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Video Quality in IP Based Mobile Systems

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MASTER'S THESIS EX038/2008

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Master's Thesis in the Digital Communication Systems and technology

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Abstract

Digital video transmission over packet switched wireless networks is increasingly utilized by customers. It is of high importance to have reliable test solutions to measure the perceived end user video quality.
Study of video characteristics, IP networks and video quality metrics, are the bases of this thesis work. The compressed video transmitted over IP, suffers from network impairments such as packet loss and jitter. However this network impact can not be easily measured in the form of perceived video quality degradation. Analysis a way of video quality measurement for mobile video streaming is one of the objectives of this thesis. We also aim to investigate and evaluate the existing video quality test tools.

Acknowledgment

In the course of preparing this master’s thesis, I have received invaluable help from numerous people. I would therefore like to express my deepest gratitude to everyone who has helped me. My supervisor at Ericsson Mats Åsberg who supported me and guided throughout the whole thesis work and in every single step. Magnus Thorstenson for believing in me and allowing me to do the thesis work at Ericsson.
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Further more I am grateful to all those at Ericsson who have been so nice and helpful to me, during my stay at Ericsson. And finally thanks to my loving husband and my parents for their love and support.

Göteborg May 2008
Maliheh Yousefi

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Abbreviations and Acronyms

3GPP	3^{rd} Generation Partnership Project
3GP	3GPP File extension
ACK	Acknowledgement
AU	Access Unit
AV	Audio and Visual
AVC	Advanced Video Coding
BSC	Base Station Controller
BSS	Base Station System
Cb	Blue Chroma
Cr	Red Chroma
DCT	Discrete Cosine Transform
DF	Delay Factor
DMOS	Differential Mean Opinion Score
DVB-H	Digital Video Broadcasting-Handheld
DVB-T	Digital Video Broadcasting-Terrestrial
EDGE	Enhanced Data Rates for Global Evolution
ES	Elementary Stream
FR	Full Reference
FTP	File Transfer Protocol
GGSN	Gateway GPRS Support Node
GSM	Global System for Mobile communication
GPRS	General Packet Radio Service
GTP	GPRS Tunneling Protocol
GUI	Graphical User Interface
HTML	HyperText Markup Language
HTTP	HyperText Transfer Protocol
ID	Identity
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IS	International Standard
ISO	International Organization for Standardization
ITU	International Telecommunication Union
JPEG	Joint Photographic Experts Group
KPI	Key Performance Indicator
LTE	Long Term Evolution
MAC	Multiple Access Control
MBMS	Multimedia Broadcast Multicast Service
MDI	Media Delivery Index
MLR	Media Loss Rate

MMS	Multimedia Messaging Service
MOS	Mean Opinion Score
MPEG	Moving Picture Experts Group
MPEG-4	Digital video compression standard and file format,
MTU	Maximum Transmission Unit
NAL	Network Abstraction Layer
NETEM	NETwork EMulator
NIC	Network Interface Card
NR	No Reference
ISO	International Organization for Standardization
PDP	Packet Data Protocol
PES	Packetized Elementary Stream
PEVQ	Perceptual Evaluation of Video Quality
PID	Packet Identifier
PS	Packet-Switched
PSS	Packet Switched Streaming Service, 3GPP
QCIF	Quarter Intermediate Format
QoE	Quality of Experience
QoS	Quality of Service
RFC	Request For Comments (formal document from IEEE)
RLC	Radio Link Control
RR	Reduced Reference
RTCP	RTP Control Protocol
RTP	Real-Time Transport Protocol
RTSP	Real Time Streaming Protocol
SGSN	Serving GPRS Support Node
TCP	Transmission Control Protocol
TPA	Triple Play analyzer
TS	Transport Stream
TVQM	Telchemy Video Quality Metrics
UDP	User Datagram Protocol
UE	User Equipment
UMTS	Universal Mobile Telecommunication System
VCEG	Video Coding Expert Group
VCL	Video Coding Layer
VoD	Video on Demand
VQA	Video Quality Analyzer
VQM	Video Quality Measurement
VSQI	Video Streaming Quality Index
WAP	Wireless Application Protocol
WCDMA	Wideband Code Division Multiple Access
WWW	World Wide Web
XHTML	eXtended HyperText Markup Language
Y	Luma

Chapter 1

1. Introduction

This chapter starts with a brief introduction of the thesis, followed by problem statement, goal and scope of the thesis. The chapter ends with the thesis structure.

1.1. Overview

Transmitted video over IP experiences a variety of different network conditions. The codec schemes and IP packetizing have also a wide range of effects on video quality. Also the various clients in the receivers have much impact on the video. Voice services have rigidly standardized encoders and decoders and no client can introduce a wide range of different effects on the signal. In contrast, video streaming has not such unanimous standards and is always application specific. Thus there is not such a best test tool for all kinds of video streaming; where the conditions differ from phone to phone and between different applications. However it is always better to measure something rather than to have no quantified data.

1.2. Problem Statement

In this thesis work we aim to evaluate end user video quality, and pinpoint the IP network impacts on the final perceived video. Analysis of a way of testing video quality for MobileTV/ Streaming (way of working) and investigating and evaluating possible test tools which match best with our requirements, are the body of the Master's thesis. Running preferably some automated tests with the selected test tool, is the final part of the work.

1.3. Goals and Scope

We intend to study quality of transmitted video over IP networks, in a wireless system.
The application is Mobile TV in GSM network. Mobile TV service may include unicast , multicast or broadcast services. However we study Video on Demand (VoD) which is unicast and point to point link. Though any transmission of video is typically accompanied with the audio signals, the focus is on the video quality. Audio quality is not included in this Master's thesis.
The EDGE GSM network is the environment for all the tests we run, with the assumption of pure and perfect radio channel. The video codecs and protocols are according to standards for streaming over IP. Though the study is on GSM network, the principles of video quality remain the same for 3G or Long Term Evolution (LTE). Some of the tools that are introduced in chapter 4 are 3G tools.
In this thesis work, we try to get a deeper view to video characteristics over IP and pinpoint the shortcoming of assessing video quality over IP.

1.4. Structure of the Thesis

We start in chapter 2 with a brief study of different quality metrics, video measurement methods , related protocols and video codecs. This chapter does not consider the mobile TV application, but only focuses on video over IP and video quality measurement basics.

Then in the next chapter, mobile TV is introduced. Media retrieval ways, streaming, mobile phone client and buffering, are the next steps of the work. Finally the chapter ends with the mobile TV signaling diagram.

Chapter 4 introduces video quality test tools provided by a number of vendors. An understanding of compressed video over IP and mobile TV application from previous chapters, helps us to find more relevant tools to our objectives.

Some tests are done with a selected tool, in the Ericsson test lab. The description of test scenario, network emulation and video quality score results are brought in chapter 5.

Finally in chapter 6, conclusions of the work and suggestions for further work are described.

Chapter 2

2. Background

In this chapter, we study briefly the common keywords in video quality over IP, standards and protocols. Firstly the emerging need for video quality testing is discussed, followed by definition of quality, QoS and QoE. Secondly the video quality measurement methods are brought, from subjective to objective solutions. In order to have a good idea of video transportation a fairly good background in related IP protocols is essential, which is also brought in this chapter. Finally video codecs, the methods and base algorithms in video compression are shortly discussed.

2.1. The Need for Video Quality Testing

The need for perceived video quality testing is crucial as a growing number of customers are utilizing streaming media than ever before; Spirent Communications.(2005,p1) states "In a 2004 report released by Aberdeen Group and Streamingmedia.com, nearly 100% of the 3,000+ respondents were using streaming media technology for a business application within their organization." Web conferencing and Webcasting lead in business applications, were used 65% and 72%, respectively, in the end of 2004 [1].

Good and reliable testing solutions are key elements to guarantee the customers' satisfaction. Some times the vital information of a video stream might be lost due to lossy coding techniques or noisy transmission channels, that may cause a totally different understanding of a scene. A Rally race car that leaves the track and rushes into the green, is a good example. A high video quality is essential to correctly observe whether the driver is hurt or not.

2.2. Quality- What it is?

ISO 8402 suggests a definition of quality as: "The totality of characteristics of an entity that bear on its ability to satisfy stated and implied needs"(cited by [2]).

2.2.1. QoS versus QoE

Quality of Service and Quality of Experience are of our interest. They have different definitions in the standards world and we try to bring the well accepted ones here.

2.2.1.1. Quality of Service

ITU-T E.800 (1994) defines Quality of Service (QoS) as "the collective effect of service performance which determine the degree of satisfaction of a user of the service."(cited by [3]).

The quality of service is characterized by the combined aspects of service support performance, service operability performance, service ability performance, service security performance and other factors specific to each service [3]. QoS does not always guarantee the end user satisfaction.

DSL Forum WT-126 explains:
“QoS is a measure of performance at the packet level from the network perspective. QoS also refers to a set of technologies (QoS mechanisms) that enable the network administrator to manage the effects of congestion on application performance as well as providing differentiated service to selected network traffic flows or to selected users. QoS metrics may include network layer measurements such as packet loss, delay or jitter.”(cited by [3]).

2.2.1.2. Quality of Experience

A Quality of Experience (QoE) definition proposed by ITU-T SG 12 (TD 44rev1 GEN,2004) states: “The overall acceptability of an application or service, as perceived subjectively by the end-user”(cited by [3]).
Quality of Experience includes the complete end-to-end system effects (client, terminal, network, services infrastructure, etc) e.g. IPTV channel change time, video quality, VoD download times and voice quality, so it is quite clear that the overall acceptability may be influenced by user expectations and context [3]. Video QoE has two parts: “Zapping” and “Video Quality ”. Zapping time can be easily characterized and measured, but video quality or Video Mean Opinion Score has no unanimous definition [4].

2.2.2. Classifying Quality Layers

The above descriptions and definitions on QoE and QoS and their relation to our application, which is streaming video quality, may seem quite confusing.
Figure 2-1 shows a scheme of Quality of Service metrics which merge to Quality of Experience Indicators. The layered structure defines different areas of Network and Application layers. Let us review the Master’s thesis subject; measuring MobileTV quality over IP based networks implies that we are not interested in “Content Quality” layer which is irreversibly impaired mostly by the compression algorithms, not by IP network, this layer leads to a set of QoE Indicators such as blurring and jerkiness.
On the other hand, studying “Transmission Quality” layer or “Streaming Quality” comes to the results that are fairly relevant to IP network impacts on the video quality.
This classification is also very much helpful for investigating the Media Test Tools; provided by various vendors. By simply comparing the QoE Indicators in this figure with the tools functionalities and features, we could easily get a clue to find out whether each tool is fulfilling our requirements or not.

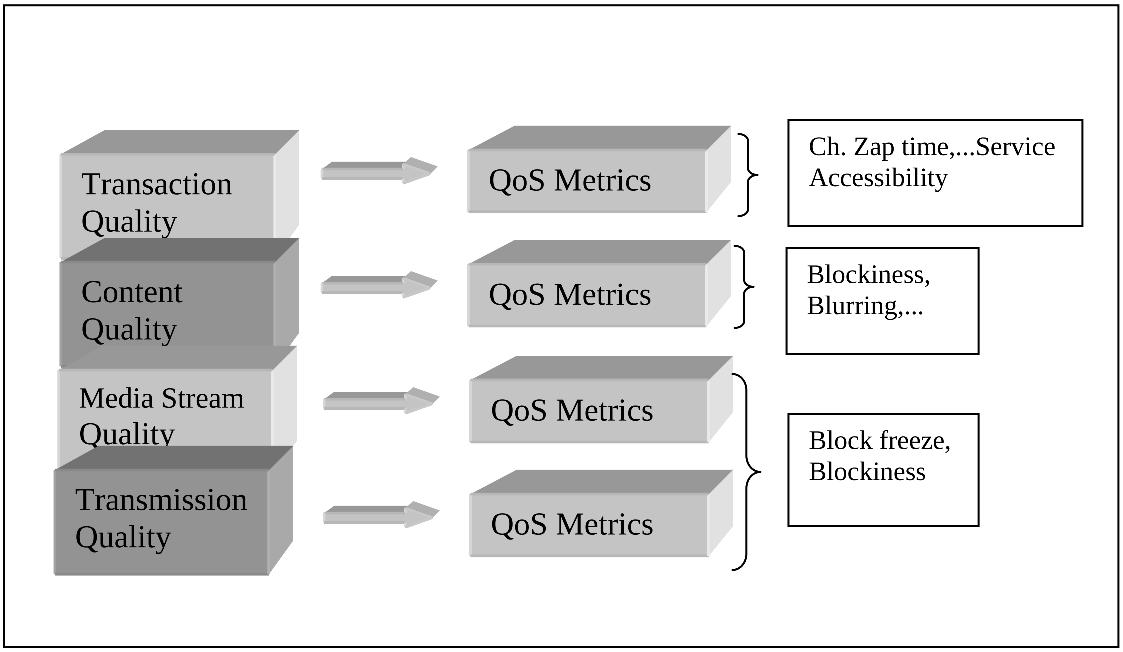

Figure 2-1: Quality Classification for Video streaming

2.2.3. Mean Opinion Score

Mean Opinion Score (MOS) is one of the QoE metrics which are typically used in subjective and some of objective measurements to quantify the perceptual impacts of various forms of service degradation [3].

A typical MOS for describing the video quality ranges from 1 for bad quality to 5 for excellent quality [8] as shown in Table 2-1.

MOS	Quality
5	Excellent
4	Good
3	Fair
2	Bad
1	Poor

Table 2-1: MOS score

2.3. Video Quality measurement Methods

Here, we divide the video quality tests into two categories; human viewer tests and scoring, and tests based on objective models that try to accurately predict the human quality scoring.

2.3.4. Subjective Testing

A formal subjective test is an experiment carried out according to rigorous protocols based on psychophysical methodologies [5].

In Subjective method a team of test persons should watch the video sequence, evaluate and score it, obviously it is time consuming, expensive and in some cases impractical. However these subjective tests are essential before developing any objective measurement algorithms, as they model subjective tests and human vision with the help of a big fund of subjective measurement data [6].

2.3.5. Objective Testing

The main goal of objective testing is to find automatic metrics which provide computed quality scores, which are well correlated to the subjective test scores. Thus video transmission quality control is provided in broadcasting or unicasting like Mobile TV [7].
According to [7] objective image quality measurement methods can be divided in three categories :
- full reference method (FR) for which the original image and the distorted image are required,
- reduced reference method (RR) for which a description of the original image into some parameters and the distorted image are both required,
- and no reference (NR) method which only requires the distorted image.

2.3.5.1. Full Reference Method

FR methods are normally more precise and more correlated with the subjective perceived quality scores than NR. Synchronization is needed to correctly align the impaired frame with the undistorted frame. FR is a computationally intensive process as it not only involves per-pixel processing but also time and spatial alignment of the input and output streams. FR methods are ideal for evaluating the compression algorithm performance and comparing different coding schemes, which implies that they can only be used in certain applications - for example in lab testing or pre-deployment tests.

2.3.5.1.1. Typical Metrics in Full Reference Method

Blockiness, Jerkiness and Blurring are some of the known metrics in FR method, mostly caused by coding impairments. Here, we bring a brief definition of most well known video degradation metrics by ITU-T recommendation P.910:

-**Jerkiness**
describes the smoothness of a video playback which is often impaired by down-sampling, coding processes and perturbed transmissions.

-**Blockiness**
is often the result of a low bit rate coding that uses a block matching algorithm for the motion estimation and a coarse quantization for the image blocks.

-Blur
is a distortion characterized by reduced sharpness of contour edges and spatial detail. Example below compares an original image with its blurred copy.

-Brightness
The brightness of the reference and degraded signal.

-Contrast
The contrast of the distorted and the reference sequence.

-PSNR
To allow for a coarse analysis of the distortions in different domains the PSNR is provided for the Y(luma), Cb (blue chroma) and Cr(red chroma) components of digital image separately.

-Frame Skips and Freezes
are temporal artifacts occurring in video transmissions caused by e.g. overloaded networks.

-Effective Frame Rate
Down-sampling of a video signal on a frame by frame basis often results in loss of information which often leads to the degradation of the video signal. The effective frame rate is an indicator quantifying the severeness of such a process.

-Temporal and Spatial Activity
Temporal and spatial activity indicators quantify the amount of activity /movement in the video content. The former is about the motion in a video scene and the latter is each frame complexity in colors, edges, shape and regions.
These indicators are cited by Opticom GmbH (2006,p.2) [8]. As stated before, the FR metrics indicate the codec impairments rather than network and channel impacts, where NR emerges more relevant to our objectives.

2.3.5.2. Reduced Reference Method

In this video quality method, only reduced bandwidth features are extracted from the original signal and compared with the corresponding features of the impaired signal. The features are extracted from spatial-temporal regions of video scenes which characterize image edge and motion respectively . Since this amount of information can be transmitted over communications networks, the end-to-end in service quality monitoring is possible [9].

2.3.5.3. No Reference Method

NR methods do not need the original signal, so they are more applicable; as access to the undistorted signal may be difficult or impractical. Thus no synchronization is required. In fact NR methods do not measure degradation but assess the perceived video quality based on their own metrics and extracting some predefined properties of the video signal [10].
NR algorithms are generally more suitable for in-service monitoring of video services as they are less computationally heavy and can analyze live streams.

2.3.6. Perceptual versus Non-Perceptual Input

Perceptual evaluation is based on modeling the behavior of the human visual system [8]. This means only those parameters are taken into account which are actually perceived by a the human viewer like Y, Cr and Cb components of digital image. Perceptual artifacts are like blurriness, jerkiness; according to these metrics the perceived quality is stated in terms of MOS. The algorithm usually uses a database of subjective mean opinion scores and a model of human visual system, which make it computationally complex.

On the other hand, input to Non-perceptual algorithms are parameters that can not be perceived by a human, such as throughput or block error over a communication link, yet trying to estimate the perceived quality. So it is predictable that in general; non-perceptual methods are less robust, unless they are trained for a specific set up. Hence for an optimized quality estimation, the video codec and a limited set of bit rates should be specified. Methods which use non-perceptual parameters, will not give information on each single impaired frame, instead an average performance is assessed. The main non-perceptual parameters may be listed as choice of codecs, packet loss level, frame rate, throughput and rebuffering [10].

Now, one can see that FR methods by nature have access to perceptual parameters, while NR methods have only non-perceptual parameters in hand, on which the algorithm tries to estimate the perceived quality. In other words, the non- perceptual parameters are mostly those of satisfying QoS, but yet there are some methods that estimate QoE , out of these network events.

2.3.7. Media Delivery Index

Media Delivery Index (MDI), defined in RFC 4445 [11], is a figure calculated by non-perceptual inputs. The inputs are two network parameters; Delay Factor (DF) and Media Loss Rate (MLR) . MDI is presented as DF :MLR that gives an estimate of network conditions. There are a lot of discussions on the reliability of MDI, for assessing the perceived video quality. Here DF and MLR are introduced shortly try to find out how reliably one can estimate QoE or QoS by use of MDI, but before that jitter and its relation to the buffer size is introduced.

2.3.7.1. Jitter

Jitter is the variation in the end-to-end latency in time domain, caused by network congestion, route changes or other reasons. If packets arrive at a constant rate there is no jitter, while variable arrival rate of packets exhibits non-zero jitter. See Figure 2-2.

To overcome the jitter a buffer in the receiver is needed to collect a sufficient number of packets before feeding them to the decoder. The buffer may experience overflow or underflow, depending on the rate of arriving packets. Packets arriving at such a high rate that fill the buffer, cause packet drop at the receiver, known as buffer overflow. Underflow happens when packets arrive so slowly that the buffer has not enough data to feed the decoder.

Apparently both of these two cases are undesirable, for they degrade the QoE.

The more severe jitter, the larger buffer is needed to be able to eliminate the jitter effect. On the other hand large buffer sizes introduce larger delays before media playback to the user.

2.3.7.2. Delay Factor

Drain rate refers to the payload media rate, e.g. for a 3.75 Mb/s MPEG TS, the drain rate is 3.75Mb/s , which is the displayed rate at the receiver decoder.
Delay Factor (DF) is the maximum difference, between the arrival of media data and the drain of media data, that is observed at the end of each media stream packet [11]. In other words, DF is a time value, indicating the minimum buffer size, in order to eliminate jitter [12].
[12] states DF can be employed at the video receiver side to assess the video quality from the user's perspective, for example the maximum acceptable DF for IPTV may vary between 9-50 msec. This range is due to the wide variation in the buffer sizes of available user set top boxes.

2.3.7.3. Media Loss Rate

The Media Loss Rate (MLR) is the number of out of order or lost flow packets over a selected time interval. Flow packets carry media data [11].Thus in a typical IP packet , 7 MPEG2-TS are carried and one IP packet loss, results in 7 media packet loss. Out of order packets are also counted , as many devices do not attempt to reorder packets, before presenting them to the decoder, so out of order packets are simply dropped. This shows how quality of video is dependent on the mobile phone client characteristics.
Any non-zero MLR adversely affects the video quality, and can make visual distortions.
Upon the above descriptions of DF and MLR, an MDI of 3:0.001 means a delay factor of 3 milliseconds and loss rate of 0.001 media packet per second.

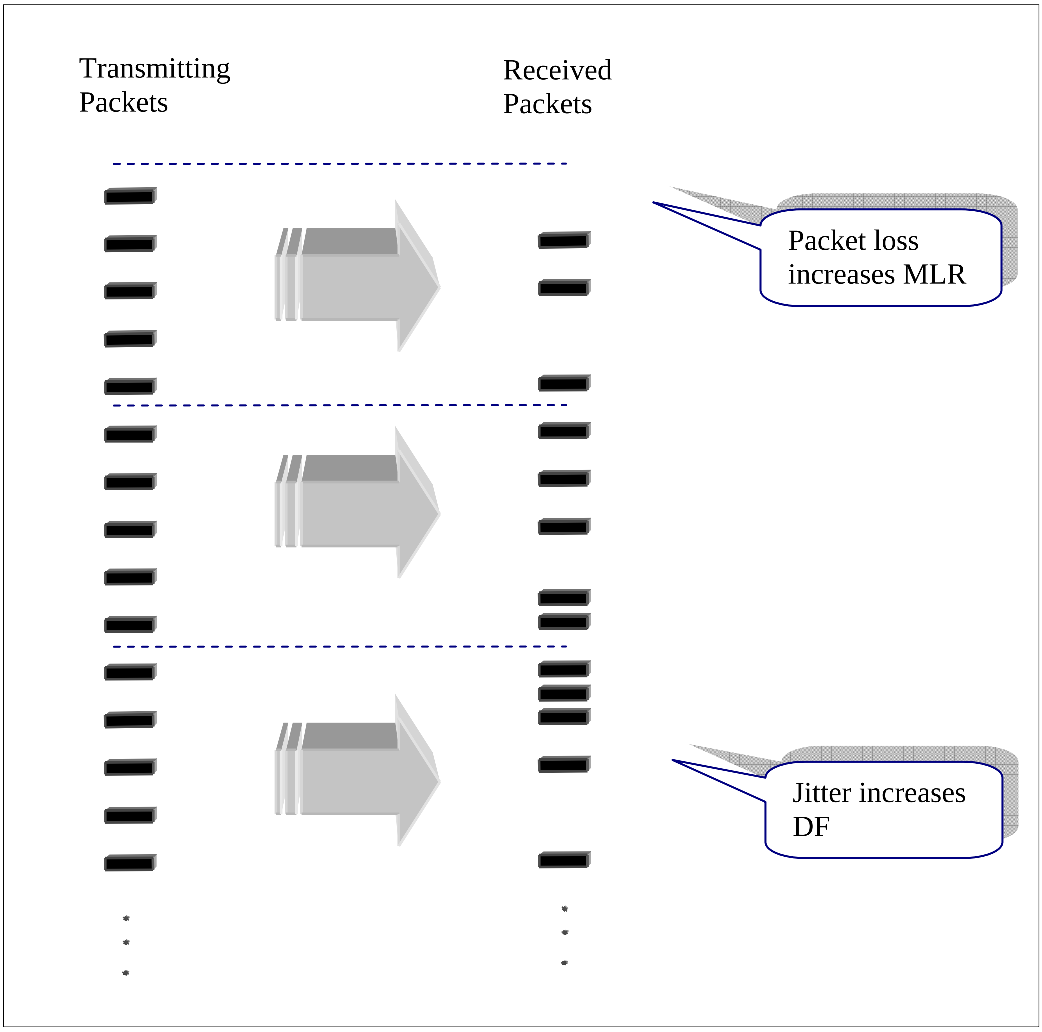

Figure 2-2 : Concepts of jitter and packet loss

2.3.7.4. MDI Reliability Discussions

MLR is a rate, and does not consider whether the lost packets are consecutive or not, which is an important issue in quality assessment. Considering the fact that the lost IP packet can carry important I frames or relatively less important B frames (see Section 2.5.2), implies that MLR correlates loosely with user-oriented video quality assessment. Actually MDI is more considered as a network evaluation tool or network visibility tool.

Another example that shows MDI may not correctly reflect the end user video quality is that the delivery of a video frame may suffer from packet loss, but loss–resilient transmissions using FEC and ARQ and error concealing codecs may retrieve the lost packets and correct all visual impairments. In such case MDI yet reports a bad quality due to the measured loss rate.

2.4. Protocols

Transportation of media and control data consists of encapsulation of the coded media and control data in a transport protocol (see 2.4.1) [14] with use of some other protocols according to the application. This is shown in the protocol stack of Figure 2-3.

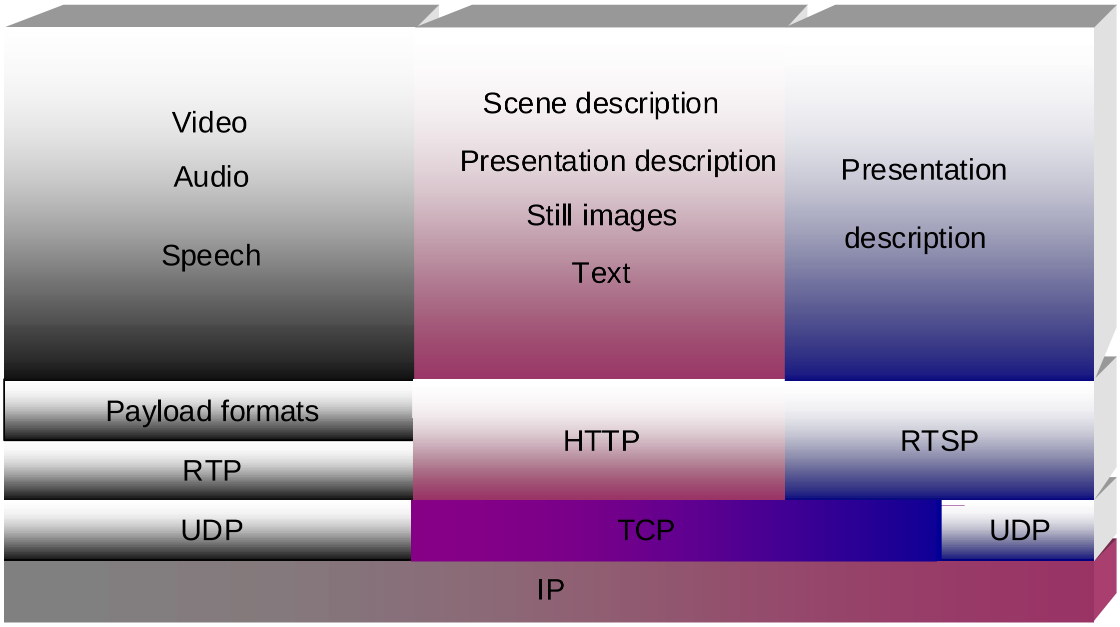

Figure 2-3: Overview of the protocol stack

2.4.1. Real Time Transport Protocol

Real Time Transport Protocol (RTP) is the Internet-standard protocol for the transport of real time data, including audio and video over UDP. RTP is defined by RFC 3550 [13]. The encoded media is encapsulated in the RTP packets. RTP header has presentation timestamp and sequence number. Presentation timestamp defines for the client that at what time the content of a particular packet should be displayed. It also synchronizes the audio and video. Sequence number in the header is used to detect packet loss [32].

The encoded video should be one of the specified RTP payloads (See 2.4.1.1). RTCP is also provided by RTP which gives feedbacks about the transmission quality [14] and information on the presence or leaving of participants in an on-going session. RTCP is over UDP/IP. RTP was originally designed as a multicast protocol, but has been also used in many unicast applications. Applications typically run RTP on top of UDP, to make use of its multiplexing and checksum services.

RTP in conjunction with RTSP is frequently used in media streaming.

2.4.1.1. RTP payload formats

For RTP/UDP/IP transport of continuous media the following RTP payload formats shall be used for Video [14]:

- MPEG-4 video codec . RTP payload format for MPEG-4 is according to RFC 3016 [15].

- H.263 video codec . RTP payload format for H.263 is according to RFC 2429 [16].
- H.264 (AVC) video codec .RTP payload format for H.264 is according to RFC 3984 [17].

2.4.2. Real Time Streaming protocol

Real Time Streaming Protocol (RTSP) is a session control protocol which sets up and controls the individual media streams and is defined by RFC 2326 [18]. It allows a client to remotely control a streaming server, issuing commands such as 'setup', 'describe' and 'teardown' [14].

The sending of streaming data itself is not part of the RTSP protocol. The streams controlled by RTSP may use RTP, but the operation of RTSP does not depend on the transport mechanism used to carry media. An RTSP session has a session ID and the server keeps track of the session by this ID. Thus there is no need to a permanent TCP connection and during an RTSP session, may many reliable or unreliable (TCP or UDP) connections be opened and closed to issue the RTSP requests.

2.5. Video Codecs

A video codec is a tool that does the digital video compression/ decompression. It usually utilizes a lossy data compression algorithm. There is no optimal compression method that could be applied to all kind of media. Even for the same media there might be different codecs according to the applications [32].

Here we bring a short description of MPEG-2 TS, MPEG-4 [19] and H.264 [20]. MPEG-4 basics on compression and estimations and then improved H.264 block estimation is brought here. H.263 [21] was designed for video conferencing and H.264 provides a significant improvement in capability beyond H.263.

2.5.1. MPEG-2 TS

MPEG-2 encoding is a standard for lossy compression of audio and video [22]. Here we briefly describe how an MPEG-2 encoded signal is packetized to construct the MPEG-2 Transport Stream (TS), because it is yet used with some developers, even if the video is not encoded by MPEG-2. Thus it is useful to distinguish the differences and not to get confused.

After encoding the media, the compressed content is organized into Elementary Streams (ES). Then ES is cut up and a header is added to construct Packetized Elementary Stream (PES). The PES header at least has a stream ID, the PES packet length and some other information.

Then PES is cut up and TS header is added to construct Transport Stream (TS). The TS is typically either 188 byte or 204 byte. By cutting off the long variable length PES packets to shorter TS packets of constant size, it is easier and faster to recover errors. In fact TSs are defined for transmission networks that have occasional transmission impairments .TS header consists of a synchronization byte, flags and indicators, packet identifier(PID) and other information for error detection and timing [23].

Thus MPEG-2 Transport Streams are composed of 188/204 Byte TS packets. Each TS packet has a 4 Byte header.

2.5.1.1. *MPEG-2 TS over IP*

Transmitting the TSs over IP can be directly done over UDP/IP. With considering the typical Ethernet Maximum Transmission Unit (MTU), normally 7 TS packets can be carried in an IP packet. Figure 2-4 shows TS/UDP/IP method.

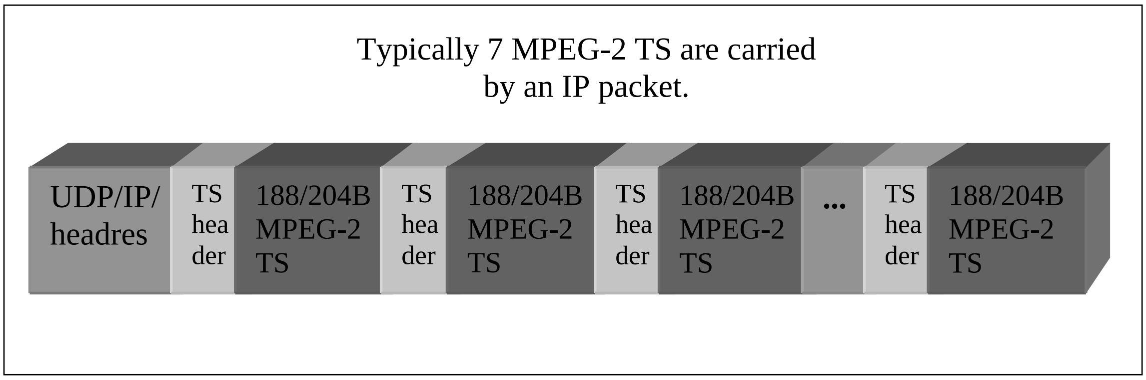

Figure 2-4: shows an IP packet carrying TSs, in TS/UDP/IP method

The second method carries MPEG-2 TSs over RTP/UDP/IP specified in RFC 2250 [24], which also transmits 7 TS packets over Ethernet based networks, and the method can be shown by TS/RTP/UDP/IP.

In the both of above solutions, sequential TS packets are carried in IP packets without any specific knowledge about the content of the packet.

The important thing to remember is that for IP transmission of media, which is compressed by other encoding schemes e.g. MPEG-4, one may use MPEG-2 Transport Streams or the so called TS/RTP/UDP/IP. This utilization of MPEG-2 TS for MPEG-4 compressed media has some drawbacks in comparison with the RTP payload format stated in 2.4.1.1, of which discussion is beyond the scope of this work and one can fine the details in [25].

2.5.2. MPEG-4

MPEG-4 is a standard to compress audio and visual (AV) digital data. MPEG committee developed MPEG-4 as 'ISO/IEC 14496' standard in late 1998 [26]. Streaming media, video telephony, and digital TV all benefit from compressing the AV stream by MPEG-4.

MPEG-4 is a very rich toolbox that targets a number of diverse applications. Each application has its own set of requirements. Thus most of the features in MPEG-4 are left to the individual developers deciding whether to implement them. This implies that probably the entire MPEG-4 set of standards have not been implemented for a specific application.

MPEG-4 consists of several parts, e.g. MPEG-4 part2 is a compression codec for visual data, and MPEG-4 part10 is Advanced Video Coding (AVC). MPEG-4 part2 is known as MPEG-4 Visual.

MPEG video coding makes use of similarity of subsequent frames in a video sequence.

Based on this concept of only sending the differentiated information; I,B and P frames are introduced:

Intra frame(I-Frame) is a complete video frame that is compressed without making reference to a previous or subsequent frame. An I frames can be decoded by its own.

Predictive frames(P-Frame) are motion predicted from the previous I or P reference frame, which means for decoding a P frame, the reference frame is needed that appeared at an earlier time instant in the sequence. If the reference frame is lost , then the P-frame can not also be recovered. This is known as forward prediction, for P frames are predicted from only the previous frames.

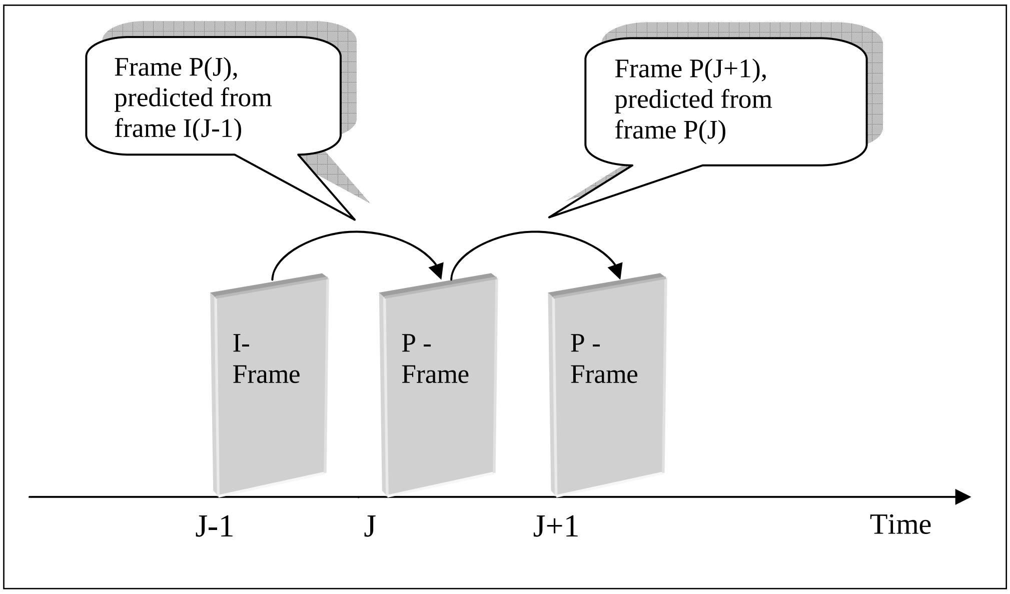

Figure 2-5: Concept of a P-frame

Bi-directionally Predictive frames (B-Frame) are bi-directionally motion predicted, from two or more reference frames. This means that decoding the B frame, needs an earlier I or P reference frame and a subsequent I or P reference frame , which are known as forward prediction and backward prediction respectively.

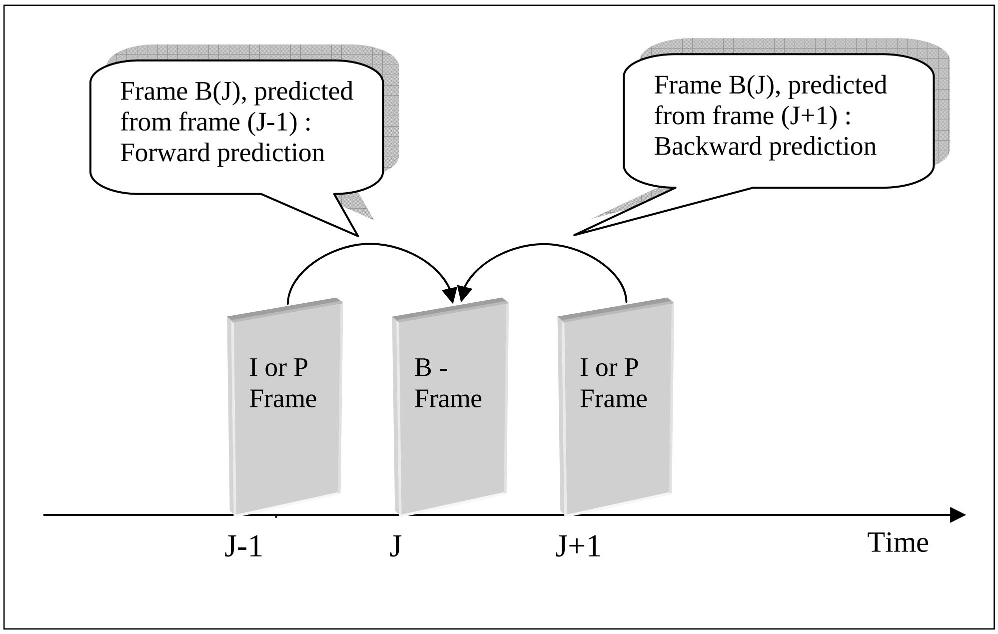

Figure 2-6: Concept of B frame

B frames are not used for predicting any other frame. If any of the two reference frames are lost the B frame can not be reconstructed. Since using B frames in coding introduces extra delay, it can not be used in conversational applications.
Order of delivery in a video sequence can be like IPBBPBBPBBIPBBPBB….

The frames are compressed by transforming spatial blocks of 8x8 pixels to frequency coefficients by use of block-based Discrete Cosine Transform (DCT).
The MPEG-4 compression schemes are defined in the ISO/IEC specifications 14496-2 [19] and 14496-3 [27].
For example if an uncompressed digital video stream has 830KB per frame , the ‘compressed’ I,P and B frame have approximately a size of 80KB,25KB and 10KB respectively. These compressed sizes vary due to the spatial and temporal complexity of the pictures in the video sequence.

To the compressed I , P and B frames, some headers are added to construct the Elementary Stream (ES). The header includes data for synchronization, identification and other source information. Then ESs are organized in Access Units (AU), which are the smallest elements with individual timestamps. Standard RFC 3016 [15] defines the MPEG-4 payload format for transporting over RTP/UDP/IP. The scheme of video transmission over RTP/UDP/IP is shown in
Figure 2-7. The payload length might vary, but the total IP packet size should not exceed the Maximum Transmission Unit (MTU) which is, e.g. 1500 bytes in Ethernet based networks.

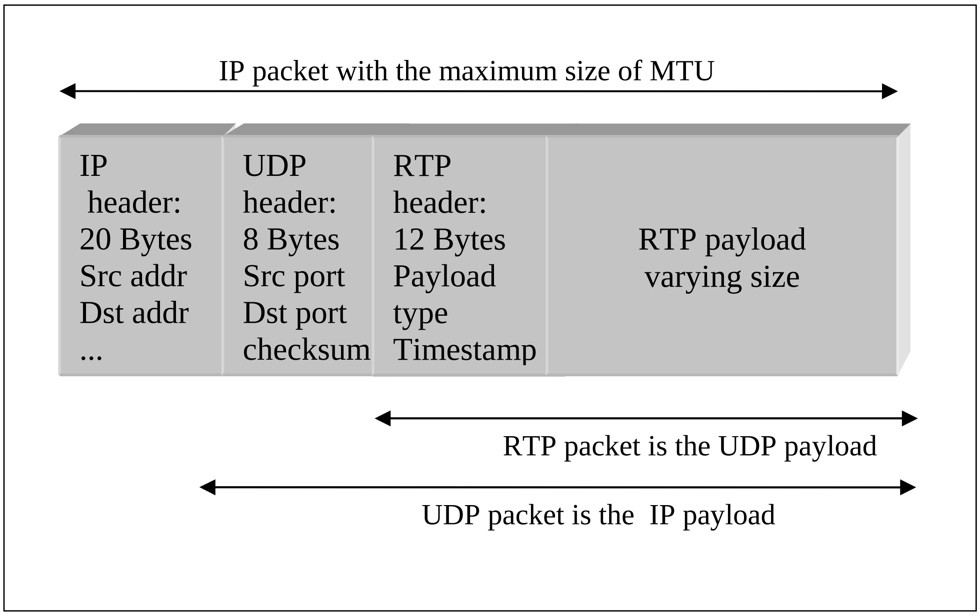

Figure 2-7 : IP packet encapsulating UDP /RTP packet.

2.5.3. H.264

ITU-T H.264 standard is technically identical to the MPEG-4 Part 10 Advanced Video Coding(AVC) [20]. It has been jointly developed by experts from ITU-T's Video Coding Expert Group(VCEG) and ISO/IEC's Moving Picture Experts Group(MPEG). It was targeted to provide good video quality at lower bit rates than the previous standards. Flexibility for different applications was another goal of designing H.264. It is built on the concepts of earlier standards such as MPEG-2 and MPEG-4 part2.

An H.264 encoder carries out prediction, transform and encoding processes.
A frame is divided to a set of variable block sizes and prediction of these blocks is formed based on the previously coded data. The prediction method is more flexible than those in previous standards. It is either inter prediction or intra prediction. Intra prediction, shown in Figure 2-8 uses blocks of surrounding previously coded pixels within the same frame, to predict the macroblock.

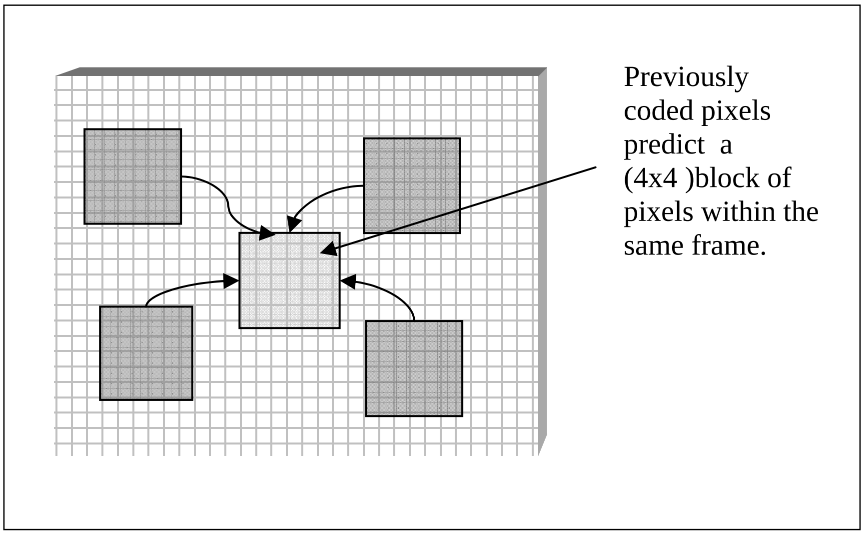

Figure 2-8 : Intra prediction

In contrast, inter prediction uses the previously coded pixels from the similar regions in previously coded frames. The variable block size enhances motion estimation precision. Figure 2-9 shows the idea of inter prediction with different block sizes.
The previously coded frames may be in the past or future (in time order) of the video sequence.

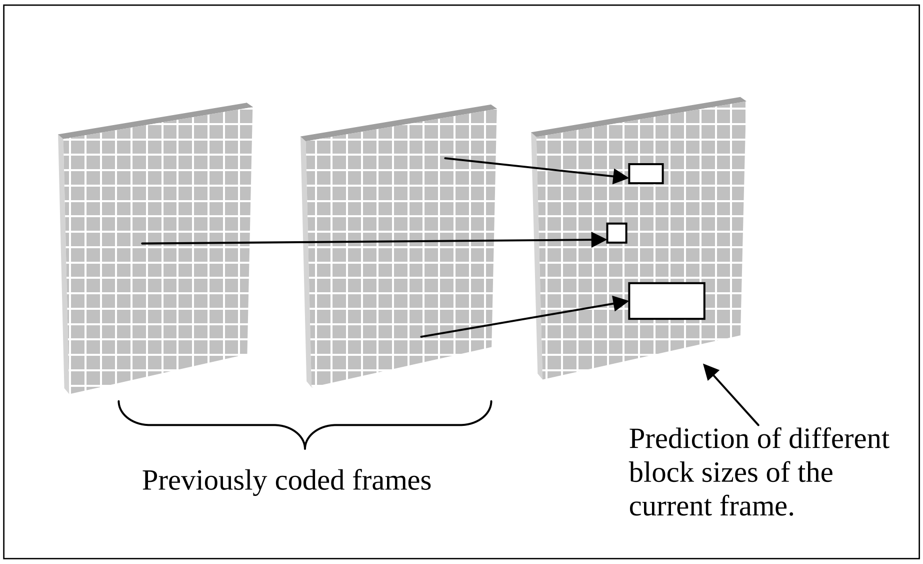

Figure 2-9: Inter Prediction

The encoder then subtracts the prediction form the original block to form a residual [28].
The residual samples are transformed by Discrete Cosine Transform (DCT).
A set of coefficients are the DCT transform output that are quantized in the next step.
All the above steps are performed by the Video Coding Layer(VCL).
The outputs of the VCL encoder are slices. A Slice is a bit string that contains the data of an integer number of macroblocks, and the slice header [17].
The next part of the encoder is called Network Abstraction Layer (NAL), which encapsulates the slices into NAL Units, which are suitable for transmission over packet networks [17]. The process of encapsulating NALs in an RTP packet is described in Annex B of H.264.
H.264, in comparison with the previous standards, can deliver better image quality at the same compressed bitrate or in other words, the same image quality with lower bitrate. H.264 was designed for mobile devices, but it has been standardized for wide range of applications.

Chapter 3

3. *Mobile TV and Mobile Phone Client*

In this chapter, Mobile TV as one of video over IP services, is discussed. Firstly some terminologies in the literature are introduced to avoid possible confusing between similar concepts. Then the end user media retrieval solutions, streaming client architecture and buffers are introduced. Finally the signaling between the end user mobile phone and the streaming server in the wireless network is drawn.

3.1. Mobile TV

Mobile TV is a multimedia service provided by an operator to its customers. The service gives the possibility of watching TV or TV like content on the subscribers' mobile phones [34]. The different streamed services are discussed in 3.2.
Ericson Mobile TV solution consists of two parts: Ericsson Content Delivery System (ECDS) and a dedicated client in the mobile phone [34].
However our discussion in this work is focused on the general parts and definitions of Mobile TV, so the Ericsson solution and its requirements are not discussed anymore, which can be found in more details in [34] and [32].

Video on Demand (VoD) is a service that allows subscribers to select and watch video content over a network. VoD can either stream or download the content (See 3.2).

Another common term in media networking is Triple Play. Triple Play describes three main services of Telephony, Television and high-speed Internet. The former is a narrow band service and the two latter are broadband services, while all are run over a single broadband connection. Thus triple Play stands for combined services including voice communication, Internet access and multimedia services like IPTV.

IPTV is viewing television contents by a computer network-based technology, e.g. a broadband connection.
Digital Video Broadcasting- Terrestrial (DVB-T) is the standard for digital media broadcasting for digital terrestrial televisions [29].
Digital Video Broadcasting-Handheld (DVB-H) is a technical standard to bring broadcast mobile TV services to mobile phones [30]. It is mostly based on DVB-T specifications, and adds some features for handling the mobile phone requirements such as battery limitation.
Multimedia Broadcast Multicast Service (MBMS) is a broadcasting service that can be offered via existing cellular networks. Thus the network infrastructure is already available [31].

3.2. End User Digital Media Retrieval ways

The end user may retrieve media content in one of the four ways offered by the streamed service: Download, Progressive Download, On Demand Streaming and Live Streaming.

3.2.1. Download and Progressive Download

By downloading the complete media content is downloaded to and saved in the end user equipment before, and then it is displayed. Therefore the transmission id free of loss but may suffer from long start-up time. Requirements on memory capacity in the end user equipment also should be considered [32].

The difference between download and progressive download is that in the latter, before storing the complete media content, the media presentation can start, so it has the advantage of shorter start-up time. The protocol is HTTP.

3.2.2. On Demand streaming and Live Streaming

The media is not stored permanently in the user equipment and after a short delay of buffering media is played by the client software as it is delivered. Since it is a real time service, there is no end to end retransmission between the streaming server and the user client , which may introduce packet loss and delays over a wireless network.

In On Demand Streaming service the media content is prepared and stored in the streaming server in advance. For Live Streaming, a real time source is feeding [32].

When PS access is used, variation in the transmission speed does not let have a steady content stream. Number of available radio channels, varying level of congestion in the network and retransmissions in the radio link due to packet loss, cause packet jitter [32].

As stated before, jitter is the variation of end-to-end latency in time.

To minimize this impact, the Quality of Service streaming class can be supported, which allows the operator to assign fixed radio resources to each user. The streaming QoS class is defined in [33].On the other hand there is the background or interactive QoS, commonly referred to as “Best Effort”. Control mechanisms give priority to the transport of media content with QoS streaming class, compared to other PS traffic with the QoS classes interactive and background [32].

Protocols for PS streaming are RTSP over TCP/IP, RTP over UDP/IP and RTCP over UDP/IP.

3.2.3. Mobile TV Session

A mobile TV session consists of a control session and a streaming session. Control session uses HTTP requests to send commands from the Mobile TV client to the server. The streaming session handles the actual media transfer [34].

3.2.4. Streaming Session

Delivering media from a server to a client over a network is streaming. It is real time and the media is not downloaded to the viewer’s hard drive [35].

A streaming session uses RTSP/RTP/RTCP to transfer media from the streaming server to the streaming client in the user equipment(UE). RTSP DESCRIBE is the message sent by the client to start the streaming session and RTSP TEARDOWN is the end message by the client to teardown the session [34].

3.3. Streaming Server

Darwin and Helix are two known streaming servers. Darwin Streaming server allows streaming media across internet using RTP and RTSP protocols. It is open source and capable of streaming different media types like H.264, MPEG-4 part2 and 3GP files.
Helix streaming server also delivers several media types. However for the 3GPP format files and mobile streaming applications the Helix Extension-Unlimited to Enterprise customers should be installed [36].
Helix has transmitting rate adaptation according to the received RTCP messages, when bandwidth limitations introduce excessive packet loss, but Darwin does not have such rate adaptation.

3.4. Mobile Phone Client

In order to have streaming service, the mobile phone should have a streaming client. Figure 3-1 shows architecture of a typical streaming client in the PS domain. The streaming client can request the streaming server to start, pause and stop sending the media content.

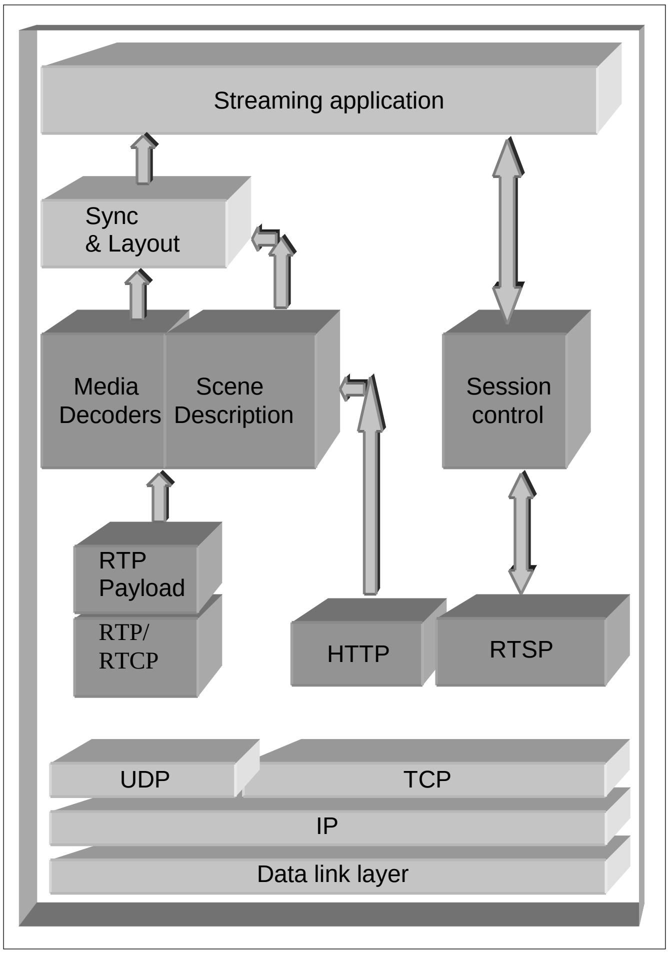

Figure 3-1: A Typical Streaming Client Architecture.

In the client some post processing and error concealing algorithms are implemented in order to retrieve the final media with a better quality. This feature is individual for every client, so the perceived video quality is very much dependent on the client that performs buffering, decoding and post processing in the UE.

3.4.5. Buffer in User Equipment

The client in the user equipment should have a jitter buffer and a pre-decoder buffer.
The jitter buffer is for compensating the jitter introduced in the radio and core network [32]. As described before, in a PS network, number of available radio channels and retransmissions in the radio link due to packet loss are of main causes of packet jitter.
The pre-decoder buffer needed for the variable rate encoding. The encoding rate varies due to the temporal and spatial complexity of the video sequence. Figure 3-2 shows that the video content is encoded with a variable rate, then the server transmits the data over the network with a constant rate. The data also experiences the network jitter and the received data should recover firstly from the jitter and then the pre-decoder buffer is applied. It is clear that even with unlimited bandwidth and no network jitter, the pre-decoder buffer is needed if the encoding is variable rate.
Choosing the buffer size is a trade off between startup delay time and assurance of having a streaming session without interrupts. Thus there is no strict rule for that, but a few seconds length can be a typical buffer size. From 3GPP Release 5 [14], there is an option that the buffer needed for the variation in content bitrate can be signaled [32].
The same buffer is used for handling both network jitter and content bitrate variation.

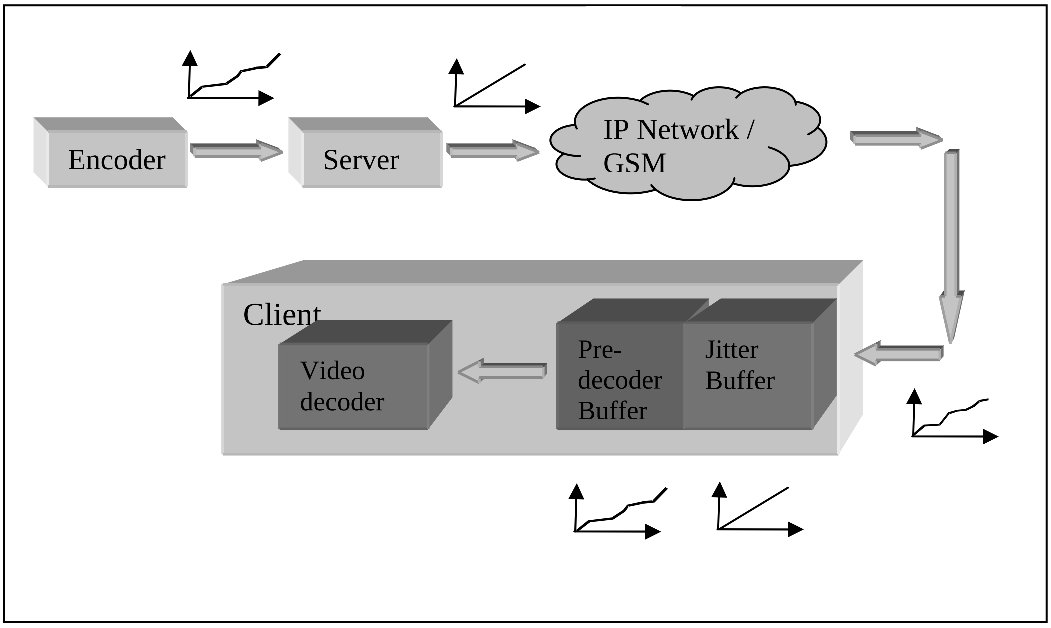

Figure 3-2: Jitter Buffer and Pre decoder Buffer, Packet Streaming through Wireless Network

3.5. Signaling Diagram

The signaling basics for a mobile TV service, can be as below in Figure 3-3 and Figure 3-4 . This is simplified as does not contain the browser in the UE client and also there is no QoS streaming class included. In case of QoS streaming class, before RTSP setup request a secondary PDP Context request would be generated from the UE side, and after some signaling similar to the primary PDP Context, the negotiated QoS would be assigned to the streaming.

Packet Data Protocol (PDP) Context is a data structure on both SGSN and GGSN which contains the user's session information, when the user has an active session.
GTP is the GPRS Tunneling Protocol. GTP Create (GTP-C) is for signaling between GGSN and SGSN. It allows the SGSN to activate a session on behalf of the subscriber. It also adjusts QoS parameters and deactivates the session.
BSS Packet Flow Context contains the BSS QoS profile that is created or deleted after the PDP Context Creation or deletion.

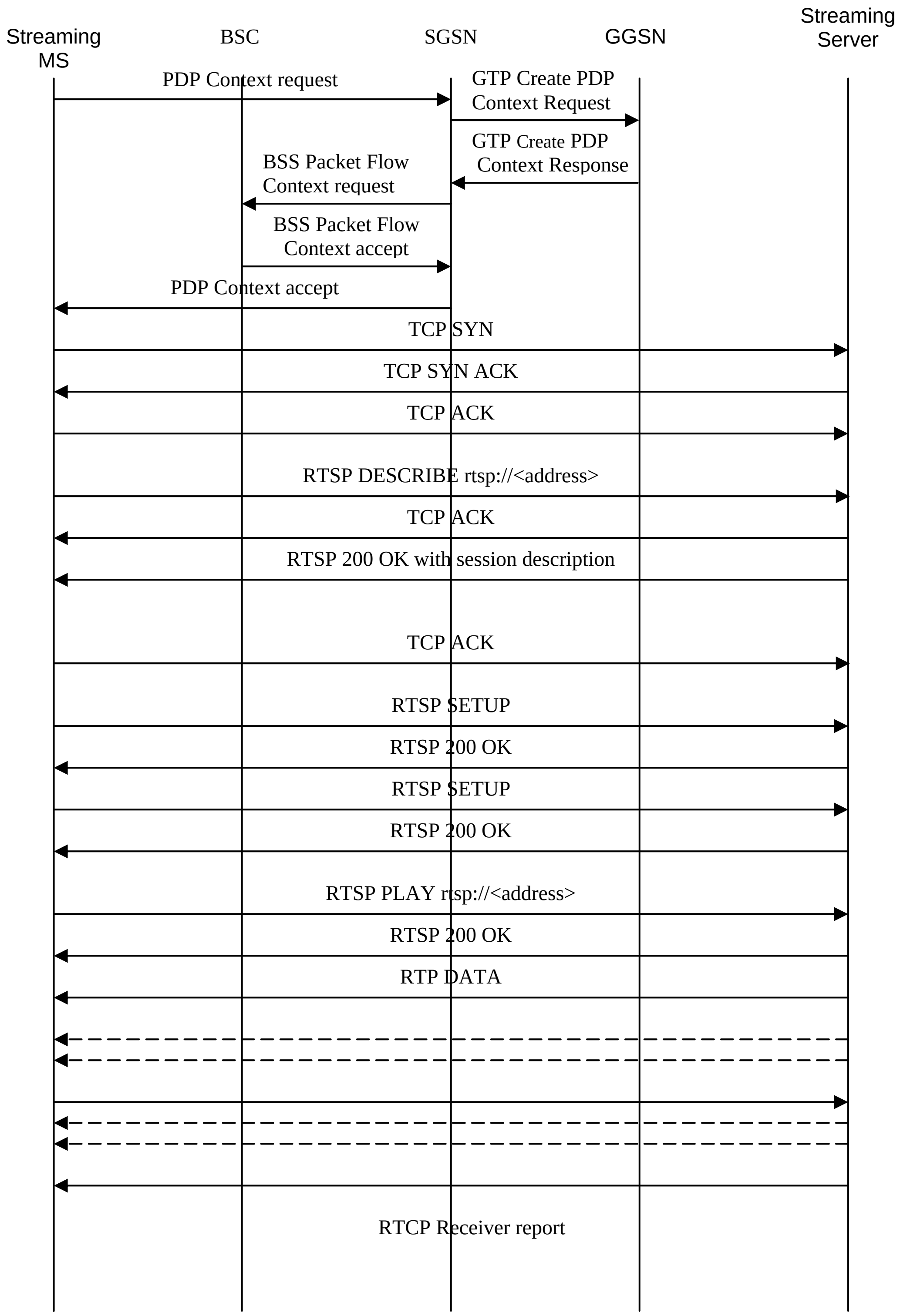

Figure 3-3 : Signaling diagram for streaming in a wireless network.

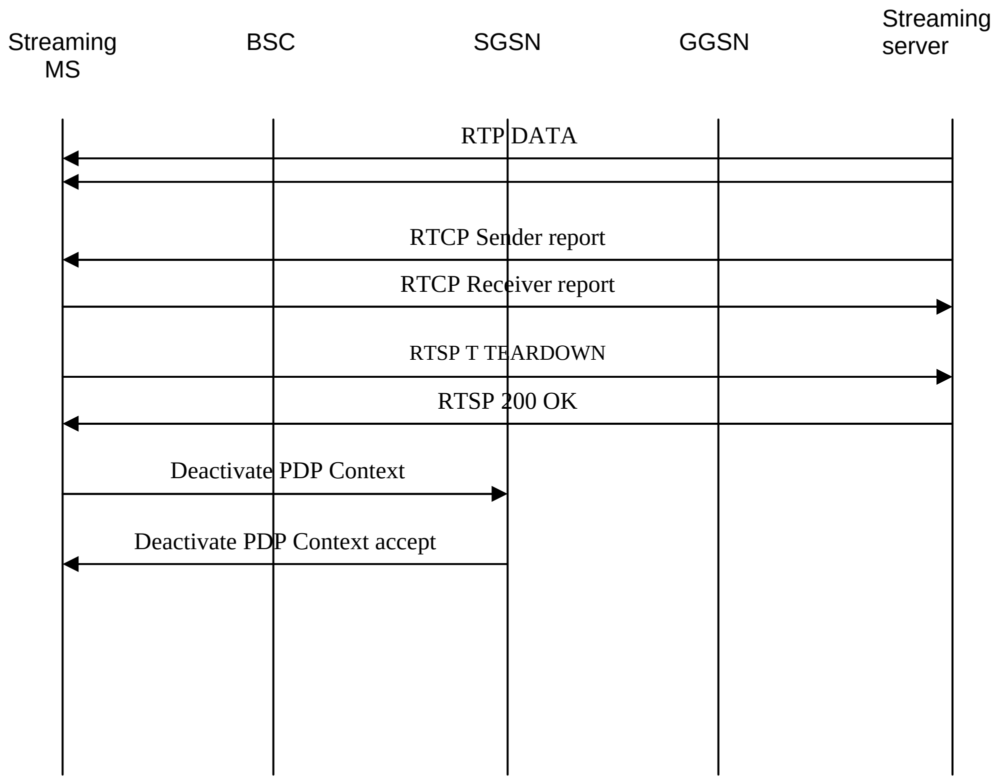

Figure 3-4: Signaling diagram for terminating a streaming session in Wireless Network.

These signaling can easily be tracked by a typical network protocol analyzer like Ethereal.

3.6. File Format

A file format contains data in a structured way. 3GP is the 3GPP file format, defined in TS 26.244 [37], and can contain timing, structure and media data for multimedia streams. MMS and PSS use it for timed visual and aural multimedia.

3GP is designed to decrease storage and bandwidth requirements in order to accommodate mobile phones.

File extension for a 3GPP formatted file is '.3gp' . The registered video codecs that can be included in 3GP files are H.263 [21], MPEG-4 [19] and H.264 [20].

Chapter 4

4. Overview of Several existing Quality Measure Softwares and the Related Vendors

In previous chapters, we studied Video Quality Methods and Metrics, Video over IP Network Requirements, Protocols and Standards, Mobile TV services and the Mobile Phone Clients. Now we are ready to go through the existing Video Quality Measurement Tools and Solutions among different vendors. In this chapter, the study of the vendor is limited to the related video tool. Based on previous chapters, a very short description of the tool features comes to the conclusion of whether the tool satisfies our objectives or not. Among the main vendors, providing video quality measurement tools; Opticom, Agilent Technologies, Spirent Communications, Ericsson Research, Ixia, Radvision, Shenick and Tektronix were studied thoroughly or partially in this work. Finally the more relevant tool to our objectives is chosen.

4.1. Opticom GmbH

Opticom video quality measurement algorithm is named PEVQ which stands for "Perceptual Evaluation of Video Quality". PEVQ is a Full Reference, intrusive method which provides Mean Opinion Score (MOS) estimates of the video quality degradation due to coding and compression. The method makes use of the perceptual parameters in the video signal.

Basic structure is shown in Figure 4-1, which includes 4 blocks:

- Pre processing, which aligns each distorted frame to the corresponding original frame.
- Calculating the perceptual differences of the aligned signals.
- Classifying the calculated indicators to detect distortions.
- Forming the MOS, on a range from 1 to 5; very bad to excellent quality respectively.

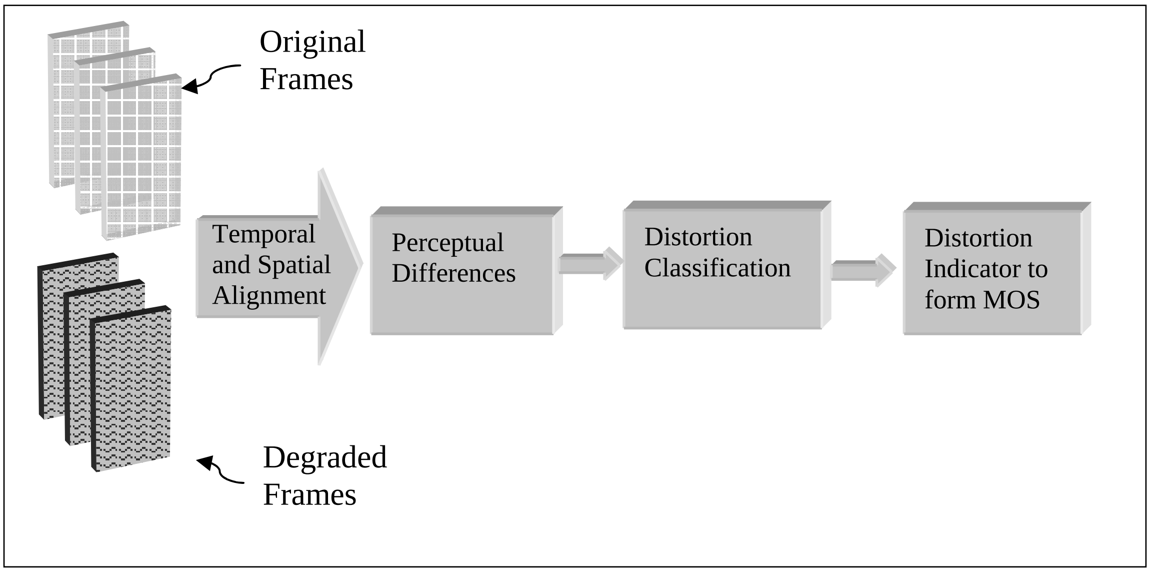

Figure 4-1: PEVQ Basic Structure

PEVQ performs a quite powerful FR measurement and is suitable for a detailed study of compression schemes and their impacts; like blurring, distortion and jerkiness on each an every video frame. Although it provides some packet level information (loss, jitter), PEVQ is not fully suitable for our requirements, or to say its main functionality is not of our interest. More detailed information on basic structure, key features, inputs, outputs and complexity is in [2] and [4].

4.2. Agilent

Agilent offers Triple Play Analyzer (TPA) software to monitor, analyze and troubleshoot voice, data and video services over IP networks, including IPv4 and IPv6.
TPA evaluates video quality degradation by No Reference and Non-Perceptual method.
The method is a proprietary algorithm, and gives the video MOS degradation by use of neural networks and trained classifier. The existing TPA is trained for typical IPTV frame sizes, and bit rate, thus not suitable for MobileTV frame resolution which is QCIF.
TPA provides I,B and P frame statistics, VoIP MOS, and a lot of IP level and MPEG level statistics. MDI in terms of DF and MLR , Group Of Pictures (GOP) pattern, Average jitter, packet loss, throughput and ZAP response time are examples of the outputs and graphs presented by TPA. ZAP time or channel switching time is of special importance and is one of Key Performance Indicators (KPIs) for IPTV.
The existing TPA is designed for video files that use MPEG-2 transport stream (MPEG2-TS) for IP transmission (See 2.5.1.1). However RTP payload format standards, for MPEG-4 and H.264 as stated in 2.4.1.1, do not use MPEG-2 TS anymore.
For perceived video quality, TPA software draws a graph of video degradation MOS, instead of assigning absolute scores (ranging 1 to 5) to the clip. This arises a discussion that how much this MOS degradation values can be reliable, when according to NR method there is not any reference image to compare with and extract the degradation.

4.3. Spirent

Spirent Communications offers video quality measuring tools in two methods.
Spirent provides IPTV Full Reference video quality measuring software as an option for Spirent's Abacus 5000 product [38]. The perceptual Video Quality Measurement (VQM) is based on ITU J.144R standard [9].
VQM outputs are video MOS rating on a scale of 1 to 5 and several other video degradation metrics that some of them are brought in 2.3.5.1.

Spirent uses no reference video quality solution provided by Telchemy, known as Telchemy Video Quality Metrics (TVQM). TVQM technology will be available on Spirent Video Quality Analyzer (VQA) test tool and the release is planned to Q2 2008. The measurements done include network metrics, MDI , video MOS , H.264 , and video quality measurement in presence of encryption.

4.4. Ericsson Research

Ericsson has developed Video Streaming Quality Index (VSQI) algorithm to assess video streaming quality [39]. VSQI a is No Reference objective method applied in TEMS Investigation tool and gives a MOS value ranging from 1 to 5 to both video and audio quality. There are two versions of VSQI : static and dynamic. The former assigns one MOS value to the entire clip as the overall multimedia quality. The algorithm is optimized for clips of 30 second length. The latter is a real time estimation of played video clip, roughly every second. The dynamic VSQI declines very fast; if Rebuffering occurs during the clip playback. The viewers are much less tolerant to rebuffering in comparison with initial prebuffering.

The tool is connected to the mobile phone by USB port, and the mobile phone is just acting as a modem to retrieve video streams from the wireless network. The buffering, post processing and decompressions of the received signal are all done by TEMS Investigation software. Thus it tries to emulate the mobile phone clients.

However the buffering behavior of different clients is application specific, which has major effects on the perceived quality. TEMS Investigation software replaces all the different clients in mobile phones. Thus in different scenarios, the same algorithm is used, though it may not be exactly the same as any of the existing mobile clients.

VSQI is tuned for QCIF video resolution (176x144 pixels).

4.5. Ixia

Ixia provides IP performance testing tools. In study of those Ixia tools which provide video quality measurements ,two methods are pinpointed:

- Full Reference video quality analysis, using PEVQ provided by Opticom GmbH [40].
- Transmission quality assessment using MDI, and other packet level statistics [41].

Obviously our objectives are aimed to transmission metrics, but as discussed before; MDI value is not that much reliable for end-user quality tests. Although Ixia tools provide measurements including throughput, latency, jitter, MDI and MPEG level statistics, they do not provide a Mean Opinion Score (MOS) of the end user video quality in NR transmission network tests [42], which is quite vital for end-user test cases.

The Ixia tools are more suitable for emulating load tests, where video servers and thousands of subscribers watching IPTV and requesting VoD are emulated, and benchmarking the network and video server performance. More information on Ixia IPTV tools can be found in [43] which implies the tools are more appropriate for IPTV load tests rather than Mobile TV.

4.6. Radvision

Radvision provides Prolab Testing suite; an automated testing and validation solution for voice and video over IP , IMS and 3G networks. The entire network system is emulated by

Prolab. It gives a full reference perceptual video quality measurement on a 1-5 MOS scale. It uses a database of subjective experiments, to make the algorithm based on human perception models [44].

4.7. Shenick

Shenick provides converged IP network test and monitoring systems. For video quality tests, it uses Opticom PEVQ solution [45] as a Full Reference, perceptual method for a frame by frame robust quality measurement which mostly reflects the coding impacts. More information on Shenick IPTV test tools can be found in [46]and [47] which states the tools are more appropriate for IPTV tests rather than Mobile TV.
Shenick also provides no reference MOS value using Telchemy's TVQM technology. Both the above methods are implemented in Shenick ServiceEye tool [48].

4.8. Tektronix

Tektronix offers a full reference image quality Analysis tool. The system is PQA500 which is ideal for optimal codec design [49]. It predicts Differential Mean Opinion Score (DMOS). Another tool by Tektronix is MPEG Test Systems-MTS400 series that provide problem diagnosing in the network environment, including transmission links like IP networks/RF or content processing like Transport Stream (TS) layer. MTS400 is a fault analyzer of a variety of codecs, designed for Digital TV debugging [50].

4.9. Table of Vendor tools

The above test tools have various features and a lot of detailed and different specifications. Table 4-1 summarizes the tool name, the technology or method used for video quality assessment.
Some Vendors have the proprietary algorithms for evaluating video quality, and use a term as a trademark, like PEVQ and TVQM .
The video quality measurement tools are usually trained for optimized performance for a set of specific frame resolutions, rate or clip length . For some of the tools these information are provided in the 'comment' column, from either datasheets or the meetings we could arrange with some of them.

VENDOR	Tool	Technology for MOS	FR/ NR	Frame rate/s	Comments
Opticom	Opticom OPERA	PEVQ	FR	30,25, 15,12.5 5,8,2.5	QCIF, CIF, VGA
Ixia	Aptixia IxLoad	PEVQ	FR		
	Aptixia IxLoad	TVQM	NR		
Agilent	DriveTest	Proprietary algorithm	FR,		
	Triple Play Analyzer (TPA)		NR	23.97, 29.97, 59.94	MOS Degradation value every 4 second, MDI ,IPTV ZAP time, Min resolution: 320x240 max resolution: 1920x1080 recording
Shenick	ServiceEye	PEVQ	FR		
		TVQM	NR		
Radvision	Prolab Testing Suite		FR		3G
Ericsson	TEMS Investigation 8.1	Dynamic VSQI	NR	5,8,10, 12, 15, …	QCIF, Real time, Recording KPIs available for GSM, WCDMA Networks
		Static VSQI			QCIF, about 30 sec length video clips
Tektronix	PQA500		FR		Differentiated MOS
Spirent	Video Quality Analyzer (VQA)	TVQM	NR,		
	Abacus 5000	VQM	FR		

Table 4-1: Table of tools

4.10. Choice of the Test Tool

Obviously the solutions with No Reference method are those we should choose the tool from. As discussed before, video quality measurements are highly dependent on the application and the mobile phone client. We are interested in GSM mobile TV services. Thus test tools that are designed for IPTV applications, though provide a complete set of scores and parameters, can not be applied to small size mobile TV frames.
On the other hand, TEMS Investigation 8.1 uses a built-in algorithm to decode and perceive video signals instead of the mobile phone client. The software is designed to simulate typical mobile phones' clients behavior. Thus the test results are independent of the mobile phone client.
Considering the above discussions and also the end user video tests as our objectives, finally TEMS Investigation 8.1 was chosen. In the next chapter we will run some test scenarios using this test tool.

Chapter 5

5. Video Quality MOS Tests

Based upon the study on the existing test tools in Chapter 4, TEMS Investigation 8.1 is selected to run some tests with. The tool measures video MOS changes due to network emulated impairments.
It is almost obvious that no two different test tools give identical video scores of the same network conditions. Thus the start point is to rely on a selected test tool MOS scores, to get an idea of network impacts on the video in real implementations. In other words, in this chapter we wish to observe; according to TEMS tool, which network conditions can really impact video quality and possibly to what extend.
The chapter starts with a description of test scenario, IP network emulation and measurable KPIs by TEMS tool. Then tables of test files and test cases, followed by graph results is presented.

5.1. Test Environment

The tests are done in an EDGE network, BSS release is R.07B. TEMS Investigation ver.8.1 is the tool which does the video quality and network performance analysis. IP Network emulation is done by Netem (See 5.2.6.1). Netem machine is placed in two different points of the network.
The typical video frame size in GSM Mobile TV is QCIF , which is 176 × 144 pixels.
A set of QCIF video clips with the same scene, but different rates and codecs, provided by TEMS are under test.

5.2. Test Scenario

Figure 5-1 depicts the typical video streaming scenario in EDGE Network [51]. Air interface is simulated by coaxial cables and splitter. The splitter attenuates the signal strength to a typical value of -73 dbm and the air interface has no impacts on the data, so IP network emulation effects can be measured correctly.
The Netem machine, which does the network emulation was firstly placed on Abis over IP interface; between BSC and BTS. Secondly Netem machine is placed between the streaming server and the switch in the IP network, where the core network node; GGSN is connected to the same network by Gi interface.

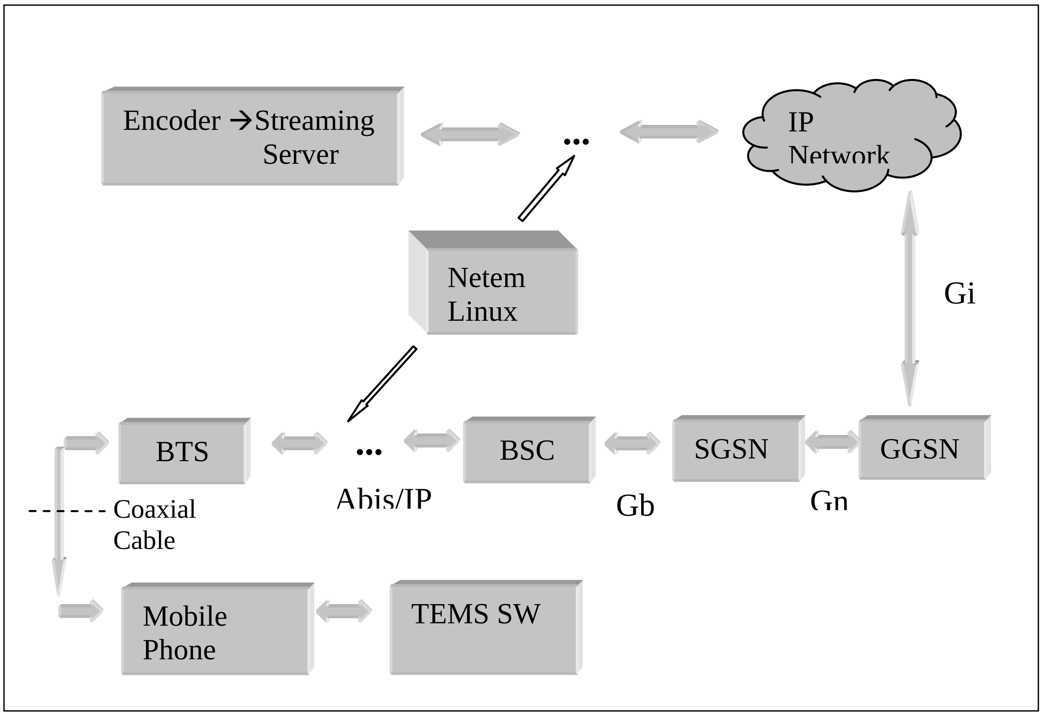

Figure 5-1 : Streaming Scenario in EDGE Network, where Netem machine was placed in either of the interfaces.

The “mobile TV client” part consists of a PC and a TEMS modified mobile phone. The PC or laptop is connected to the mobile phone via USB port. The mobile phone, which in our tests ,is Sony Ericsson-W600i should be TEMS modified in order to be able to capture Key Performance Indicators (KPIs) (See 5.3.). It is connected to the network through the splitter and the coaxial cable.

5.2.6. IP Network Emulation

For emulation the IP network impairments, the typical parameters; delay, jitter and loss, were changed by use of Netem, in the selected points of the network.
It should be noted that intelligent switches and routers, and advanced routing algorithms, all make the emulation of a real IP network more complicated.

5.2.6.1. Network Emulator

NETwork EMulator (Netem) is a is a Linux queuing discipline that can be instructed to impair streams of packets, which in turn allows emulating the properties of wide area networks. Netem is controlled by the command line tool ‘tc’ which is part of iproute2 package of tools [52]. The Netem machine has got two Network Interface Cards (NICs), and by installing the bridge package can act like a bridge.
Netem can also emulate a busrty case, where if a random loss or jitter occurs, the probability of next packets’ loss or jitter will be increased by a pre-defined value in percentage , which has to be set in the Netem configuration.

5.2.7. TEMS Investigation 8.1 Tool

The tool, which was ultimately chosen to be used for video streaming tests, as stated before is the TEMS Investigation 8.1 software, which captures, post processes and scores video streams, It also does KPI measurements and a lot of other uplink and downlink radio measurements. The latter is not of our interest, but for KPIs, we should know TEMS measures which KPIs and according to what standard or definition. Thus the corresponding section is extracted form the product datasheet [10].
TEMS Investigation tool communicates with the streaming server through the mobile phone which is acting like a modem.

5.3. KPIs in Packet Switched Network

Key Performance Indicators (KPIs) show how the network is performing according to certain parameters. In TMES Investigation 8.0, KPIs which are calculated for streaming service are divided in two categories of Service Independent and Streaming Specific KPIs. The former are calculated for any kind of service (FTP, HTTP, WAP) and the later are defined for a streaming service.
Figure 5-2 depicts a streaming session and shows some of the KPI definitions versus time and the trigger points for KPI calculations. Here we bring definitions of all the KPIs that TEMS measures, regardless of whether the network emulation settings could affect them or not. The definitions according to [10] are based on ETSI TS 102 250-2 v1.4.1.

5.3.1. Service Independent KPIs

-Attach Setup Time (sec)
is the time taken to attach to the GPRS PS network, and is the period between attach request message and attach accept message.

-Attach Failure Ratio (%)
Show the probability that the user fails to attach to the GPRS PS network.

-PDP Context Activation Time (sec)
Denotes the length of the time to activate a PDP context, which is the time difference between the PDP Context Activation Request time, by the phone; and the PDP Context Activation Accept message by the network.

-PDP Context Cut-off Ratio (%)
Is the probability of deactivation of PDP Context, with no customer intention .
It may be caused by SGSN failure or GGSN failure.

5.3.2. Streaming Specific KPIs

-Service Access Time (sec)
Is the time duration between the stream requesting from the phone (RTSP setup) and the first stream data packet (RTP data packet) received by the phone.

-Service Non-Accessibility (%)
Denotes the probability that after RTSP setup sent to the streaming server, the first RTP data packet can not be received by the phone.

-Streaming Quality
Is the Mean Opinion Score (MOS) of the reproduced video quality, assessed by VSQI algorithm, It takes into account both video and audio perceived quality and gives a static score for the whole video clip ranging from 1 to 5, scaling from bad to excellent respectively. VSQI is a proprietary algorithm.

-Reproduction Start Delay(sec)
Is the time interval between reception of RTP data packet by the phone and starting of reproduction of the stream by the phone.

-Reproduction Start failure Ratio(%)
Is the probability of unsuccessful stream reproduction.

-Reproduction Cut-off Ratio (%)
Denotes the probability of unintentional cutting off the reproduction, after it has started successfully.

According to Figure 5-2, a network emulation applied to the interface between Streaming server and GGSN , can not have any impacts on service independent KPIs. The same is for Abis/IP network emulations that can by no means affect the GTP Create PDP Context Request. However the video quality MOS is of most importance, and in the following sections the main discussions are focused on VSQI score, as a service dependent KPI, rather than other KPIs.

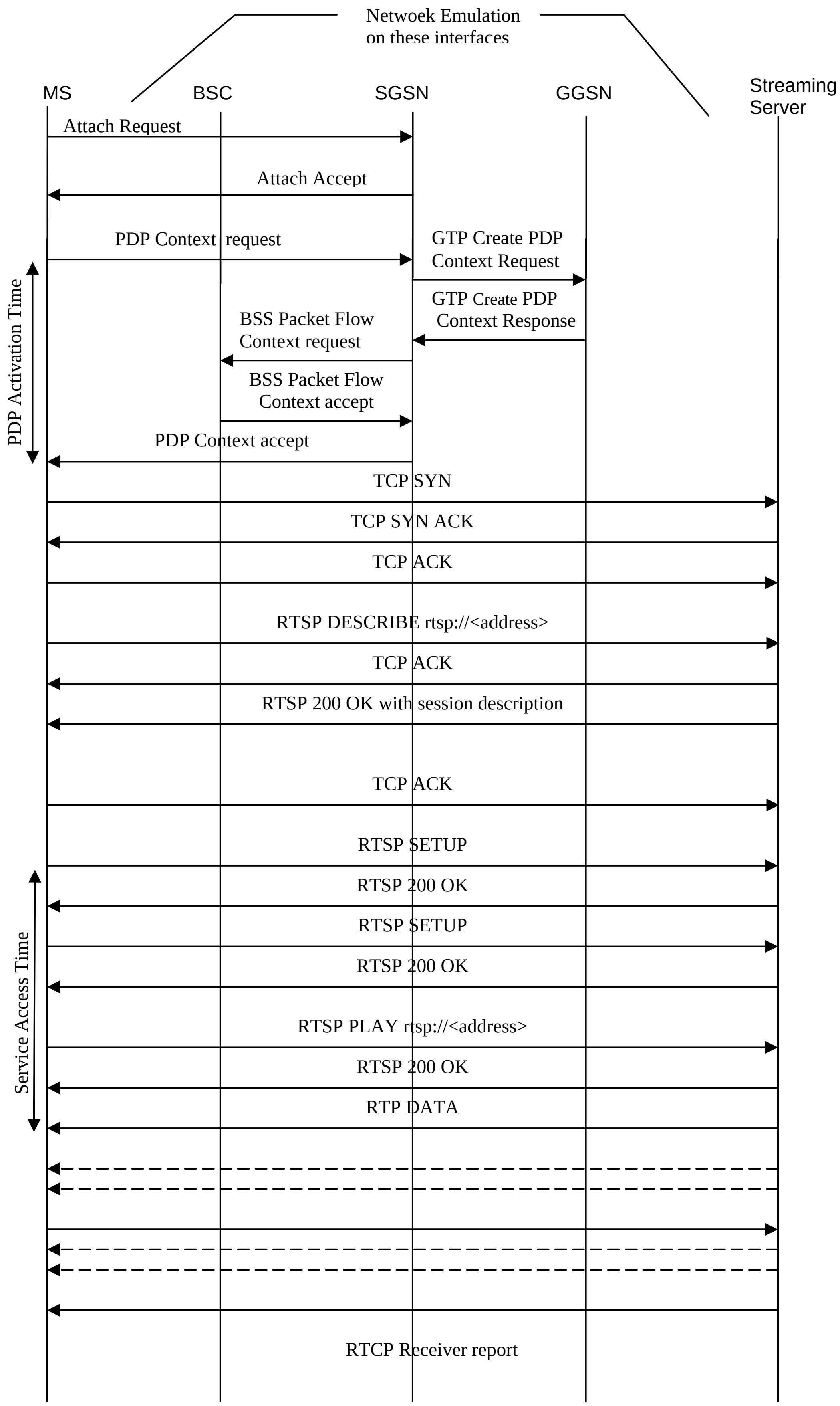

Figure 5-2 : Signaling Diagram for Streaming

5.4. Network Emulation on Abis/IP Interface

The Netem machine is placed between BSC and the BTS. A set of typical delays, jitter and packet losses are applied to the interface.
The PDP Context Activation Time and Attach Set up Time experiences higher delays.
Thus service independent KPIs get affected.
However due to the Radio Link Control/ Medium Access Control protocol (RLC/MAC) retransmissions, all the packet loss in the link between BSC and mobile phone is recovered. The prebuffer in the mobile phone client (See 3.4.5) can easily compensate for the delay caused by retransmissions or jitters. Thus the video quality score, perceived by the TEMS software does not show any destructions.
This implies that the perceived video quality is not impaired by jitters or packet losses that occur any where between the BSC and the mobile phone.

5.5. Network Emulation on GGSN- Streaming Server Interface

In this case, the Netem machine affects packets that exit the streaming server and are sent to the GGSN. Thus according to Figure 5-2 only streaming packets are affected and the PDP Context Activation Time or Attach Setup Time experience the undisturbed network.

Table 5-1 shows the specifications of video clips used in tests; such as video bitrate and codec. As stated before the scene and the video length time are same for all the clip.
Table 5-1, last column, gives the clean MOS of each clip before transmission.
For all the clips, MOS is degraded from 5 because of inevitable codec impacts on the video quality. The lower bitrate codec, the lower value for Clean MOS is achieved. These codec affects are irreversible, and the due to network conditions the perceived video quality is always lower than the Clean MOS.

No. of File	Reso lution	Clip length (sec)	Total Bitrate (Kbit/s)	Video Rate (Kbit/s)	File Format	Codec	no. of Frames /s	Clean MOS
File 01	QCIF	33 sec	20kbps	15kbps	3gp	MPEG4	5 fps	1.88
File 02	QCIF	33 sec	27kbps	22kbps	3gp	MPEG4	5 fps	2.13
File 03	QCIF	33 sec	27kbps	22kbps	3gp	H.263	5 fps	2.01
File 04	QCIF	33 sec	33kbps	28kbps	3gp	MPEG4	8 fps	2.32
File 05	QCIF	33 sec	48kbps	36kbps	3gp	MPEG4	8 fps	2.72
File 06	QCIF	33 sec	48kbps	36kbps	3gp	H263	8 fps	2.57
File 07	QCIF	33 sec	68kbps	56kbps	3gp	MPEG4	10fps	3.13
File 08	QCIF	33 sec	84kbps	72kbps	3gp	MPEG4	10fps	3.37
File 09	QCIF	33 sec	102kbps	90kbps	3gp	MPEG4	12fps	3.58
File 10	QCIF	33 sec	136kbps	112 kbps	3gp	MPEG4	12fps	3.84
File 11	QCIF	33 sec	174kbps	150 kbps	3gp	MPEG4	15fps	4.00

Table 5-1: Details of Video Clips under test: video bit rate/ codec, clean MOS before transmission

In Table 5-2 a set of network emulations and tests run on the network are brought. Table 5-2 summarizes the varying values of parameters applied to the network which includes delay, jitter, packet loss and change in mobile client prebuffer and rebuffer. A short review on the cases is as follows:

The prebuffer in the mobile client can be set in TEMS Investigation Software, a value between 1 to 20 seconds. The larger prebuffer, the more start delay the user experiences before playback of the video, but on the other hand less destructions on the video clip may be experienced due to the large prebuffer. A typical value of 4 second prebuffer is selected for all the tests, except for Case1 and Case2. Case1 examines the video quality under smaller values of prebuffer, and a relatively large packet loss. Case2 does the same test, but with a much lower packet loss.
Thus study of these two cases provides an idea of relation between the prebuffer length or initial playback delay and the packet loss rate.
Rebuffering may be used during the clip replay. In case that the network conditions were so harsh, that the client run out of buffered data, it halts the clip replay. The rebuffer starts to buffer packets before playback can continue. This freezing time in the middle of video clip is much less tolerable by the viewer, so a value of 2 second is chosen for almost all the tests.

Setting a typical value for the whole network delay and jitter, can be around 200 ms delay and 30 ms jitter. This means the packet delay average is 200ms with the standard deviation of 30 ms. By default Netem chooses a uniform random distribution for the jitter. Netem also has defined some other distributions like normal and pareto distribution.
Since network jitters are not purely random and are somehow correlated, we also consider 30% probability of dependency of a jitter value to the jitter value of previously received packet.
So next jitter value = 30% dependent to the previous jitter value + 70% random value form the uniform distribution. In this way Netem tries to approximately emulate the correlation and bursts.

Cases 3 to 8 are focused on packet loss effect when the above typical values of delay and jitter are set. Packet losses of 2% to lower values of 10^{-4} % are studied on almost all the video clips presented in Table 5-1.
Now it comes to study the jitter effect, when the packet loss zero. Case9 gives the video quality under a network that does not have any packet loss and jitter, but the typical delay is applied. It seems obvious that pure delay may not disturb the quality, unless it is lower than the prebuffer length. Case9 proves it; pure delay causes no MOS degradation.
But what about different jitter values?
Case10 includes several sub-cases. Jitter values of 5,10,15,20,30,40,60,80,100,150,180 milliseconds are applied to a network with zero packet loss.
In all the above cases, there is no bandwidth limitation in the IP network, in comparison with the video clip bitrate ranges. But when it comes to considering the radio channel, the modulation and coding schemes apply bandwidth restrictions. However in this work, IP networks are under study, and radio channel is considered to be perfect.

	Video clips	Delay(ms)	Jitter (ms)	Loss %	Correlation%	BW Limit	Prebuffer(s)	Rebuffer (s)	KPI	Packet capture
Case1	File 10 File 11	200	30	2	30	-	1,2,3,4	1,2	+	+
Case2	File 10 File 11	200	30	10^{-4}	30	-	1,2,3,4	1,2	+	-
Case3	File 01 – File 11	200	30	2	30	-	4	2	+	+
Case4	File 01 – File 11	200	30	1	30	-	4	2	+	-
Case5	File 01 – File 11	200	30	10^{-1}	30	-	4	2	+	+
Case6	File 07 – File 11	200	30	10^{-2}	30	-	4	2	+	-
Case7	File 01 – File 11	200	30	10^{-3}	30	-	4	2	+	+
Case8	File 01 – File 11	200	30	10^{-4}	30	-	4	2	+	+
Case9	File 11	200	0	0	0	-	4	2	+	+
Case 10	File 11	200	5,10,15, 20,30,40 60,80,100,150,180	0	30	-	4	2	+	some

Table 5-2 : List of all test cases, and parameters run on the EDGE network.

Figure 5-3 shows a sample of test results. A video clip under case5-network settings is received and shown by the TEMS software. Variations in dynamic VSQI real time score are subject to packet loss and jitter. Each case is iterated hundreds of times, and the reports are extracted from hundreds of graphs similar to Figure 5-3.

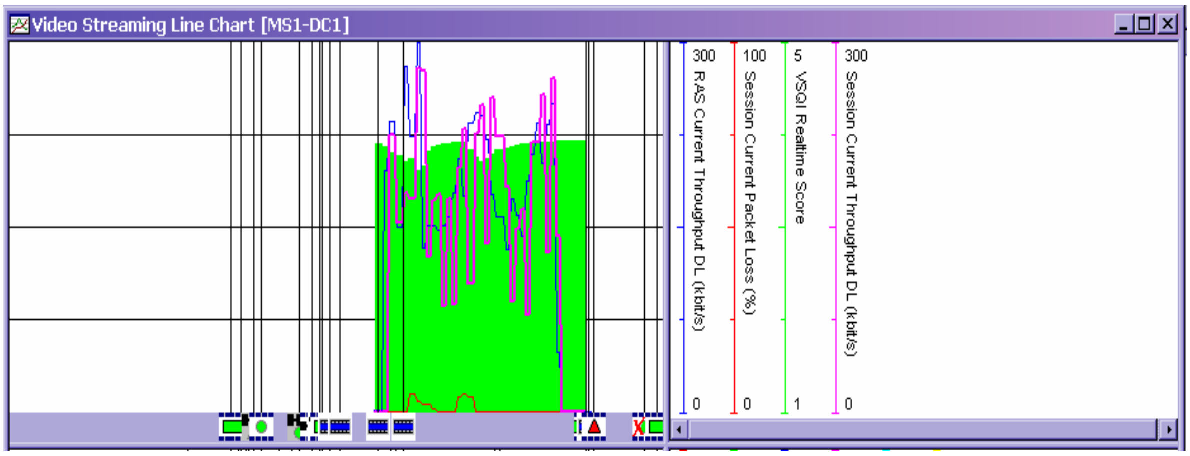

Figure 5-3 : File 11 under test case15. The variation in MOS due to packet loss and jitter.

5.6. Results

In this section all the test cases are summarized in a few graphs .Let us start from case1 and case2 in Table 5-2 which are prebuffer effect on VSQI. We considered two extreme cases of packet loss, in order to study the prebuffer effect in presence of low or high amount of packet loss.

Figure 5-4 shows the decreased VSQI which is $VSQI_{clean} - VSQI_{impaired}$.

So higher values in y axis means worse video quality. Larger prebuffers are needed when there is packet loss. On the other hand the VSQI algorithm, takes into account the duration of prebuffering on calculating the final VSQI. This means that large prebuffer size, which causes more delay, is not desirable by the viewer, and makes the total VSQI decreased. That is why, though prebuffer of 4 second, may compensate for a lot of packet loss, gives a worse score than the small 1 second prebuffer. The test shows a prebuffer of 3 second length gives the best VSQI scores.

However the graph in Figure 5-4 implies that packet loss has totally stronger impact on overall VSQI rather than prebuffer size.

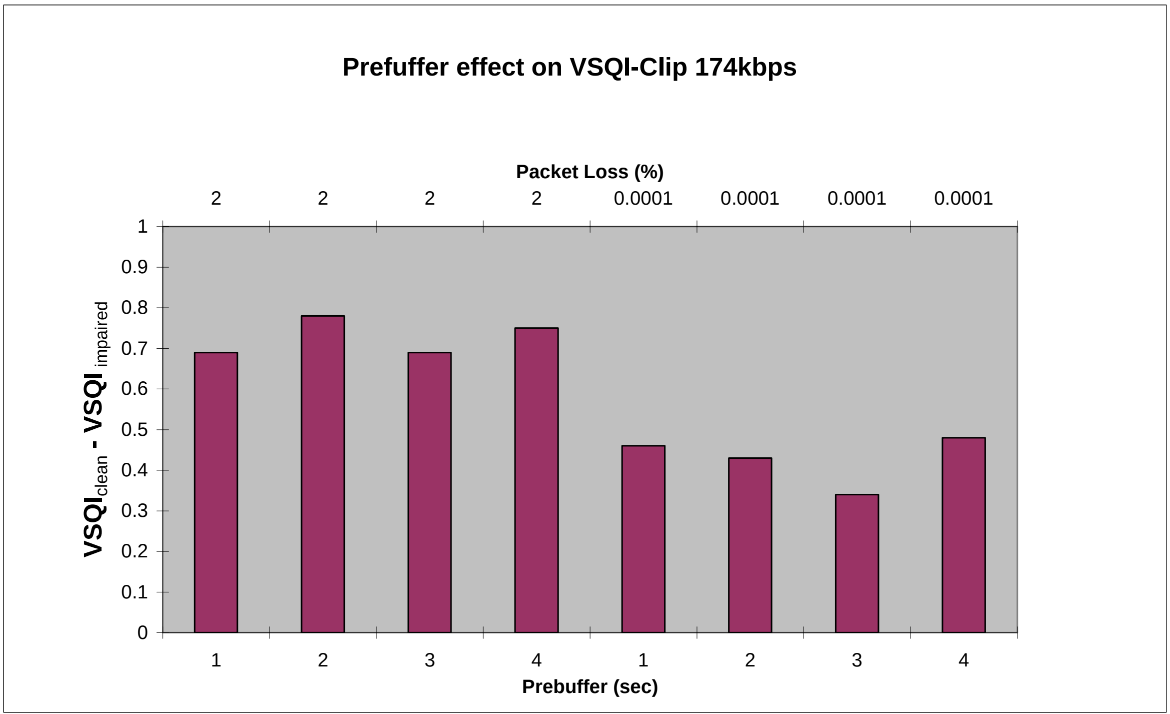

Figure 5-4 : Prebuffer effect on VSQI, with two different packet loss

Case3 to case8, compare effect of various packet loss percentage on a set of varying bitrate clips with MPEG-4 and H.263 codecs.

Figure 5-5 summarizes hundreds of raw data in one graph. One may study the

Figure 5-5 by considering the growth of packet loss effect on a specific clip. On the other hand, assuming constant packet loss, different bit rate clips show different VSQI decline.

We may expect that increase in packet loss, should always makes the video quality worse.

Although any packet loss value, lowers the VSQI value; there is no such linear relation between packet loss increase and VSQI decrease. The reason is that the loss occurs

randomly . Due to compressed video characteristics (See 2.5.2), the lost packet might be a part of I frame , which has bad impact on the consequent IP packets too, or a B frame that does not make the decompression of the next frames impossible. Comparison between extreme values of packet loss, shows that higher packet losses generally cause worse video quality , which is quite obvious and out of question.
A specific packet loss percentage has more impacts on higher bitrate clips, rather than low rate ones, though there are still exceptions.

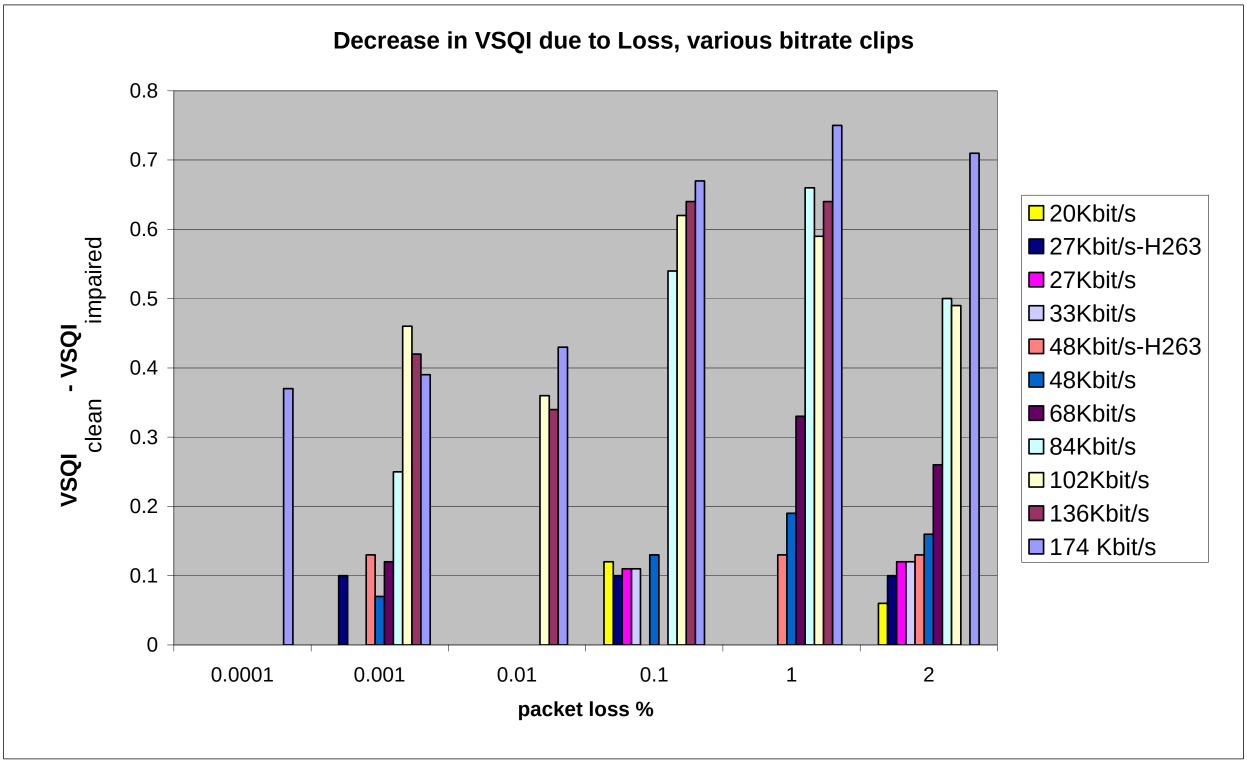

Figure 5-5: Packet loss impact on perceived video quality of different bitrate clips.

This is more visible in Figure 5-6. Almost all the clips are MPEG-4. Except for 84kbps video clip that has got a VSQI difference, even worse than higher bit rate files, the trend is reasonable.

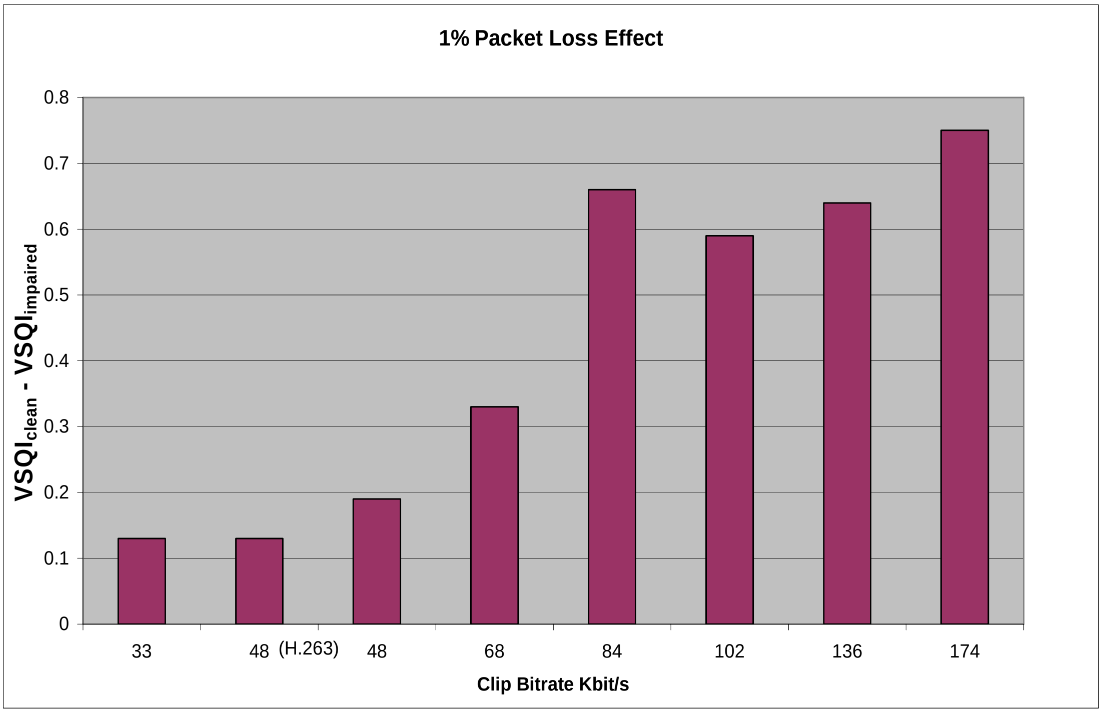

Figure 5-6: Different bitrate clips, under the same condition; 1% packet loss

Figure 5-7 shows the effect packet loss only on one clip with different packet losses. The changes in VSQI are reasonable. 1% packet loss has worse results than 2%.Maybe the number of iterations were not enough to get reliable statistics or the random characteristics of packet loss effect has made this output.

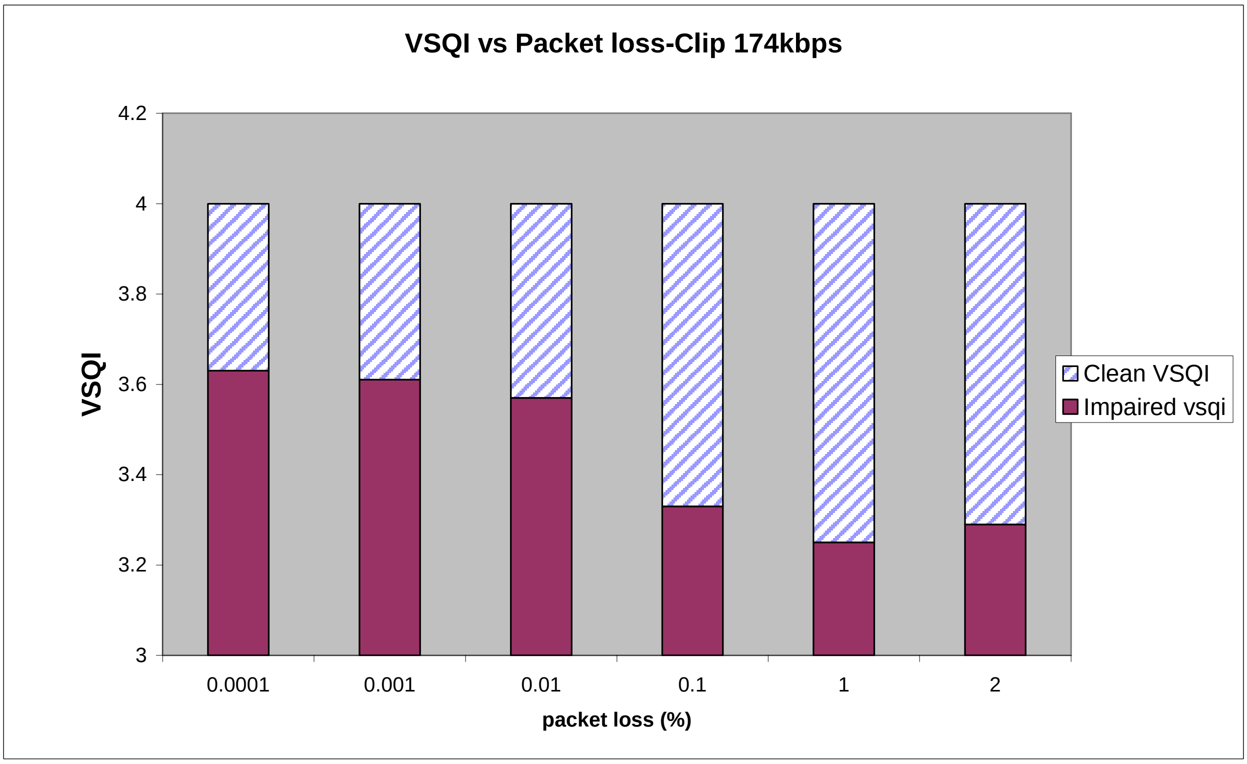

Figure 5-7: An example of Packet loss effect on VSQI.

Now it comes to jitter and the complicated discussions around it. Case 9 and case10 include a lot of jitter tests. Case9 is a test with no jitter ,but 200 millisecond constant delay. This case gives no VSQI impacts . The reason that VSQI in case 9 is not equal to clean VSQI is the prebuffer that makes the perceived VSQI a little lower than the clean VSQI. This can be seen in Figure 5-8 in the first column. It does not have a relation between values of jitter and VSQI. The best results are jitters of 20 and 30 millisecond for a 200 millisecond average delay. I could not find any description for the jitter 5 and 10, that resulted in the worst cases. Maybe some other network conditions at the test time were disturbed, which means the test case was totally wrong.

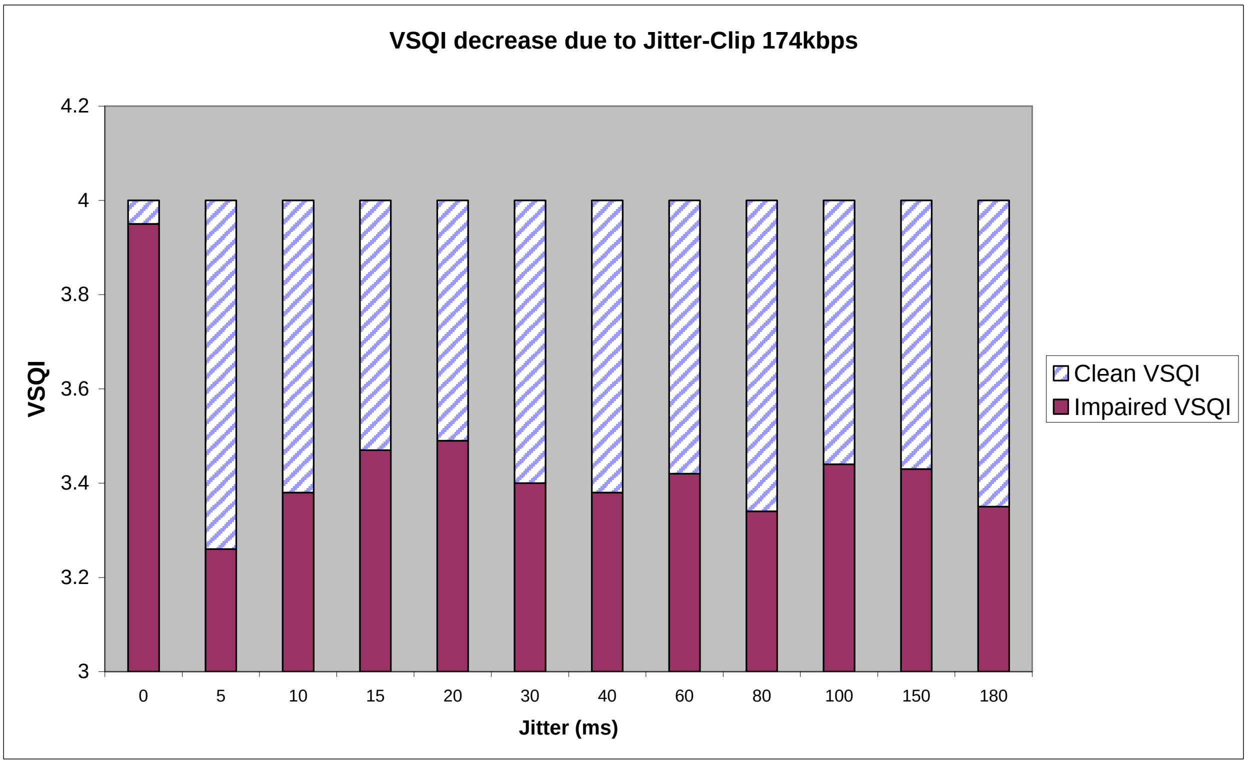

Figure 5-8: Jitter effect on a network with 200msec delay.

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The whole above tests, that are in a reasonable ranges of IP loss or delay, the MOS has not decreased for more than 1 scale. However this reduction is clearly visible on the video playback. Especially VSQI does audio video concurrent assessment, so these degradations also show loss of synchronization between audio and video signals.

Chapter 6

6. Conclusion

Setting up a test case and choice of a proper test tool for video quality assessment, largely depends on the application and the client. Test tools are designed and trained for a specific set of video resolutions and/ or length. The clients' decoder behavior and the error concealment algorithms implemented in different clients are the parameters that affect the perceived video quality. Thus a test tool that simulates a mobile client in the receiver, and its scores are independent of the model of mobile handset used in the test, may give us a more reliable network effect estimation.
On the other hand any kind of network impairment impacts video quality. Tests show that, though increasing packet loss or jitter generally decreases video quality, there is not always such a direct relation between network impairments and video quality degradation.

Among the existing tools, TEMS Investigation 8.1 could almost fulfill the mobile TV test requirements. Agilent TPA solution also seems very attractive, for higher resolution mobile TV, like what may be offered in LET networks.

6.1. Further Work

The study of TVQM provided by Telchemy, which is implemented in some vendor tools is suggested. Especially Spirent solutions with TVQM method that arose by almost end of the thesis work are left to be studied. It is also recommended to set up a test case that two different test tools simultaneously measure the same video quality. This may not be easily implemented, because each test tool requires a specific set of video parameters, like format or resolution, or the streaming/end user set up may differ.
As a more reliable work on network emulation tables (see Table 5-2), one can extract the real network logs and apply the real world distributions.

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