frame times which exceed the 100 ms at the above conditions. The large gaps between yellow and blue curve indicate higher values of a for yellow curve which leads to higher risk of IFTs exceeding nominal value. These curves meet at 300 ms and then there is a change in trend i.e. vice-versa behaviour as yellow curve is below than others.

S[dBm]	N[dBm]	a	b	$-1/b$(s)	R^2
-45	-70	0.417727	-4.180672	0.239195995	0.963826
-55	-70	0.301062	-3.252656	0.307441057	0.965532
-65	-70	0.647066	-6.006813	0.166477631	0.932700

Table 4.6: Approximation coefficients for video 241 and Threshold 100 ms in case of different signal and noise strengths

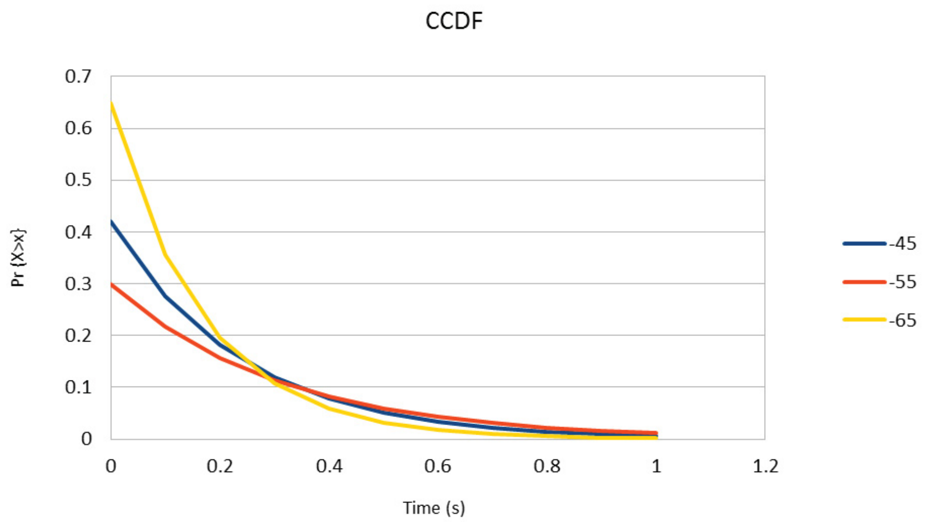

Figure 4.4: CCDF graph for the data in table 4.6 Inter frame time (in s) on X-axis and $\Pr\{X > x\}$ on Y-axis

As shown in the graph (Figure 4.5), a value is high for S = -75 dBm curve, there is a considerable gap between yellow and other two curves. We can see a linear increase in the a, b value as the SNR value is reduced. Hence, at this juncture the larger percentage of IFT exceeding the nominal value which leads to large freezes at thresholds 100 ms. These curves intersect at around 290 ms and then there is a shift in the behaviour as the yellow curve is below the other curves and other laps.

S[dBm]	N[dBm]	a	b	$-1/b$(s)	R^2
-45	-80	0.35947	-3.568527	0.280227668	0.957328
-55	-80	0.381187	-3.702118	0.270115647	0.885183
-65	-80	0.43904	-4.046108	0.247151089	0.897042
-75	-80	0.582661	-5.413287	0.184730645	0.981558

Table 4.7: Approximation coefficients for video 241, in case of different signal and noise strengths

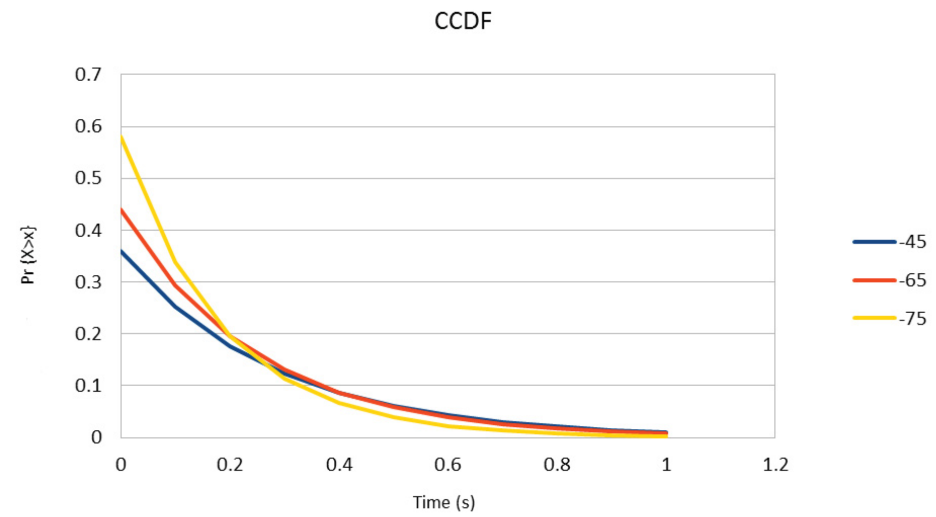

Figure 4.5: CCDF graph for the data in table 4.7 Inter frame time (in s) on X-axis and Pr$\{X > x\}$ on Y-axis

In the below graph (Figure 4.6), the blue, green curve represents curves of signal powers {-45, -55} dBm. As the a values are high for S = -45 dBm curve, there is a large gap between two curves.
Hence, the larger percentage of IFT exceeding the nominal value which leads to heavy freezes. These curves intersect at the threshold x=100 ms and then there is a complete shift in their behaviour i.e. the blue curve is below the red curve.
Therefore, a curve that is higher than another means that values higher than x appear more frequently.

S[dBm]	N[dB]	a	b	$-1/b$(s)	R^2
-45	-90	1.66414	-25.36437	0.039425383	0.99191
-55	-90	0.38680	-3.888042	0.257198868	0.91767
-65	-90	0.34284	-3.546511	0.281967263	0.89674
-75	-90	0.37985	-3.854742	0.259420734	0.90761

Table 4.8: Approximation coefficients for video 241, in case of different signal and noise strengths

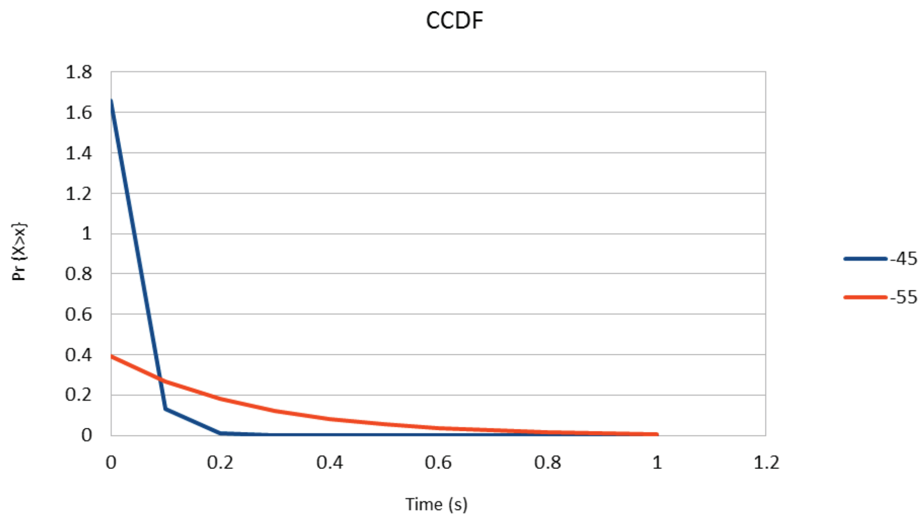

Figure 4.6: CCDF graph for the data in table 4.8 Inter frame time (in s) on X-axis and Pr$\{X > x\}$ on Y-axis

4.1.1 Analysis and Comparison

SNR: 15 dB:

For the analysis, lets consider SNR of 15 dB for video - 240 at different instances of signal and noise powers. They are,

S[dBm]	N[dBm]	a	$-1/b$(s)
-55	-70	0.166	-4.476
-65	-80	0.199	-5.62
-75	-90	0.243	-5.45

Table 4.9: Video 240, SNR = 15 dB

We can observe that a value has an increasing trend as the signal and noise power increases, whereas the b value doesn't follow the same trend. It increases and then decreases. If we compare this to the user experiment table below of the video 240 from the figure 4.4, the video perturbed with

SNR value of 15 dB (S= -59.3 and N = -74.3) has no freezes and has a smooth playout.

SNR: 5 dB:

Now, looking into the case where SNR is 5 dB.

S[dBm]	N[dBm]	a	$-1/b$(s)
-65	-70	0.293	-6.89
-75	-80	0.305	-6.69

Table 4.10: Video 240, SNR = 5 dB

Here, a values are high and are increasing. This follows the same trend as above as b value decreases. This high values of a will result in freezes which can be seen in the user experiment table below of video - 240 at S=-69.3 dBm and N= -74.3 dBm.

4.2 User Experience Results

The behaviour of the video application to the SNR disturbances and how it is behaved at the different signal powers at a constant noise power is presented below. The results are based on the experiments conducted by varying signal powers and observing the videos at two different buffer levels i.e. 32 KB and 320 KB (default value) using the procedure which is explained in chapter 3 at the network conditions of Downlink/Uplink: HSDPA/ 384 kbps. At this setting, the user experience of videos 240, 230, 140, 130 and 040 has been tabulated below

4.2.1 Video 240

By keeping the noise power constant at -74.3 dBm (default), the signal power is varied from -40.1 dBm in incremental steps. It is observed that there are small and frequent freezes when the cache is 32 kB and no freezes, smooth play in the same video when it is played with 10 times increase of cache i.e. at 320 kB. This behaviour is seen until the SNR value is 10 dB, a change in signal power by 2 dBm (i.e. from -64.3 dBm to -66.3 dBm) makes a notable change in the video behaviour. Freezes are introduced in video at cache level of 320 kB and increase in freezes and flat throughput is seen in video which is played at 32 kB cache that can be observed on the monitor of CMU200.

Signal (dBm)	Noise (dBm)	SNR (dBm)	Cache: 32kb	Cache: 320kb
-40.1	-74.3	34.2	Small, Frequent freezes	No freezes
-54.3	-74.3	20	Small, Frequent freezes	No freezes
-64.3	-74.3	10	Small, Frequent freezes	No freezes
-66.3	-74.3	8	Freezes, Flat throughput	Freezes, Flat throughput
-68.3	-74.3	6	Freezes, Flat throughput	Freezes, Flat throughput
-69.3	-74.3	5	Freezes, Flat throughput @ 1680 kbps	Freezes, Flat throughput
-70.3	-74.3	4	Freezes, Flat throughput @ 1680 kbps	Freezes, Flat throughput
-71.3	-74.3	3	Connection Lost	

Table 4.11: Video 240

It is to be noted that a change in SNR value by 2 dB i.e. from 10 dB to 8 dB, has a notable impact on the behaviour of the video at both the cache levels. This behaviour is observed until the SNR value is 4 dB and connection is lost between CMU 200 and Client when the SNR value is reduced to 3 dB.

4.2.2 Video 230

Video 230 has 640×368 resolutions same as 240 but has different layers and frame rate of 15 fps and bitrate of 1615.40 kbps. It is observed from the above table that there is almost same behaviour between these two videos at the same network conditions.

Signal (dBm)	Noise (dBm)	SNR (dBm)	Cache: 32kb	Cache: 320kb
-40.1	-74.3	34.2	Small freezes	No freezes
-54.3	-74.3	20	Small freezes	No freezes
-64.3	-74.3	10	Small freezes	No freezes
-66.3	-74.3	8	Freezes, Flat throughput	Freezes, Flat throughput
-68.3	-74.3	6	Freezes, Flat throughput	Freezes, Flat throughput
-69.3	-74.3	5	Freezes, Flat throughput @ 1680 kbps	Freezes, Flat throughput
-70.3	-74.3	4	Freezes, Flat throughput @ 1680 kbps	Freezes, Flat throughput
-71.3	-74.3	3	Freezes, Flat throughput @ 1680 kbps	Freezes, Flat throughput
-71.8	-74.3	2.5	Connection Lost	

Table 4.12: Video 230

Observation: From the tables, we can observe that the smooth play out of the both the videos is seen from S = -40.1 dBm to S= -64.3 dBm at N= -74.3 dBm (i.e. at SNR values of 34.2 dBm to 10 dBm) for cache levels of 320 kB.

4.2.3 Video 140

Video 140 has 352×288 resolutions with frame rate of 30 fps. It is observed that video 140 doesnt have any freezes and a smooth play out is seen in video when it is being played with default cache i.e. 320 KB but when this video is played with 1/10th of default cache size i.e. of 32 kB, we have freezes and jerky behavior in the video while being played. This can been seen in the initial conditions at S = -40.1 dB till -59.1 dBm. As the SNR value is reduced i.e. at 14.8 dBm, the increase in freezes is seen in video at 32 kB cache. This leads to bad experience for the user.

Signal (dBm)	Noise (dBm)	SNR (dBm)	Cache: 32kb	Cache: 320kb
-40.1	-74.3	34.2	Frequent Freezes + Jerkiness	No freezes
-45.1	-74.3	29.2	Frequent Freezes + Jerkiness	No freezes
-50.1	-74.3	24.2	Frequent Freezes + Jerkiness	No freezes
-55.1	-74.3	19.2	Frequent Freezes + Jerkiness	No freezes
-59.1	-74.3	15.2	Frequent Freezes + Jerkiness	No freezes
-59.5	-74.3	14.8	Freezes + Cache not filling	No freezes
-69.6	-74.3	4.7	Freezes + Cache not filling	No freezes
-69.7	-74.3	4.6	Freezes + Cache not filling (continuously)	No freezes
-70.8	-74.3	3.5	Freezes + Cache not filling (continuously)	No freezes
-71.8	-74.3	2.5	Connection Lost	

Table 4.13: Video 140

4.2.4 Video 130

The lower layered video 130, which has a temporal layer difference by one when compared to video 140. There are no freezes in video at initial conditions from $S=$ -40.1 dBm to -59.5 dBm at 32 kB cache, but we see them only from 59.6 dBm up till connection lost, we have initial freezes, initial delay but smooth play later. At cache 320 kB, the video has a smooth play with some initial delay.

Signal (dBm)	Noise (dBm)	SNR (dBm)	Cache: 32kb	Cache: 320kb
-40.1	-74.3	34.2	No freezes	No freezes
-45.1	-74.3	29.2	No freezes	No freezes
-50.1	-74.3	24.2	No freezes	No freezes
-55.1	-74.3	19.2	No freezes	No freezes
-59.1	-74.3	15.2	No freezes	No freezes
-59.5	-74.3	14.8	No freezes	No freezes
-59.6	-74.3	14.7	Initial Freezes + Initial Delay + Smooth play	No freezes
-69.7	-74.3	4.6	Initial Freezes + Initial Delay + Smooth play	No freezes
-71.8	-74.3	2.5	Connection Lost	

Table 4.14: Video 130

Observation: If the videos 140 & 130 are played at default cache levels, there is a smooth play of video without any freezes, which in turn gives good user experience.

4.2.5 Video 040

Signal(dBm)	Noise(dBm)	SNR(dBm)	Cache:32kb
-40.1	-74.3	34.2	No freezes
-55.1	-74.3	19.2	No freezes
-59.6	-74.3	14.7	No freezes
-69.8	-74.3	4.5	No freezes
-71.8		Connection Lost	

Table 4.15: Video 040

Video 040 is the base layer video which has no spatial, quality enhancement layers. In this case, we observe that until the connection is lost, the video is played without any disturbances.

From the above results, video 040 has a smooth play without any freezes in all the different network conditions, but it has less quality. For a better video quality, we can opt for video 140 or 130 but with cache (i.e. with 320 kB) as there is a smooth play without any freezes, which in turn affects the user experience.

4.3 Statistics

We have calculated the Average, Standard deviation, Minimum and Maximum values of inter-frame times of the videos 040.264, 140.264, 240.264 and 241.264 through the traces obtained. This is done through the analysis of logs which are collected at the user interface. In the 12 different scenarios, after calculating the above the values, we have tabulated three different scenarios namely better, medium and bad conditions.

4.3.1 Better condition at S = -45 dBm, N = -90 dBm.

	040	140	240	241
Average [s]	0.033	0.033	0.033	0.045
Standard Deviation [s]	0.000	0.005	0.011	0.044
Minimum [s]	0.016	0.018	0.005	0.007
Maximum [s]	0.038	0.048	0.177	0.300

Table 4.16: Inter-frame time statistics of Better Condition of video at S = -45 dBm, N = -90 dBm

In this network condition, the videos 040, 140, 240 has a smooth playback with constant frame rate matching the nominal value of 3 frames per 100 ms interval. As it can be observed from the above table that the averages are 33 ms (i.e. 0.033 s) for the three videos and video 241 has slightly more average among them, which will result in small freezes.

4.3.2 Medium condition at S = -45 dBm, N = -70 dBm

	040	140	240	241
Average [s]	0.033	0.033	0.061	0.113
Standard Deviation [s]	0.001	0.005	0.106	0.185
Minimum [s]	0.012	0.024	0.003	0.007
Maximum [s]	0.037	0.049	0.779	1.113

Table 4.17: Inter-frame time statistics of Medium Condition of video at S = -45 dBm, N = -70 dBm

In this condition, 040 and 240 video has a nominal average of 33 ms, but in case of video 240 and 241, average is doubled. There is a steep increase in standard deviation by 10 times and difference between maximum values and minimum values is also increased. This indicated the gap which leads to delay in frames i.e. frequent freezes can be observed at the user interface.

4.3.3 Medium condition at S = -65 dBm, N = -90 dBm

	040	140	240	241
Average [s]	0.033	0.033	0.060	0.112
Standard Deviation [s]	0.001	0.005	0.011	0.198
Minimum [s]	0.012	0.024	0.003	0.007
Maximum [s]	0.035	0.049	0.932	1.117

Table 4.18: Inter-frame time statistics of Medium Condition of video at S = -65 dBm, N = -90 dBm

This condition also depicts the same behaviour as of above. The slight increase of standard deviation and maximum inter frame time value when compared to above condition. The maximum value in video 240 is more than 0.9 s which is slightly tending towards 1 second.

4.3.4 Bad condition at S = -55 dBm, N = -70 dBm

	040	140	240	241
Average [s]	0.033	0.033	0.061	0.113
Standard Deviation [s]	0.000	0.005	0.119	0.206
Minimum [s]	0.016	0.024	0.003	0.007
Maximum [s])	0.038	0.048	1.228	1.216

Table 4.19: Inter-frame time statistics of Bad Condition of video at S = -55 dBm, N = -70 dBm

In this condition, the videos 040, 140 are not affected. They have a smooth playback as it has average of 33 ms and standard deviation is also approximately zero, whereas videos 240 and 241 maintain the same double average than nominal value. The standard deviations are high and maximum inter frame time approximately 1.2 sec and the minimum value is lower which shows gaps and leads to higher amount of frequent freezes in video. This will lead to bad experience to the user and ultimately affecting the QoE.

From all the values at different cases, we find better condition for the video is at S = -45 dBm, N = -90 dBm, medium conditions at S = -45 dBm, N = -70 dBm, S = -65 dBm, N = -90 dBm and bad condition at S = -55 dBm, N = -70 dBm.

Chapter 5

Conclusion and Future Work

This thesis work presents the analysis and demonstrates the effects of SNR on frame rates of a streaming video with the help of some demonstrations.

5.1 Answers to Research Questions

RQ 1: How does the radio-level disturbances affects the instantaneous frame rates of an SVC video stream?

By comparing the statistical results, we conclude that when signal and noise powers are high i.e. SNR is low, there is more inter-frame duration between frames and this leads to more freezes in a video. When SNR is high, there was less percentage of freezes and video had a smooth play out.

RQ 2: How do irregularities in inter-frame times affect the QoE?

The irregularities in inter-frame times will affect in freezes, delay, etc which will impact the user experience. User gets disturbed when the freezes are longer. This effect is explained in result section with graphs and tables. As the SNR value is reduced from high to minimum (i.e. SNR=34 to 3 [dB]), there is increase in freezes which are due to large inter-frame times. This will have an impact on the QoE.

Research gives the picture of user experience, when there are disturbances and delays in a video. In literature and past research works, when MOS is taken, the users gave least score even if the delay is 150 ms [26]. This in case of real-time video streams that has strigent delay requirements suffer with more adverse end user experience.

RQ 3: Which are the telling use cases for demonstrating the effect of link-level disturbances on QoE of SVC-coded video?

Telling cases: The process of demonstrating the interesting cases or scenarios which are useful for research, education, etc. Through the demonstrator setup, we analyzed the disturbances imposed by fundamental system settings and parameters on the videos, and developed instructive cases for demonstration (these are explained in results section of previous chapter), which serve the set of stakeholders defined in subsection 5.2.

5.2 Stakeholders

Students: The thesis results and work can be included in academic coursework to explain how the disturbances in lower layer such as physical layer, etc. can impact the applications (in particular video). Through these live demonstrations, students can gain insight in a unique way how the layers in the OSI reference model are interconnected to and affect each other.

Researchers: The researches can use the setup as test bed, and the results as a reference. They also can extend the work to find more relations between QoE and underlying system parameters, as well as derive ways to overcome the corresponding challenges in providing high-quality video streaming.

Visitors: By demonstrating the results, a non-technical person can also understand the technical concepts and can relate how the disturbances impact the applications (i.e. video) in mobile devices. They can see how a small change in signal-to-noise ratios can make a significant impact on the video quality.

Companies: The telecom companies (service providers) / Web based service providers/ online music / video providers etc., can use the study to come up with optimal solution to plan to provide quality video services and the this study can be also applicable to audio streaming where, quality of audio can be enhanced. The thesis can be used to enhance the business around internet / mobile multimedia services.

5.3 Future Work

Our future work can be extended to testing the video with both audio and video encoders along work with different transmission protocols like UDP, RTMP, etc. The work can be done by adding a shaper in between the server and base station. This can introduce network level artifacts. Thus we can quantify QoE by combining physical layer, network layer disturbances. The work can be extended by exploring traffic models which relate application layer with physical layer.

Bibliography

[1] MPlayer. www.cisco.com. [Online]. Available: http://www.cisco.com/en/US/prod/collateral/wireless/ps6302/ps8322/ps10315/ps10325/white_paper_c11-577721.pdf

[2] J. Klaue, B. Rathke, and A. Wolisz, "EvalVid A Framework for Video Transmission and Quality Evaluation," in *Computer Performance Evaluation. Modelling Techniques and Tools*, ser. Lecture Notes in Computer Science, P. Kemper and W. Sanders, Eds. Springer Berlin Heidelberg, vol. 2794, pp. 255–272. [Online]. Available: http://dx.doi.org/10.1007/978-3-540-45232-4_16

[3] L. Lu, X. Lu, and J. Li, "Quality monitoring of streaming video based on packet-loss artifacts," in *Wireless and Optical Communications Conference (WOCC), 2010 19th Annual*, May 2010, pp. 1–5.

[4] Home Page. www.etsi.org. [Online]. Available: http://www.etsi.org/deliver/etsi_ts/126200_126299/126234/04.01.00_60/ts_126234v040100p.pdf

[5] T. Minhas, M. Fiedler, J. Shaikh, and P. Arlos, "Evaluation of throughput performance of traffic shapers," in *Wireless Communications and Mobile Computing Conference (IWCMC), 2011 7th International*, July 2011, pp. 1596–1600.

[6] J. Shaikh, T. Minhas, P. Arlos, and M. Fiedler, "Evaluation of delay performance of traffic shapers," in *Security and Communication Networks (IWSCN), 2010 2nd International Workshop on*, May 2010, pp. 1–8.

[7] W. Cherif, A. Ksentini, D. Negru, and M. Sidibe, "A_PSQA: Efficient real-time video streaming QoE tool in a future media internet context," in *Multimedia and Expo (ICME), 2011 IEEE International Conference on*, July 2011, pp. 1–6.

[8] T. Minhas and M. Fiedler, "Impact of disturbance locations on video quality of experience," Blekinge Institute of Technology, Research Report, 2011.

[9] K. Piamrat, C. Viho, J. Bonnin, and A. Ksentini, "Quality of Experience Measurements for Video Streaming over Wireless Networks," in *Information Technology: New Generations, 2009. ITNG '09. Sixth International Conference on*, April 2009, pp. 1184–1189.

[10] D. Wu, S. Member, Y. T. Hou, W. Zhu, Y. qin Zhang, J. M. Peha, and S. Member, "Streaming video over the Internet: approaches and directions," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 11, pp. 282–300, 2001.

[11] T. Wiegand, G. Sullivan, G. Bjontegaard, and A. Luthra, "Overview of the H.264/AVC video coding standard," *Circuits and Systems for Video Technology, IEEE Transactions on*, vol. 13, no. 7, pp. 560–576, July 2003.

[12] L. Angrisani, K. Kyriakopoulos, A. Napolitano, D. Parish, M. Vadursi, and W. Whittow, "An experimental analysis of the effects of noise on Wi-Fi video streaming," in *Instrumentation and Measurement Technology Conference (I2MTC), 2010 IEEE*, May 2010, pp. 1551–1555.

[13] T. Varisetty and P. Edara, "Systematic Overview of Savings versus Quality for H.264/SVC ," Master's thesis, Blekinge Institute of Technology, Sweden, 2012.

[14] D. Marpe, T. Wiegand, and G. Sullivan, "The H.264/MPEG4 advanced video coding standard and its applications," *Communications Magazine, IEEE*, vol. 44, no. 8, pp. 134–143, Aug 2006.

[15] J. Ohm, "Advances in Scalable Video Coding, year=2005," in *Proceedings of the IEEE*, no. 1, January, pp. 42–56.

[16] "MPEG Video Subgroup Chair, Registered Responses to the Call for Proposals on Scalable Video Coding," ISO/IEC JTC 1/SC29/WG11, Research Report, 2004.

[17] H. Schwarz, D. Marpe, and T. Wiegand, "Overview of the Scalable Video Coding Extension of the H.264/AVC Standard," *Circuits and Systems for Video Technology, IEEE Transactions on*, vol. 17, no. 9, pp. 1103–1120, Sept 2007.

[18] T. A. Le, H. Nguyen, and H. Zhang, "Scalable Video Transmission on Overlay Networks," in *Advances in Multimedia (MMEDIA), 2010 Second International Conferences on*, June 2010, pp. 180–184.

[19] Home Page. www.eetimes.com. [Online]. Available: http://www.eetimes.com/design/signal-processing-dsp/4017613/Tutorial-The-H-264-Scalable-Video-Codec-SVC

[20] T. P. Nguyen and A. Zakhor, "Partial Video Replication for Peer-to-Peer Streaming," ser. In: SPIE/ACM MMCN.

[21] T. Kim and M. H. Ammar, "Receiver buffer requirement for video streaming over TCP," vol. 6077, 2006, pp. 607 718–607 718–10. [Online]. Available: http://dx.doi.org/10.1117/12.643583

[22] N. Ahmed, S. S. Kanhere, and S. Jha, "The Holes Problem in Wireless Sensor Networks: A Survey," *SIGMOBILE Mob. Comput. Commun. Rev.*, vol. 9, no. 2, pp. 4–18, Apr 2005.

[23] I. Kofler, R. Kuschnig, and H. Hellwagner, "In-Network Real-Time Adaptation of Scalable Video Content on a WiFi Router," in *Consumer Communications and Networking Conference, 2009. CCNC 2009. 6th IEEE*, Jan 2009, pp. 1–2.

[24] H. Hisamatsu, G. Hasegawa, and M. Murata, "Non Bandwidth-intrusive Video Streaming over TCP," in *Information Technology: New Generations (ITNG), 2011 Eighth International Conference on*, April 2011, pp. 78–83.

[25] N. Cranley and M. Davis, "Study of the Behaviour of Video Streaming over IEEE 802.11b WLAN Networks," *2013 IEEE 9th International Conference on Wireless and Mobile Computing, Networking and Communications (WiMob)*, vol. 0, pp. 349–355, 2006.

[26] Y. Alahari and B. Prashant, "Analysis of packet loss and delay variation on QoE for H.264 and WebM/VP8 Codecs," Master's thesis, Blekinge Institute of Technology, Sweden, 2011.

[27] T. N. Minhas and M. Fiedler, "Quality of experience hourglass model," in *Computing, Management and Telecommunications (ComManTel), 2013 International Conference on*, Jan 2013, pp. 87–92.

[28] Flumotion streaming server. www.flumotion.net. [Online]. Available: http://www.flumotion.net/doc/flumotion/manual/en/0.6.1/html/index.html

[29] MPlayer. www.mplayerhq.hu. [Online]. Available: http://www.mplayerhq.hu/design7/news.html

[30] C. Chen, "MPlayer Based Home Media Center Implementation on Intel Olo River," Master's thesis, Blekinge Institute of Technology, Sweden, 2007.

[31] Home Page. www.rohde-schwarz.com. [Online]. Available: http://www.rohde-schwarz.com/file_1256/n186_cmu200.pdf

[32] Home Page. www.rohde-schwarz.com. [Online]. Available: http://www.rohde-schwarz.com/file_11584/CMU200_bro_en.PDF