HELSINKI UNIVERSITY OF TECHNOLOGY
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Performance Evaluation of H.264/AVC encoded Video over TETRA Enhanced Data Service (TEDS)

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Masters Thesis submitted in partial fulfillment of the requirements for the Degree of Master of Science in Technology

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Abstract of the Master's Thesis

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Abstract

Public Safety Systems (PSS) are communication networks oriented towards supporting activities of public safety actors (police, medical, fire-fighters, etc.). TErrestrial Trunked RAdio (TETRA) is a Professional Mobile Radio (PMR) standard designed to meet PSS requirements with specialized voice communication features and reliable, secure communication links. TETRA Release 2 introduces TETRA Enhanced Data Service (TEDS), to support emerging data-intensive applications such as online navigation and tele-medicine by providing higher, scalable data rates.

This thesis studies the feasibility of streaming video over a wideband TEDS link using the H.264/AVC codec, a video compression standard that manages to retain high decoded video quality while dramatically reducing streaming bit rate. A bandwidth limiter is used to emulate a link that supports data rates equivalent to those specified in the TEDS standard. Effects of video streaming parameters such as codec rate and play-out buffer size coupled with link-induced delay variation on decoded video quality are investigated. Visual quality is rated using objective quality metrics to quantify results with some measure of reliability.

The overall aim is to identify the technical requirements needed to support an acceptable quality of video transmission over TEDS. To this end, we measure decoded video quality in different channel loss conditions, varying video streaming parameters and at different channel bandwidths, plus enhancements such as data traffic prioritisation as defined in the TEDS specification.

Keywords: TErrestrial Trunked RAdio, TETRA Enhanced Data Service, H.264/AVC, video streaming, objective video quality measurement

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Sunbul Zahid Hussain

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List of Acronyms

3GPP – 3^{rd} Generation Partnership Project

ACK – Acknowledgement packet

AI – Air Interface

APCO – Association for Public-Safety Communications Officials

APSNR – Average Peak Signal to Noise Ratio

AVL – Automatic Vehicle Location

B-Frame – Bi-Directional Frame

BS – Base Station

CBR – Constant Bit Rate

dB – Decibel (unit of measurement of ratios)

DCT – Discrete Cosine Transform

DMO – Direct Mode Operation

DREC – Dynamic RECeiver

DTS – Display Time Stamp

ES – Elementary Stream

ETSI – European Telecommunication Standards Institute

FR – Full-reference video quality measurement model

GOP – Group of Pictures

GSM – Global System for Mobile (Communication)

HD – High Definition

HITL – Hardware-in-the-Loop

HTTP – Hyper-Text Transfer Protocol

HVS – Human Visual System

I-Frame – Intra-frame or Intra-coded frame

IP – Internet Protocol

IPDV – Inter-Packet Delay Variation

ISDN – Integrated Services Digital Network

ISI – Inter-System Interface

JPEG – Joint Photographic Experts Group

LLC – Logical Link Control

MAC – Media Access Control

MEX – Multimedia Exchange

MGEN – Multi-GENerator

MLE – Mobile Link Entity

MOS – Mean Opinion Score

MP4 – Extension for MPEG 4 Video Coding Standard coded video files

MPEG – Moving Picture Experts Group

MS – Mobile Station

MSE – Mean Square Error

MT – Mobile Terminal

MTU – Maximum Transmission Unit

NALU – Network Abstraction Layer Unit

NR – No-Reference video quality measurement model

OSI – Open Systems Interconnection

PABX – Private automated Branch Exchange

PAMR – Public Access Mobile Radio

PCR – Programme Clock reference

PDN – Packet Data Network

PDP – Packet Data Protocol

PDU – Packet Data Unit

PEI/TEI – Periphery/Terminal Equipment Interface

PES – Packetized Elementary Stream

P-Frame – Predicted Frame

PHY – Physical

pixel – picture element

PMR – Professional Mobile Radio

PSNR – Peak Signal to Noise Ratio

PSS – Public Safety Systems

PSTN – Public Switched Telephone Network

PTS – Presentation Time Stamp

PTT – Push-to-talk

QAM – Quadrature Amplitude Modulation

QoS – Quality of Service

RF – Radio Frequency

RMSE – Root Mean Square Error

RR – Reduced-reference video quality measurement model

RTP – Real-time Transport Protocol

SCH – Signalling Channel

SIM – Subscriber Identity Module

SNDCP – Subnetwork Dependent Convergence Protocol

SSIM – Structural SIMilarity

SwMI – Switching and Management Infrastructure

TCP – Transport Control Protocol

TDMA – Time Division Multiple Access

TEDS – TETRA Enhanced Data Services

TETRA – TErrestrial Trunked RAdio

TRPR – TRace Plot Realtime

TS – Transport Stream

UDP – User Datagram Protocol

UMTS – Universal Mobile Telecommunication System

V+D – Voice + Data

VBR – Variable Bit Rate

VCL – Video Coding Layer

VLC – VideoLan Client

WAN – Wide Area Network

WLAN – Wireless Local Area Network

Chapter 1 Introduction

This chapter provides an introductory background of the thesis topic, before formulating the scope of the work. At the end, rest of the thesis content is summarised.

1.1 Background

Public safety encompasses protection of the general public from all kinds of calamities, whether natural or manmade and taking preventive measures wherever possible. Civil bodies like the police force, fire brigade, hospitals' emergency units, maritime or coast guard are public safety agents. They are the first responders to emergency situations. Public Safety Systems (PSS) refer to the telecommunication infrastructure that enables first responders to co-ordinate their relief efforts via mobile communication. An example can be a fire rescue mission, where a centralized command can keep track of the fire brigade team's whereabouts and inform the medical teams of any survivors found and their condition in a timely fashion. Such systems need to be highly adaptable to rapidly changing scenarios, to allow integrated communication between different teams as they arrive on location.

The regular commercial mobile telecom systems such as Global system for Mobile Communications (GSM) or Universal Mobile Telecommunication System (UMTS) based mobile communication systems, are inadequately equipped to deal with public safety response situations, as they do not provide an interface for the centralized command to co-ordinate teams built in ad-hoc fashion. In contrast, Professional Mobile Radio (PMR) systems, which include Private Mobile Radio as well as Public Access Mobile Radio (PAMR), are dedicated wireless communication systems with features designed for PSS such as creation of communicating groups, specifying communication priorities on the go and an interface that connects to a dispatch console to enable co-ordination of activities between the different actors on site.

Furthermore, PMR systems provide voice communication features that are geared toward satisfying PSS communication needs, such as group call where all members of

a team can hear conversations simultaneously, direct mode operation where one handset can act as relay for another outside of base station coverage and the half-duplex push to talk service, with overall emphasis on high reliability and security. These systems use handheld, portable radio devices and provide for portable base stations as well that can be rolled out on-demand.
The TErrestrial Trunked RAdio (TETRA) standard is the outcome of PMR standardization drive initiated by the European Telecommunication Standards Institute (ETSI). The standard's development has continued to date in conjunction with a variety of industry partners under the umbrella of the TETRA Association, to ensure inter-operability across implementations.

TETRA Release 1 is the currently deployed version that supports voice and data (V+D) services. This version adequately fulfils the voice communication requirement as well as supporting narrowband data applications such as Automatic Vehicle Location (AVL) updates for tracking vehicles and monitoring transport systems, messaging services for status updates, email and fax, data access services enabling database retrievals from intranets and remote control via embedded telemetry service [10]. Security-related alerts and natural disasters have demonstrated the usefulness of such data communications in emergency situations [3]. These narrowband data services are now included in most statements of requirements for PSS in addition to the voice services [2][3].

Data communications for public safety purposes can be categorized into interactive and non-interactive [2]. Interactive implies a query-and-response type of communication. The queries may be manually initiated or automated to provide public safety practitioners with information such as location maps, floor plans or even information from real-time monitoring devices (security cameras, heat sensors, alarms etc.). Non-interactive data communication assumes one-way information stream model. Typical examples are remote monitoring of personnel location, biometric information and live video feed from the field. These services increase the overall situation awareness at the command and control centre, allow expedited response planning and improve the practitioners' own safety[1].

Of the above-mentioned non-interactive services, the transmission of real-time full-motion video from remote locations is now considered essential for mobile units of public safety and security (PSS) organisations [4][5]. In a survey on user requirements carried out by Motorola and the Association of Public Safety Communications Officials (APCO), 30% of PSS officials and more than 60% of police officers say they view mobile video systems as useful in their day-to-day as well as emergency situations [9]. Video streaming provides new capabilities to emergency responders. It enables a clear, immediate view of an emergency situation from a distance; much better than the transfer of still images alone. This kind of information helps to improve the timeliness and effectiveness of the emergency response and relief efforts [6]. There are a variety of ways in which this application can be used ranging from live streaming to reviewing recorded sequences of past incidents that can be helpful in investigations [7]. However, while broadband data communication has a great potential to enhance quality of public safety response mechanisms, it also demands higher network resources than the currently deployed TETRA systems can support. So far, TETRA is only able to support low quality, slow-scan video such as from surveillance cameras.

The current TETRA deployment is not capable of providing data rates high enough to support video streaming. TETRA Release 2 introduces TETRA Enhanced Data Service (TEDS), an update of TETRA Release 1 that utilises incremental channel bandwidths, coupled with improved modulation and error correction coding schemes, to provide scalable datarates high enough to support the next generation of broadband data services for PSS. TEDS has not been designed for any particular data application, rather it provides a bit-pipe that can enable new data-intensive services such as bio-data verification, tele-medicine, real-time transfer of images, video and maps, online navigation etc.

This thesis deals specifically with the case of transmitting video over a TETRA network with TEDS enhancements. The TETRA Release 2 standard proposes to provide raw datarate for various data services; however, what is lacking is an estimate of the technical requirements to support an acceptable quality of video transmission, such as: Is video transmission service feasible over the wideband TETRA/TEDS? What kind of visual quality can be expected? What factors will affect performance of video streaming over TEDS? This thesis work shows that video streaming is indeed feasible over TEDS and elaborates some of the video parameters and measures that can be used to gauge video performance. This information will help in optimising usage of the current system to be deployed, or in identifying targets for future system development.

1.2 Scope

The objective of this thesis work is to examine the feasibility of streaming digitally compressed and encoded video over TEDS backhaul link in view of the bandwidth constraints, and measure expectable received video quality. The scope is to investigate the joint impact of parameterization of the transmitted video stream, channel conditions and TEDS specifications on video performance on a TEDS link. Additionally, it covers an examination of co-existence characteristics of the video stream, in the presence of competing traffic flows over the TEDS backhaul link. The objective here is to uncover possible video quality degradation issues arising from this factor of background traffic.

The methodology adopted to conduct this thesis work is detailed in chapter 5. Here it is sufficient to state that a bandwidth limiter is used to emulate a TEDS link. Pre-recorded video clips are used to generate live traffic. The setup used allows control over video streaming parameters to observe effect of change in these on decoded video quality.

1.3 Organization

Chapter two describes the video streaming framework and the process of video compression, followed by a discussion of the constraints and performance trade-offs involved in video streaming over IP via UDP or TCP. Chapter three introduces the TETRA network in terms of specification and services offered, as well as highlighting the data capabilities that TEDS adds to TETRA release 2. The chapter also presents video over TEDS scenario in terms of the TETRA/TEDS protocol suite. The subsequent chapter outlines previous work done relevant to the thesis topic, and also highlights how this study can contribute to further development of TETRA/TEDS systems from an application point of view. In chapter five, the problem solving approach taken in this work is outlined in detail, with a description of the test bed setup as well as metrics and software tools utilised to measure system performance. In chapter six, we verify the emulated link's behaviour before commencing actual testing. Chapter seven details system performance analyses results for various video over TEDS scenarios. The concluding chapter summarises the findings of the work and discusses the implications for video over TEDS as well as highlighting possible directions for future work.

Chapter 2 Video Streaming Framework

Video streaming refers to the transport of stored or pre-recorded video in real-time over a network or link [22]. There are three possible modes for video streaming; one is the ‘download’ mode where the entire file has to be downloaded before being played out. A second mode is ‘progressive download’ where downloaded parts of the file may be played out already from the buffer while the rest of the file is still downloading. The extreme end of the spectrum is ‘streaming’ which allows for minimal buffering of video content before being played out i.e. the file is played out almost as soon as it is received.

Streaming is our chosen scenario as it closely models real-time live streaming of video. We have opted not to use live video feeds from webcams, even though it is possible, for the sake of repeatability in our tests. The problem with using live streams is that it cannot be guaranteed that the video content will not vary from trial to trial. It is important to have the same video content for repeatability as it has an impact on the encoded video characteristics. Especially the amount of motion in a video sequence can affect the size of the frames that make up the video stream [20].

The components of a framework required for video streaming include a codec (for compression and encoding of video to be transmitted at one end, and decoding of the received video at the other), a streaming server and client application, and transport protocols for carrying the video stream, among others [22]. This chapter takes a look at the composition of the video stream to be transported in order to have an idea of the challenges to the transmission of video over a packet-based data network. We also discuss the reasons behind the choice of H.264/AVC codec.

2.1 Video compression (encoding/decoding)

Video is a continuous procession of pictures (also known as ‘frames’) that give the illusion of continuity and movement i.e. picture in motion. Video in its raw, uncompressed format, needs to be compressed in order to be transported over IP networks. This is necessary as raw video is quite heavy, and requires a large amount of storage space and equivalently, transmission bandwidth. This is the job of the video codec, a compression/decompression algorithm that aims to limit a video stream to a target bitrate, referred to herein as ‘codec rate’. The basic principle of any compression algorithm is to remove redundancy in information. Therefore, the higher the codec rate, the more redundant information is retained during the encoding process and the higher is the expected quality of the decoded video. Two kinds of redundancies are encountered in video. Spatial redundancy refers to the picture elements (pixels) within a frame that contain the same information. Temporal redundancy refers to the pixels that contain the same information between consecutive frames. Thus, video codecs are designed to perform compression both in space and in time.

The video encoding process begins with the raw video in RGB format being converted to YUV format (that is, the red, green and blue picture frames captured by a video camera are converted to digital signals composed of the luminance and chrominance signals). The RGB format contains highly correlated data. It is convention to convert raw RGB video to the YUV format as it reduces the correlation and allows for greater compression without loss of data. The chrominance signals U and V are also known as ‘colour difference’ signals (B-Y and R-Y) that can be computed from the ‘luminance’ or brightness signal, Y. Thus, YUV format is also referred to as raw video.

Each frame is divided into blocks of pixels. ‘Pixel’ is short for picture element which is the smallest discrete unit of a frame, better understood as the small dots of colour that make up the display on a screen. It is a sample of the original picture signal that contains colour intensity information. Picture resolution is defined in terms of number (or matrix) of pixels used to build up a frame; the higher the number of pixels, the greater the smoothness of the display. There is a high correlation between neighbouring pixels in a frame. Transform coding is used to exploit this inter-pixel redundancy to achieve compression. A transformation is used to map the correlated data from the spatial domain to the transform domain. A frame is divided into blocks of pixels, as taking transform of a large frame can be very complex. A video codec processes the frame block-by-block to compute discrete cosine transform (DCT) of each, representing it as a matrix of coefficients. Other transforms can also be applied such as Fourier or discrete sine transform. However, from compression point of view, the cosine transform proves much more efficient as it requires a less number of coefficients to approximate a typical signal [15].

The resulting transform coefficients can be coarsely quantized without noticeably affecting image quality resulting from inverse DCT of the quantized coefficients. The quantization scale can be set so that many of the high frequency coefficients go to zero value, taking advantage of the fact that these components are not easily noticed by the human eye. The quantized values are fed to an entropy coder (such as Huffman encoder) in a zig-zag scan to increase the run of zeros, which results in a more efficient coding using run-length encoding. The resultant compressed frame is called an Intra-frame or 'I' frame as it has been 'intra-coded', in a process designed to remove spatial redundancy. This is the main frame of reference in a video stream, which contains the most information and tends to be the largest in size.

Other frame types are created by a process intended to remove temporal redundancy between frames. The frames following an I-frame can be encoded based on the difference between itself and the I-frame, that is, if there is no difference in position of a certain block of pixels an indication to the effect is added rather than re-coding. If the same block of pixels has moved, a set of motion vectors are indicated. This process is called 'predicting frames'. Frames predicted from past I or P frames are called 'Predicted' or P-frames. A third type, 'Bi-directional' or B frame is predicted from both preceding and following I or P frames. Hence, the most compression and consequent reduction in bit rates of the video stream occurs in the P and B frames. The frame encoding process and types are depicted in Figure 2-1

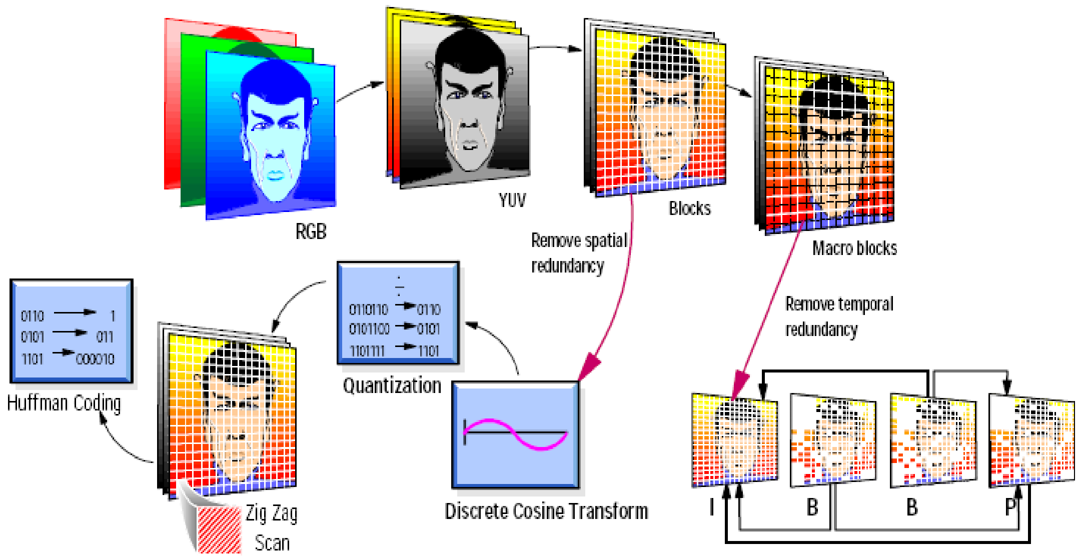

Figure 2-1 Compressing and Encoding video: The I, P & B frame concept [21]

In practice, the I, P and B are generated by codecs in a fixed sequence known as Group of Pictures or 'GOP' format; for example, repeating sequences of IBBP or IPIPIP (no B frames), etc. It follows from the description of video compression and encoding process above, that the quality of the video stream is highly dependent on the preservation of I-frames integrity during transmission, as most other frames are predicted using it as reference. Since the B frames depend also on future frames for reconstruction, the encoded video frames are not transmitted in the order they were created; rather transmitted in the order that allows for B frames to be decoded as shown in.

This indicates the practical importance of timing in video playback while streaming video; the transmitted frames must be decoded and re-ordered in the correct sequence in time for play-out at receiving end. If a frame is not available in time for play-out, it is considered lost even if it arrived at the receiver, albeit late. This creates a disturbance in playback in the form of stuttering video, where the picture becomes grainy or 'pixelated' during transition from one frame to another due to the missing information. Such continued stuttering leads to perceived degradation of video quality. Hence, video streaming has stringent requirements of low latency during transmission.

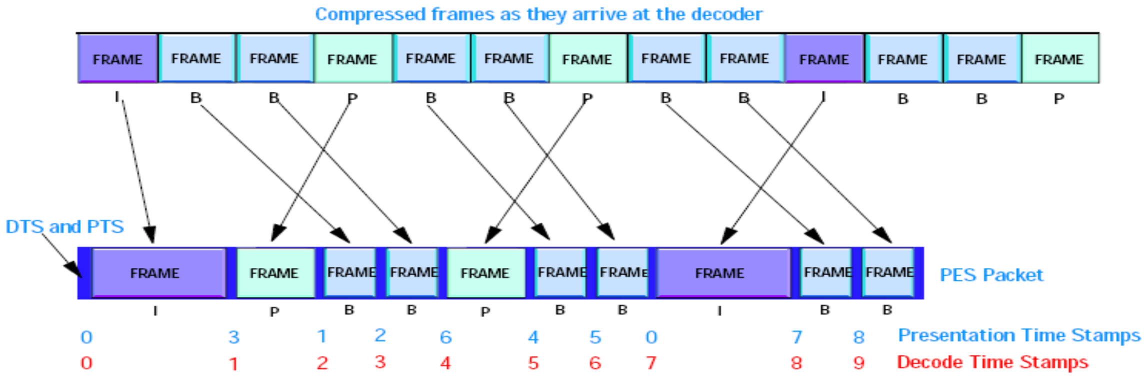

Figure 2-2 Interleaving frames for transmission [21]

Video needs to be packetized for transmission over packet data networks. The packetized video is used to build transport streams (TS) to be transmitted. To this end, each sequence of frames is grouped under Sequence, GOP, Frame and Slice (for macro-blocks i.e. a block of pixel blocks, within a frame) headers preceding each sub-grouping respectively. The headers contain all the information needed by a decoder to reconstruct the encoded video. These continuous encoded video frames comprise an Elementary Stream (ES). However, no timing information is included. To allow for re-ordering and decoding of frames in the correct sequence for playout, the timing information is embedded in Packetized Elementary stream (PES) as shown in preceding figure [21] as Presentation Time Stamp (PTS) and Decode Time Stamp (DTS) that tell the decoder when to display and decode a received frame respectively. However, a further clock reference is required to synchronize playback especially in the case where both audio and video files are transported together. For this purpose a Programme Reference Clock (PCR) is provided in the Transport Stream (TS) which chops PES into fixed sizes of 188 bytes and multiplexes different PES from different sources (audio, video or even entirely different video streams). This constitutes the structure of an MPEG-TS encapsulation as shown in Figure 2-3.

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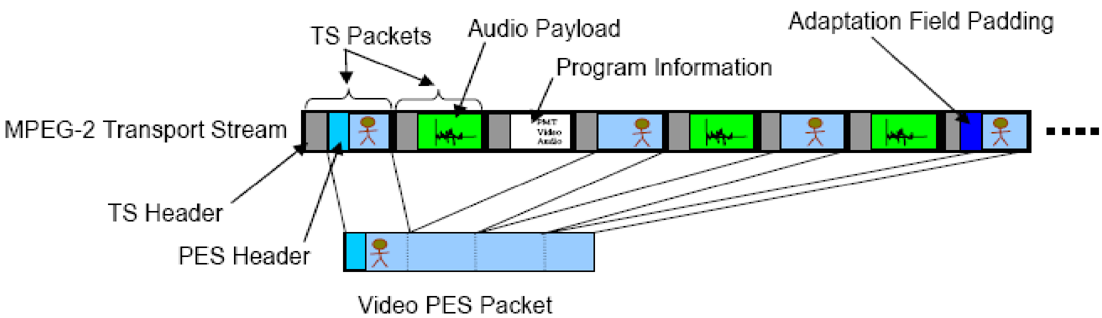


Figure 2-3 MPEG-Transport Stream: Fitting video into packetized stream [16]

2.2 Selection of video codec

Over the years, many video coding standards, both propriety and open source, have been designed by various industrial and scientific organisations. In this thesis work, we use the H.264/AVC (advanced video codec) codec, an open standard jointly developed by the Video Coding Experts Group (VCEG) of ITU-T and the Moving Picture Experts Group (MPEG) [32]. This video codec is the latest in a succession of MPEG defined codec standards. A further improvement, the H.264/SVC (scalable video codec) has also become available recently; however, it is not yet widely implemented in media players. The SVC enables on-the-fly bitstream adaptation to suit the available transmission conditions.

A further justification for choosing the H.264 codec is that it is gaining popularity in the industry for video streaming, as it manages to achieve the highest video quality while employing the lowest bitrates (nearly half the rate achievable by the previously prevalent MPEG-2 standard) [34]. Notable work has been done to study video streaming using H.264 particularly in the mobile communication environment. For instance, 3GPP (3rd Generation Partnership Project), an industry consortium responsible for developing 3G mobile communication standards, has carried out a software-based simulation testing regarding the appropriate video codec to be used for mobile video streaming applications over UMTS [28]. They used the H.263 and H.264 codecs and collected performance measures for the encoder/decoder ensemble's ability to withstand packet drops achieved by simulating channel conditions for expected best-case and worst-case scenarios. Another study uses subjective metrics for H.264 encoded video quality measurement to demonstrate its effectiveness for use specifically with mobile handset provided screen sizes and resolutions [29].

The H.264 offers flexible options for encoding video, and defines three distinct coding profiles; the Baseline, Main and Extended profiles. The Baseline profile settings emphasize minimizing complexity and high robustness, the Main focuses on coding efficiency, while the Extended profile combines the best of both [34][34]. We use the Baseline profile for our tests; however, it is conceivable that use of the

Extended profile, and eventually the SVC standard, can further improve video streaming performance.

The H.264 codec achieves its improved coding efficiency and robustness largely by fine tuning the level at which compression techniques can be applied. Firstly, instead of per-frame encoding, it employs per slice (macro-block) encoding which means that there are I, P and B slices instead of frames. Secondly, it allows the motion estimation in the P and B slices at macro-block or even sub-macro-block level. Then, it defines Switching I and P frames (SI and SP) that allow for switching between streams coded at different bitrates. These allow reconstruction of samples from different sets of references in the prediction process. This multiple reference picture support represents an improvement in the error concealment techniques employed by the codec. In addition to this, the slices within a frame are coded so as to allow independent decoding. Thus, it is not any particular technique but the sum of small improvements in the cascaded coding process that enable a significant gain in coding efficiency as compared to prior codecs [32].

2.3 Video over IP

The Internet Protocol is designed to run over a best-effort packet switched network. Since this thesis considers video streaming over IP networks, it means that the digitized video must be packed into packets that can be carried over IP. The H.264/AVC codec defines two conceptual layers. One is the video coding layer (VCL) that handles the digital signal processing part of the encoding mechanism and outputs slices containing macro-block data that make up the frames. The other is the network abstraction layer (NAL) that encapsulates these slices into NAL units (NALU) that are suitable for transport over packet-based networks. RFC 3984 defines a mechanism for encapsulating this NALU information in a new H.264 specific RTP or real-time protocol payload format [17].

However, the VLC (VideoLan Client) software which we use to generate and transmit H.264 encoded video over our test network only supports streaming H.264 encoded video over RTP/UDP by first encapsulating it in MPEG-TS format and then encapsulating the MPEG-TS in RTP [19]. The latter can be done as discussed in RFC 2250 [18]. This design choice may be due to a need to maintain compatibility between end systems that still use the widely prevalent MPEG-TS programme format.

The structure of video packet carried using Real-time Transport Protocol (RTP) over UDP over IP is shown in Figure 2-4[16]. The number of TS packets encapsulated in an IP packet is dependent on the maximum transmit unit (MTU) of the networks to be traversed. For Ethernet, MTU is 1500 bytes, so 1500/188 equals approx. 7 TS packets.

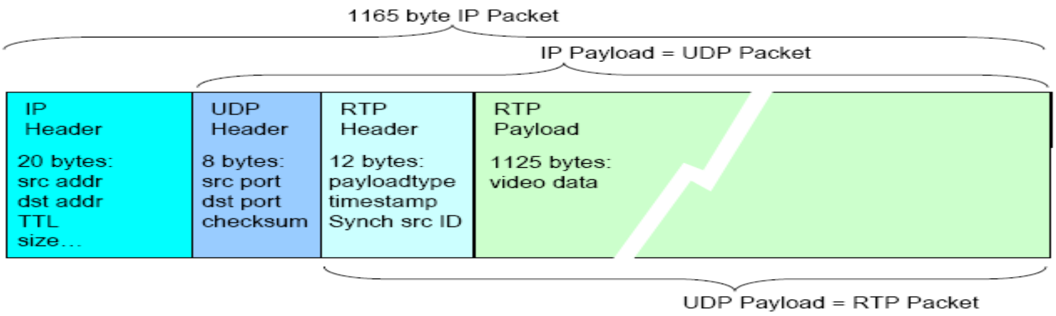


Figure 2-4 IP packet structure [16]

Referring to the OSI Network Model (Figure 2-5), the MPEG-TS encapsulation represents the Application set of protocols that can act as vehicle for video transport at application layer. This MPEG-TS encapsulated information is sent down to the lower transport layer by the application layer to be further encapsulated by the transport layer protocol header (either TCP or UDP) and so on until the information passes through the network to the receiving end. Here, the entire process is repeated in inverse order, ending with MPEG-TS de-multiplexing by the decoder at the application level of the receiving end.

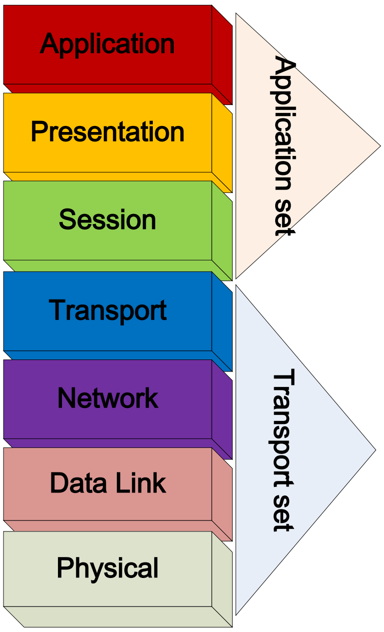


Figure 2-5 OSI Model

The transmission of video in streaming mode entails severe delay constraints that must be met in order for video reception to be workable. In an IP-based best-effort network, packets may be re-ordered or even lost during transmission. If all packets do not arrive in time for the upper presentation and application layers to decode and display the video information according to the pre-defined pattern, the reproduced video quality will suffer [22]. Thus, the choice of method for video streaming must be made keeping in view the inherent constraints of video reception at application level, in conjunction with the limitations of transport protocol employed. Other factors such as scenarios (download for storage or later viewing vs. immediate viewing) and transmission environment (wired vs. wireless) must also be considered.

There have been many studies investigating the possibility for video streaming over the Internet, an IP-based, packet-switched, best-effort network. The issues identified for video streaming in this scenario are closely related to the stringent video streaming requirements imposed by encoding and decoding ends [22]. One is a minimum bandwidth requirement for video streaming. Second is a delay constraint, i.e. the video packets must arrive in time for the decoder to decode and display video frames in sequence. If this doesn't happen, it results in noticeable quality degradation in the form of frozen video. In the case of streaming, where minimal buffering is employed at decoding end, jitter, or in other words, the variability in packet arrival times can also affect visual quality negatively in the form of pictorial distortion due to residual coding artefacts persisting on display screen [25]. This is especially the case when video decoders use error concealment techniques that keep on displaying the same frame or macro-blocks of pixels within the frame if the next frame or some parts of it are not available by the time to display. Without these techniques, there may be an even more severe problem than mere freezing, which is that the video playback may take much more time to recover from a large number of packets that were dropped by the decoder due to late arrival. Another reason for packet dropping is that packets may be lost or corrupted on the link. For whichever reason, packet drops result in a number of missing packets in the video stream, with the consequence that the decoder has to wait until the next I-frame to resynchronize playback. This is visually noticeable as jerky playback with a few seconds' chunks missing.

The issues enumerated above play a pivotal role in choice of transport protocol and the resulting streaming performance. TCP and UDP essentially perform the same functions; that is, they take packets from the upper application and presentation layers, multiplex different streams using specific transport layer identifiers (port numbers), add checksums to enable end-to-end error checking in the encapsulating headers and pass them on to the lower networking layer for transmission over the network. The difference is in the behaviour of these protocols regarding packet losses. UDP is a simple, lightweight protocol that simply discards any corrupted packets received. TCP on the other hand, is a more robust protocol that allows for re-transmission of lost information by implementing a system of acknowledged transfers where each packet or group of packets is acknowledge. Through such control mechanisms TCP is also able to perform end-to-end throughput and congestion control by limiting source output based on receiver feedback (frequency of reception of ACKs or lack thereof) using the windowing concept of sending and receiving. This results in a difference in overall behavioural characteristics of streams transported by UDP and TCP.

UDP-based flows maintain their bit rate given that link bandwidth is enough to accommodate the flow. TCP based flows are congestion controlled, and hence subject to fluctuating bit rates depending on the number and type of competing flows and leftover bandwidth. While TCP flows scale back their employed bit rates in response to network or link congestion, UDP based flows tend to try and maintain their original bit rates. This results in potential irrecoverable packet loss for these streams in congestion periods due to packet corruption. On the other hand, TCP flows may slow down but the packet loss will be almost non-existent, due to the retransmission mechanism. This means that UDP flows experience fixed delay and exhibit constant bandwidth occupation with high loss probability; whereas TCP flows experience varying delays (also known as jitter) and exhibit variable bandwidth occupation with very low loss probability. It is important to note that TCP flows do not maintain any particular data rate, which means that they try to take up all available bandwidth if no other flows are present, through constant probing of network resources via incremental increase of sending window thresholds. This TCP flow characteristic is referred to as ‘aggressive’ behaviour.

The choice between TCP and UDP poses an interesting problem for carrying video over IP. Normally, the use of TCP is widespread in the Internet, due to its inherently reliable information transfer capability. However, for video streaming UDP is preferred [22]. This is because while packet loss does adversely affect video quality, it can still withstand some amount of loss. In contrast, delay or jitter is intolerable to video streaming as it interferes with play out timing and affects visual quality more noticeably. Hence, this choice involves a trade-off between packet loss and delay. However, TCP can still be used for progressive download scenarios, also known as ‘HTTP-streaming’. This guarantees a good visual quality for video at the expense of start-up delay due to buffering requirements before play out [26].

Chapter 3 TETRA/TEDS Overview

This chapter summarises information regarding the TETRA standard for PMR Systems in terms of general architecture including interfaces and services. It goes on to present a summary of the concepts of video streaming over IP and TEDS.

3.1 TErrestrial Trunked RAdio

TETRA is billed as the first and only digital trunking standard for PMR networks. TETRA represents a highly robust platform for the provision of integrated voice and data services. The input of users, especially from first responders to emergency situations during the standards development, has led to its high recommendation for use in PSS. As is the case with any telecommunication standard, the TETRA standard defines multivendor interfaces in terms of functionality, leaving the implementation details to vendors but ensuring interoperability at the same time.

3.1.1 TETRA interfaces

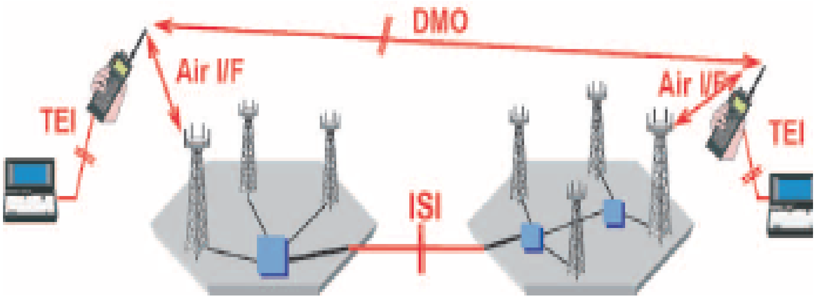


Figure 3-1 TETRA defined interfaces [11]

A brief description of the interfaces, as shown in Figure 3-1, defined by the TETRA standard is as follows:

Air Interface (AI): Defines the interface between mobile radio station (MS) and base station (BS) and allows for operational compatibility of multivendor handsets in the same network. The radio access method is based on time division multiple access (TDMA), with four user channels multiplexed on one carrier of 25 kHz bandwidth.

Inter-System Interface (ISI): The ISI allows for inter-connection of TETRA networks by different manufacturers.

Direct Mode Operation (DMO): is the interface that enables direct radio-to-radio communication outside of the BS coverage by implementing a walkie-talkie style, Push-to-talk (PTT) mechanism.

Periphery or Terminal Equipment Interface (PEI/TEI): This interface allows handheld radios to be interfaced to other devices such as laptops and facilitates development of mobile data applications.

3.1.2 TETRA features and services

The TETRA interfaces do not just define radio access methods but also enable definition of services and functionality as follows:

3.1.2.1 Central management

TETRA systems have the capability to link to a despatch console to the switch via Switching Management Interface (SwMI) in a central command and control model typical of rescue effort communications. The console allows management of individual subscriber rights, definition of communicating groups on the go, as well as communication monitoring.

3.1.2.2 Relay communication

The DMO interface allows TETRA to simultaneously support out-of-band direct as well as group mode communication. Through this interface, the handheld radio can act as relay for another radio that may be outside BS coverage. It also enables two TEs to communicate without BS on a radio frequency different from the BS carrier frequency.

3.1.2.3 Secure networking

The TETRA digital trunking standard has been designed to support effective resource sharing by allowing different public safety agencies such as the police and fire brigade to use the same TETRA network while maintaining privacy and mutual security through virtual networking. The concept of resource sharing is depicted in Figure 3-2

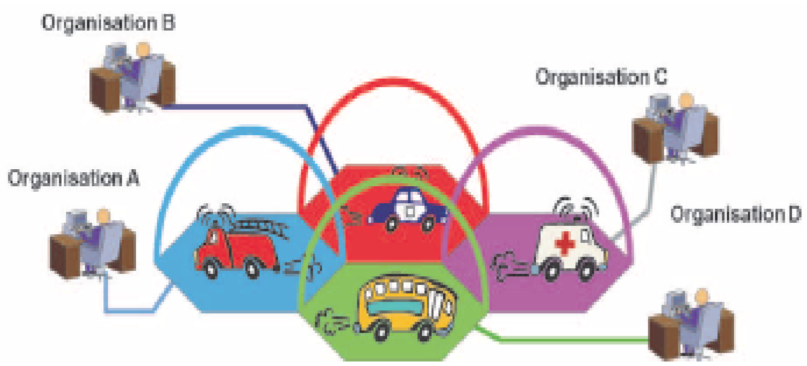


Figure 3-2 Virtual Networking in TETRA [11]

3.1.2.4 Communication features

The unique features of TETRA include fast call setup time and superior voice quality with efficient bandwidth utilisation using vocoders (voice encoders) for speech compression that also support background noise cancellation. Other features include security of communications via over-the-air or end-to-end encryption and pre-emptive prioritising capability. Pre-emption implies assigning certain subscribers or types of calls priority over others, so the resources can be freed immediately from lesser priority ongoing calls if needed, to patch the call through.

3.1.2.5 TETRA services

The services supported by TETRA include bearer services for the transport of data in both packet and circuit switched domains using multiple slots over the air interface to support the higher data rates upto 28.8 kbps without error protection (ETSI EN 300 392-2 version 2.6.8 or earlier). TETRA tele-services comprise voice calls with three possible configurations; one-to-one, group call or broadcast. TETRA also supports about 30 supplementary services including the regular telephony supplementary services as well as special priority mechanisms, fleet services and others [12]. The support of such services is required to allow seamless interworking with external networks (PSTN, ISDN, PABX, PDN, WAN, etc).

3.1.2.6 Supported data applications

The current TETRA deployments support applications that are characterised by short and frequent data transmissions such as AVLs from portable equipment or vehicles, database access from the field and status messaging. However, there is an increasing need for applications characterised by transfer of larger payloads, such as mug shots and other identification information, for example biometric info, as well as video. So far, TETRA can support the transfer of slow scan video using multi-slot packet data [13].

Streaming higher quality video, in real time or near-real time, requires much more capacity than TETRA can deliver at the current operating carrier bandwidths. Applications of the video streaming capability include remote surveillance of sites with high-risk equipment for eg. nuclear reactor stations or oil & gas on and off-shore sites, and discreet mobile surveillance by police force. As compared to slow-scan video, streaming video affords better quality and more details. Table 3-1 shows data capabilities of incremental TETRA releases.

Table 3-1 TETRA Supported Applications [6]

	TETRA 1 Circuit Data	TETRA 1 Short Data Services	TETRA 1 Single Slot Packet Data	TETRA 1 Multi-slot Packet Data	TETRA 2 TEDS
Database look up		★	★	★	★
AVL		★	★	★	★
Email			★	★	★
File transfer e.g. Still image			★	★	★
Slow scan video				★	★
QoS managed video					★

3.2 TETRA Enhanced Data Service

In view of the foreseen increase in demand for data-intensive applications, TETRA Release 2 (ETSI EN 300 392-2 version 3.2.1 or later) introduced TETRA Enhanced Data Service (TEDS), an enhancement to support high bandwidth data capability for the next generation of TETRA networks. The data rate improvement in TEDS is made possible by the following enhancements in the TETRA physical layer (PHY) and lower Media Access Control (MAC) layer 0:

- Spectrum-efficient higher-order multilevel modulation schemes
- Robust turbo coding for payload channel

- Multi-carriers (8 subcarriers per 25 kHz) for robust performance even in frequency-selective fading channels
- Wider carrier bandwidths 25 kHz scalable to 50 kHz, 100 kHz, 150 kHz
- Adaptive selection of modulation level, coding rate and bandwidth according to the varying channel propagation conditions

3.2.1 TEDS supported data rates

TEDS is scalable in the sense that it can be deployed in 25, 50, 100, 150 kHz bandwidths. Higher bandwidths correspond to higher available data rates as shown in the Figure 3-3. Also, TEDS utilizes various coding and modulation schemes optimised to support various needs. The coding schemes employed change dynamically during operation according to RF environment to optimise throughput performance. The communication coverage area decreases as usable bit rate increases. This is because the higher coding rates required for achieving high bit rate links, also make them more error-prone.

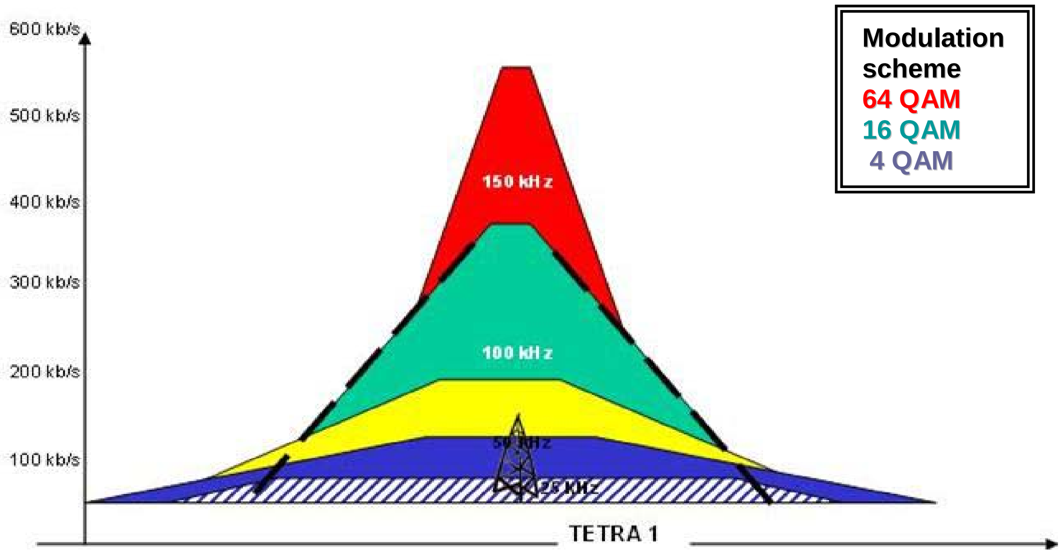


Figure 3-3 TETRA 2 bandwidth compared to TETRA 1 [13]

Table 3-2 Gross Datarates supported by TEDS with different modulation schemes[14]

Modulation type and code rate	Carrier bandwidth (kHz)			
	25	50	100	150
$\pi/4$-DQPSK, r=2/3	15	-	-	-
$\pi/8$-D8PSK, r=2/3	24	-	-	-
4-QAM, r=1/2	10	24	49	77
16-QAM, r=1/2	19	47	98	153
64-QAM, r=1/2	29	71	146	230
64-QAM, r=2/3	39	94	195	306
64-QAM, r=1	58	141	293	459

TETRA Release 2 is backward compatible, that is, TETRA 1 calls are still supported. TEDS is just an extension to complement the existing data support feature set. The TEDS spectrum can be re-allocated to voice calls if these are deemed more necessary than data capacity. The incremental deployment capability enables TEDS to affect a trade off between achieved data rate, spectrum and range. The gross data rates achievable by various TEDS deployment configurations are depicted in Table 3-2 [13].

3.2.2 TEDS uplink radio performance

In the case of QAM (Quadrature Amplitude Modulation), the TEDS MAC layer supports five Signalling CHannels (SCHs). In our case study, we consider only the logical channels SCH-Q/U and SCH-Q/ HU that are used by a MT to send full-slot and halfslot messages to the base station (BS). Note that the logical channel notation uses Q to indicate QAM modulation, U for full-slot uplink messages and HU denotes a half-slot uplink message 0. We focus on the uplink logical channels to estimate channel loss as we intend to investigate video streaming from MS to the network via TETRA/TEDS as backhaul. The minimum required reference sensitivity performance for a non-stationary TEDS channel is specified according to channel attributes such as the logical channel type, propagation condition, coding rate, modulation, channel bandwidth, operating frequency and transmitting equipment (MT or BS) as shown in Table 3-3 0.

Table 3-3 Maximum permissible MS and BS receiver MER at dynamic reference sensitivity level for frequencies below 700 MHz 0

Type of channel	Payload modulation	BS/MS	Code rate	25 kHz		50 kHz		100 kHz		150 kHz	
				TU50	HT200	TU50	HT200	TU50	HT200	TU50	HT200
SCH-Q/RA	4-QAM	BS	1/2	11,1 %	7,4 %	-	-	-	-	-	-
SICH-Q/U in CB	4-QAM	BS	1/2	5,5 %	1,8 %	3,6 %	1,6 %	3,8 %	1,2 %	5,3 %	2,0 %
SCH-Q/HU	4-QAM	BS	1/2	11 %	7,7 %	9,3 %	5,6 %	9,0 %	3,3 %	12,9 %	7,6 %
SICH-Q/U in NUB	4-QAM	BS	1/2	3,6 %	1,4 %	3,5 %	1,3 %	3,6 %	1,1 %	3,9 %	1,6 %
SCH-Q/U	4-QAM	BS	1/2	8,3 %	3,7 %	9,4 %	2,0 %	9,0 %	1,5 %	8,1 %	3,2 %
SICH-Q/D	4-QAM	MS	1/2	1,9 %	0,8 %	2,1 %	0,9 %	2,1 %	0,9 %	2,3 %	0,9 %
AACH-Q	4-QAM	MS	1/2	5,8 %	2,5 %	6,2 %	2,7 %	6,2 %	2,8 %	6,8 %	2,8 %
BNCH-Q, SCH-Q/D	4-QAM	MS	1/2	7,8 %	2,3 %	10 %	1,8 %	8,7 %	1,8 %	8,4 %	1,8 %
SCH-Q/HU	16-QAM	BS	1/2	11,9 %	8,2 %	7,9 %	3,6 %	9,9 %	3,5 %	13,2 %	7,5 %
SCH-Q/U	16-QAM	BS	1/2	8,8 %	3,9 %	7,0 %	1,1 %	9,5 %	1,6 %	8,9 %	3,5 %
BNCH-Q, SCH-Q/D	16-QAM	MS	1/2	8,6 %	2,9 %	7,2 %	1,0 %	9,0 %	1,9 %	8,7 %	1,8 %
SCH-Q/HU	64-QAM	BS	1/2	11 %	7,0 %	8,9 %	4,8 %	8,7 %	3,0 %	12,1 %	6,2 %
SCH-Q/U	64-QAM	BS	1/2	7,8 %	3,7 %	9,9 %	3,0 %	7,7 %	1,6 %	6,9 %	2,7 %
BNCH-Q, SCH-Q/D	64-QAM	MS	1/2	7,4 %	2,6 %	9,3 %	1,9 %	7,3 %	1,8 %	7,4 %	1,6 %
SCH-Q/HU	64-QAM	BS	2/3	11,2 %	11,2 %	7,8 %	7,6 %	9,9 %	7,1 %	14,1 %	11,8 %
SCH-Q/U	64-QAM	BS	2/3	9,5 %	7,7 %	8,3 %	4,4 %	9,6 %	5,1 %	9,3 %	8,1 %
BNCH-Q, SCH-Q/D	64-QAM	MS	2/3	9,3 %	6,2 %	8,1 %	3,0 %	7,3 %	3,6 %	6,9 %	3,9 %

The reference sensitivity bounds enable the evaluation of the maximum permissible receiver Message Erasure Rate (MER) for any combination of the aforementioned attributes. The MER refers to limit ratio of the messages detected as wrong by the receiver to all messages received in a given logical channel. Assuming SCHQ/U and SCH-Q/HU logical channels, QAM modulation and error protection, we note that the mean MER is approximately 10% for Typical Urban 50 km/h (TU50) propagation case and mean MER is 5% for Hilly Terrain 200 km/h (HT200) propagation case, after averaging the standardised MER limits.

The envisioned TETRA 2 deployment is depicted in Figure 3-4, where higher bandwidths are allocated to dense areas. The challenges to TETRA 2 are designing multimedia applications that can take advantage of the higher bandwidth afforded while at the same time catering to the adaptability of data rates according to radio environment as well as determining criteria for cell handover based on either signal strength or capacity or both. Video streaming over TETRA/TEDS is one of the envisioned multimedia applications with potential for use in PSS.

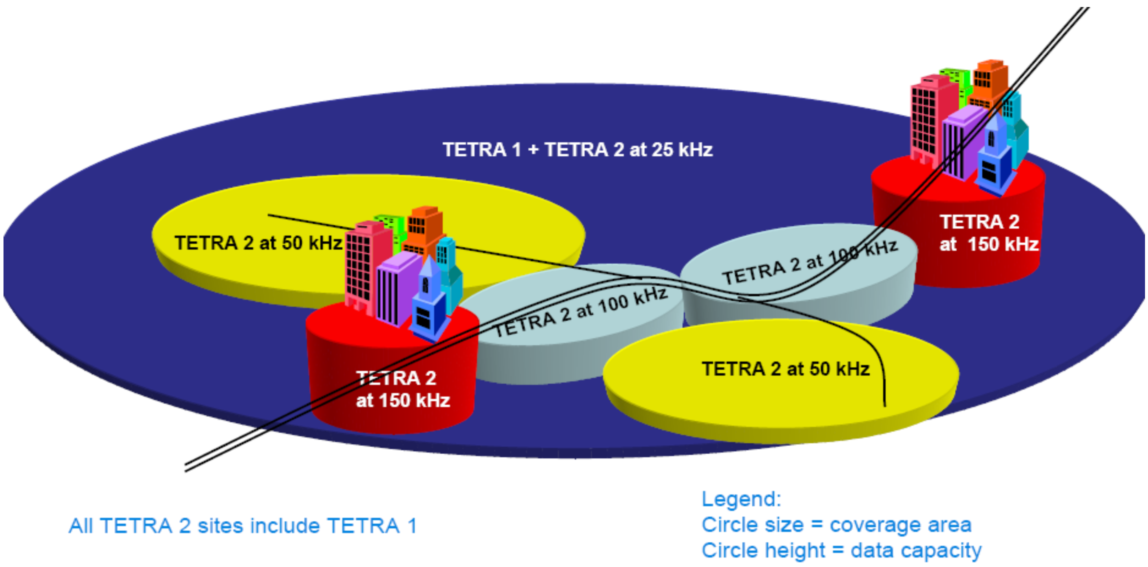


Figure 3-4 TEDS deployment plan [13]

3.3 Video over TEDS

The TETRA/TEDS standard already provides the option to use re-transmisson technique to ensure integrity of packet data streams at link layer level, so the TCP inherent retransmission mechanism appears as overhead for already constrained link bandwidth. The focus of this work is on the performance of video using the UDP transport protocol (RTP or real-time transport protocol over UDP) for video over TEDS scenario.

3.3.1 TETRA QoS classes

The TETRA standard defines traffic priorities for different traffic types; classified according to their Quality of Service (QoS) requirements (which is application dependent) as shown in Table 3-4 [14].

Table 3-4 TETRA QoS Classes

	Real-Time Class	Telemetry Class	Background Class
Delay class	o Low delay class <1s (packet size ≤ 128 bytes) <3s (128 bytes ≤ packet ≤ 1024 bytes) <5s (1024 bytes ≤ packet ≤ 2002 bytes)	o Moderate delay class <5s (packet size ≤ 128 bytes) <15s (128 bytes ≤ packet ≤ 1024 bytes) <75s (1024 bytes ≤ packet ≤ 2002 bytes)	o High delay class <5s (packet size ≤ 128 bytes) <30s (128 bytes ≤ packet ≤ 1024 bytes) <110s (1024 bytes ≤ packet ≤ 2002 bytes)
Reliability class	o Low reliability class Packet loss probability undefined in EN 300 392-2 Duplicate packet probability = 0 Out of sequence packet probability = 0 Corrupt packet probability $<10^{-4}$	o Moderate reliability class Packet loss probability $<10^{-4}$ Duplicate packet probability $< 10^{-9}$ Out of sequence packet probability $< 10^{-5}$ Corrupt packet probability $< 10^{-4}$	o High reliability class Packet loss probability $< 10^{-9}$ Duplicate packet probability $< 10^{-9}$ Out of sequence packet probability $< 10^{-9}$ Corrupt packet probability $< 10^{-9}$
Link type assigned to PDP context	o Unacknowledged basic link	o Acknowledged advanced link	o Acknowledged advanced link

A short description of the three multimedia service classes defined for TEDS is summarized as:

a. **Real-time class:** Low delay tolerance. Packet reliability can be compromised for short transmission delays. Example services: video streaming, video conferencing and packetized voice.

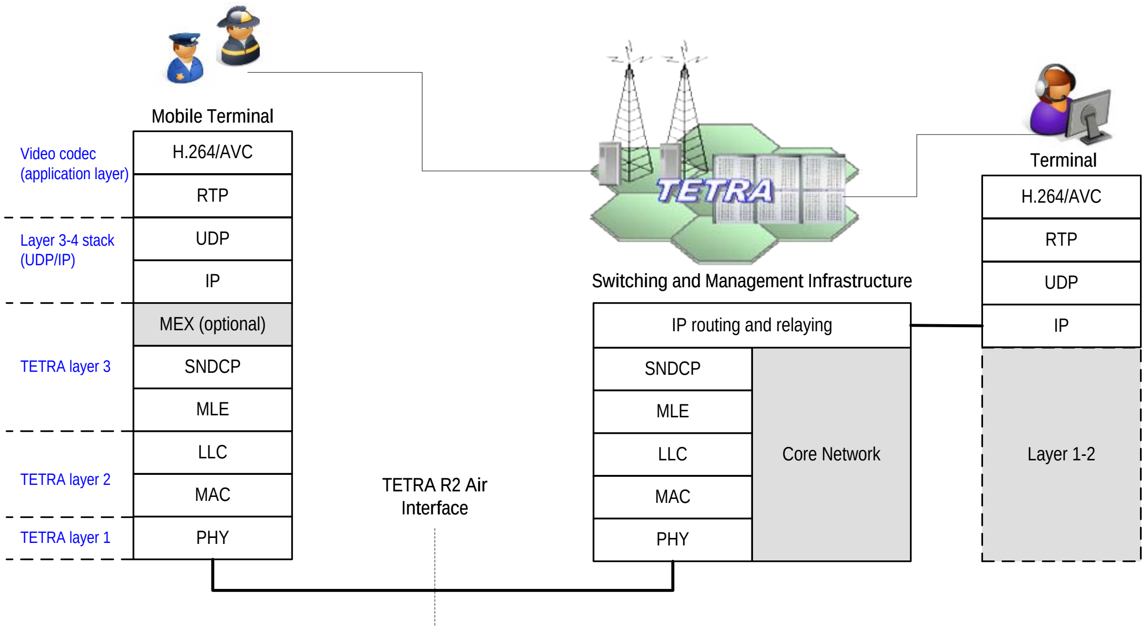
b. **Telemetry class:** Moderate delay tolerance. Packet delivery reliability can be compromised for short transmission delays. Example services: location update, medical telemetry and data logging.

c. **Background class:** High delay tolerance. Stringent reliability requirements. Example services: image transfer, Web browsing and file transfer.

It is evident that video streaming over TEDS will require use of the real-time streaming class over radio channel. According to these classes, traffic types can be assigned different priorities. The explicit implementation of prioritisation of streams is done using a different set of parameters.

3.3.2 TETRA packet data protocol stack

The TETRA Release 2 standards defines a Layer 1-3 protocol stack architecture implementing a TETRA Packet Data Protocol (PDP) that is optimised for handling IP traffic. Figure 3-5 depicts a possible end-to-end protocol stack layers for video streaming over a TEDS link between mobile station (MS) and remote fixed terminal via the TETRA Switching and Management Infrastructure (SwMI). The commonly used protocol hierarchy for conversational and streaming video applications is RTP/UDP/IP [23], whereby, RTP (Real-time Transport Protocol) provides application layer framing and packet loss detection.



Notes: AVC = Advanced Video Coding, IP = Internet Protocol, LLC = Logical Link Control, MAC = Medium Access Control, MEX = Multimedia Exchange Layer, MLE = Mobile Link Entity, PHY = Physical layer, RTP = Real-time Transport Protocol, R2 = Release 2, SNDCP = SubNetwork Dependent Convergence Protocol, TETRA = Terrestrial Trunk Radio, UDP = User Datagram Program. Source: CHORIST project group/TKK ComNet.

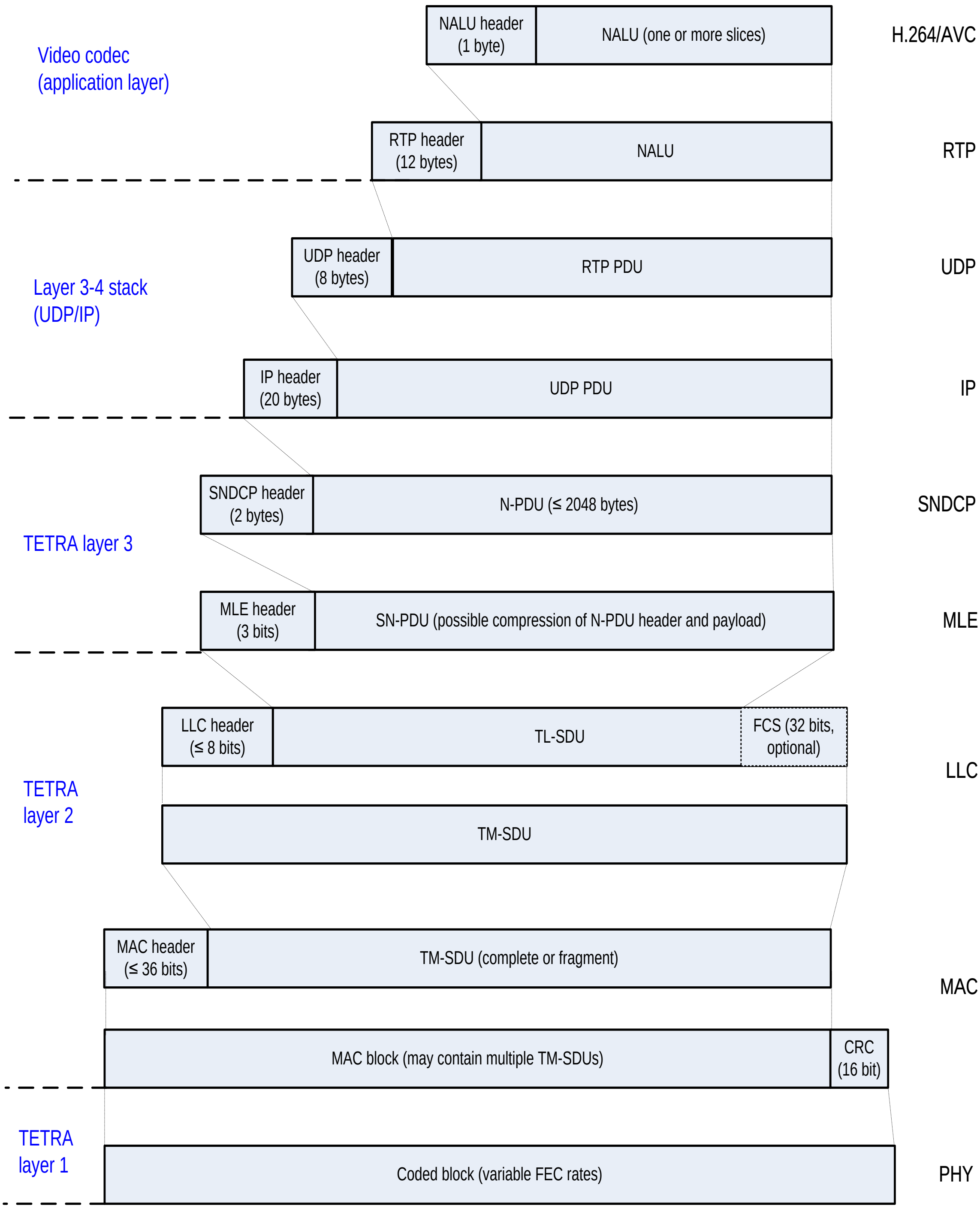
Figure 3-5 TETRA/TEDS Protocol Suite

In the case of H.264/AVC encoded video, the pictures are segmented into slices (which in turn can be segmented in macroblocks), and one or more slices are encapsulated into H.264 Network Abstraction Layer Unit (NALU) packet with a 1 byte NALU header. In simple packetization schemes a single NALU is encapsulated into one RTP packet [23] (refer to section 2.3 for description of actual video packetization scheme used in our tests). The TETRA Subnetwork Dependent Convergence Protocol (SNDCP) layer manages IP packet traffic by establishing the Quality of Service (QoS) requirements of IP packet flows, then buffers and compresses packets from multiple applications (video plus other background traffic), and transfers the data packets across the TETRA air interface using layer 2 services (channel coding, radio channel access control, radio resource management, link adaptation, air interface encryption etc.). Moreover, the SNDCP negotiates and maintains the PDP contexts (typically, QoS parameters for a particular service class) between the MS and the SwMI for each individual application.

An optional Multimedia Exchange (MEX) layer may reside above the SNDCP layer (see Figure above), and is used to routing IP packets from multiple MS applications to the correct PDP context in the SNDCP according to precedence levels (between 0 and 7), whereby, MEX routing buffers with higher precedence levels are emptied more frequently. The Mobile Link Entity (MLE) resides below the SNDCP and its primary function is to initiate cell handover and perform routing to the higher layer entities. The Logical Link Control (LLC) sub-layer provides two types of communication link as a service to the MLE: the basic link and the advanced link. The basic link is available whenever the MS is synchronized to the SwMI, and is usually utilized for unacknowledged transmission of real-time class data [14]. Video streaming is an example service that is typically assigned the basic unacknowledged link type. The advanced link provides a more reliable and efficient method for exchange of large quantities of acknowledged data, such as, packet data transfer for background and telemetry class data.

3.3.3 TETRA PDU video encapsulation

The encapsulation of video over IP in a TETRA PDU (packet data unit) packet is depicted in Figure 3-6; first, the NALU header is appended to the block of video slices to form a NALU segment. This segment is then pre-pended with the RTP protocol header, followed by that of UDP and IP to make up an N-PDU (Network Packet Data Unit) sent to the SNDCP layer. The SNDCP layer adds its own header and passes the packet on to MLE as Sub-Network Packet Data Unit (SN-PDU). The LLC receives the packet from MLE together with MLE header as TL-SDU (Tetra Link Service Data Unit). The LLC issues the PDU to the MAC as a MAC Service Data Unit (TM-SDU). Thus encapsulated by the different headers corresponding to the protocol hierarchy, the packet is transmitted over the air interface from the terminal to base station.



Notes: AVC = Advanced Video Coding, CRC = Cyclic Redundancy Check, FCS = Frame Check Sequence, FEC = Forward Error Correction, IP = Internet Protocol, LLC = Logical Link Control, MAC = Medium Access Control, MLE = Mobile Link Entity, NALU = Network Abstraction Layer Unit, N-PDU = Network PDU, PDU = Protocol Data Unit, PHY = Physical layer, RTP = Real-time Transport Protocol, SDU = Service Data Unit, SNDCP = SubNetwork Dependent Convergence Protocol, SN-PDU = SNDCP PDU, TETRA = Terrestrial Trunk Radio, TL-SDU = TETRA LLC, TM-SDU = TETRA MAC SDU, UDP = User Datagram Program. Source: CHORIST project group/TKK ComNet.

Figure 3-6 TETRA PDU Encapsulation

Chapter 4 . Review of Previous Work

This chapter gives an account of relevant work done previously in the fields of video streaming over test networks using different wireless technologies in general and video transmission over TETRA/TEDS in particular.

4.1 Video streaming

Video streaming has been studied extensively in other mobile communication standardised environments such as Wireless Local Area Networks (WLAN) and Universal Mobile Telecommunication System (UMTS) under development by the 3rd Generation Partnership Project (3GPP). These scenarios can be considered close to our TEDS scenario as they have similar functionality (to provide a wireless, transparent bearer service with fixed capacity) but different implementations and capabilities. A study about the capability of WLAN 802.11b,g and draft n standards to stream multiple High Definition (HD) videos using MPEG-4 Part 2 non-scalable encoding scheme (another name for the H.264/AVC standard) uses a similar approach to the one employed in this study for evaluating received visual quality of video streaming over these network links [27]. It uses VLC media player to stream video over actual networks setup and record the decoded version at each end for later evaluation with a different software tool. The purpose of the study was to measure video streaming capability of WLANs by comparing decoded video quality performance between the different versions of the 802.11 standard.

Regarding video streaming in general, it has been demonstrated that the interplay of the three main factors of available link bandwidth, propagation delay and packet loss affecting video streams along with the video encoding or codec rate, can help to dimension video streaming systems. One observation apparent is that video encoded at a particular rate is most sensitive to link bandwidth constraints, that is, if video is encoded at higher codec rate than available link bandwidth, video quality is affected. On the other hand, the affects of propagation delay are found to be not so straight-forward. It appears that increasing the delay in some cases can lead to better performance depending on codec rate and link bandwidth [24]. We intend to study video over TEDS taking these factors into account.

4.2 Video over TETRA/TEDS

To the best of our knowledge, there have been two studies regarding video transmission over TETRA. The first study employs simulation techniques to transmit MPEG-4 encoded video over modelled RF channels using system specified (TU50, Typical Urban at 50km/hr) propagation conditions [30]. The paper focuses on interaction of radio interface characteristics with the error resilience features of the codec. Average Peak Signal to Noise Ratio (APSNR) measures of the decoded video compared with original reference are used to quantify system performance. Specifically the performance is measured first, with respect to channel BER conditions and traffic channel capacities employed for transmission. Then, similar tests are repeated with the addition of error resilience features of the codec.

The second is a Master's thesis here at TKK that investigates the problem of choosing an appropriate video codec for video transmission over TETRA, by making a comparison of different video codecs in terms of achieved video quality. The study takes into account the constrained link bandwidth and studies factors such as frame loss and CPU load on the performance of video in terms of subjective video quality metrics. Through numerous tests, the study arrives at the conclusion that H.264 codec is the most suitable for this video transmission scenario [31].

We intend to study video transmission over the TEDS standard that is capable of providing higher bitrates than TETRA. The study mentioned above [30] makes use of the highest bitrates afforded by TETRA as 21 kbps. Also we focus more on the quality of decoded video by relying on two different kinds of objective visual quality measurement metrics instead of just one. We are interested in investigating the feasibility of video transmission over the link bandwidths provided by TEDS transparently, without regard to channel conditions as we assume that TEDS automatically scales to achievable bitrates accordingly. The objective is to investigate video streaming performance over TEDS at the various achievable bitrates.

Chapter 5 Experimental Setup

This chapter provides an overview of the experimental settings, as well as a preview of the tools and metrics employed for the analysis of experiment trials.

5.1 Approach

The unavailability of actual TEDS equipment (base station, modem, etc.) for the purpose of this study, means we have to resort to simulating a link that mirrors data rates provided by the TEDS standard in our test bed. This link will provide the bandwidth-constrained environment setting in which we study video streaming performance.

An important distinction to be made at this point is that most studies of transmitted video quality are made via simulations; using software based encoding and simulated channel environments. Simulation is by definition an imitation of a real process in abstract terms. Simulation studies involve testing technologies by modelling the system characteristics and predicting best and worst case behaviours by changing certain key system parameters. In this way, expected real results can be forecasted and adjustments to technology under development can be made accordingly to achieve desired results. This is often the pre-prototyping phase in most technology life-cycles. This study opts to go a step further and use emulation instead of simulation. Emulation refers to a simulation that interacts with real network components to give a realistic measure of the performance.

This approach allows us to use the same kind of streaming software as would be used in real-life application, and observe the interaction of the real-time traffic generated in this way, with the simulated link. Such representation of a TEDS link is adequate for our purpose because the focus of this work is on traffic handling capability of limited bandwidth links. It is assumed that the TEDS interface can provide the link capacities as mentioned earlier; hence, the underlying radio technology is not modelled. The real video traffic is generated by transcoding (encoding and transmitting) pre-recorded clips on the fly before being transmitted, thus giving a close approximation of live video streaming. Although it is quite possible to use live video feeds from webcams, that practice leaves room for some ambiguity regarding the results obtained, as it cannot be assured that video content will be exactly duplicated from one trial to another (except perhaps in the case of video based on still images). So, to ensure repeatability, we use pre-recorded video clips.

As shown in Figure 5-1, our setup consists of three nodes, a video transmitter at one end of the link; a video receiver at the other end of the link; and a middle node to simulate the link. Physically, the network is built using LAN cables and a switch. The real network components (2 laptops that act as transmitter and receiver of a video stream) have a virtual representation within the simulation environment. The virtual nodes enable interaction of the link simulation with the real environment. The simulator software injects packets received from the real network components and ejects them back onto the real network after processing.

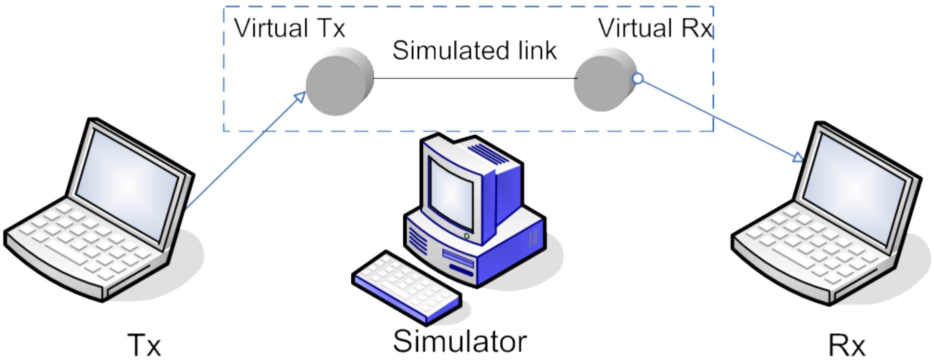


Figure 5-1 Experimental setup

5.2 Metrics

In this section, the metrics used to evaluate network performance in this study are presented. There are two categories of metrics used, one set for video quality measurement designed to give an estimate of end user perspective; and the other for performance measurement from network point of view.

5.2.1 Video quality metrics

Video codecs attempt to perform compression (reduction in storage space or transmission bandwidth) by removing redundancy. The preservation of some degree of visual quality, with highest possible compression ratio is the basis on which the codecs are rated in terms of performance. It has been attempted to qualify visual quality in terms of metrics of two types; objective and subjective. Most of these metrics operate on the basic principle of comparison with some reference; in this case, the original video sequence.

Subjective metrics are usually in the form of Mean Opinion Scores (MOS) based on collected and averaged statistics of people's responses to video quality surveys. These surveys involve showing a number of randomly picked people selected videos (both original and decoded versions) in a pre-determined order and asking them to rate the videos on a chosen quality scale [43]. Subjective testing is therefore highly expensive in terms of time and resources, and not exactly repeatable. However, it is considered more reliable than objective testing simply because of the fact that it is people who are the end viewers and therefore the best judge of perceived quality.

Objective metrics refer to algorithmically calculated video quality measures that model end-viewer response [44], that is, a highly rated objective score should translate to a high subjective MOS. This approach to determine visual quality can reduce testing time and cost. There are two types of objective metrics, the data metrics and perceptual metrics. Data metrics calculate fidelity of video signal without considering video content through either picture difference methods that focus on how closely processed signal resembles original source signal, or parametric methods that focus on network performance impact on video signal quality (in terms of bitrates, frame loss , jitter, etc.). Perceptual metrics focus on predicting quality of video signal as perceived by end viewers; by analysing characteristics of video signal content and the impact of changes in these on the processed signal [45].

Objective metrics provide a quantifiable, repeatable means of judging reproduced video quality. There are further three types of reference models for video quality metrics based on amount of required information of the original (reference) video. Full-reference (FR) metrics perform frame-by-frame comparison between reference and test video. No-reference (NR) metrics analyze only processed video signal where the challenge lies in distinguishing content from distortion. Reduced-reference (RR) methods have only some information of characteristic features of the original source video that it uses to aid quality prediction [44] [45].

We use pre-recorded video sequences in our experiments so that FR methods of video quality evaluation can be applied, for better reliability of results. We use both data and perceptual metrics to allow for a more complete visual quality assessment. The metrics used are elaborated as follows;

5.2.1.1 PSNR

Peak-signal-to-noise-ratio (PSNR) or average PSNR is an example of data metric that uses picture difference method. It is a widely used objective metric for video quality evaluation that provides an estimate of the quality of a codec reconstructed video as compared to the original, uncompressed version. However, it is equally widely acknowledged that it is not an accurate measure of perceived visual quality as it fails to take into account the human visual perception factor; rather it relies on pure computation in terms of a pixel by pixel comparison [46]. The ease of computation is mostly responsible for its popularity. The noise in PSNR refers to quantization errors introduced during video encoding and decoding, also referred to as Mean Squared Error (MSE).

MSE between a received video sequence *I* and reference sequence *R* is given by

$$\mathrm{MSE} = \frac{1}{ZXY}\sum_t \sum_x \sum_y \left[R(t,x,y) - I(t,x,y) \right]^2, \tag{5.1}$$

for video sequence of size $X \times Y$ pixels per frame and *Z* frames in the sequence. The PSNR in dB is then;

$$\mathrm{PSNR} = 10\log\frac{L^2}{\mathrm{MSE}} = 20\log\frac{L}{\sqrt{\mathrm{MSE}}}, \tag{5.2}$$

where *L* is a constant representing the dynamic range of image pixel intensities (e.g., for 8 bits/pixel gray-scale image, $L = 2^8 - 1 = 255$) and $\sqrt{\mathrm{MSE}}$ is the root mean square error. PSNR focuses on the luminance factor as most important to picture quality and therefore, the MSE is usually calculated for luminance signal only.

Typical values of PSNR range from 20 dB to 50 dB. This value has no absolute meaning as it is a ratio meant to be used as a reference scale; generally the higher the

PSNR value, the better the reproduced visual quality. A comparison of a video sequence with an exact copy of itself results in a PSNR value of infinity.

5.2.1.2 SSIM

Structural SIMilarity (SSIM), marks a departure from the usual error-sensitivity related approaches in the field of objective video quality measurement. It redefines the approach by designing a metric to capture structural distortion as a measure of perceived image distortion [47]. It is modelled on the human visual system (HVS) perception model. The HVS is specially adapted to extracting structural information from the viewing. Therefore, it follows that a loss in structural information in reproduced video would be most noticeable to the human eye in the form of picture degradation. Structural information refers to those attributes that contribute to structure of objects in a scene, independent of luminance and contrast.

The SSIM has been found to be a better measure of perceived quality than PSNR in various tests with images compressed on the JPEG2000 standard. For the same MSE, the picture quality is shown to vary drastically, while SSIM reflects these differences successfully as shown in Figure 5-2. The value of SSIM ranges from 0 to 1, the higher values indicating a closer match to the reference video which is considered of better quality.

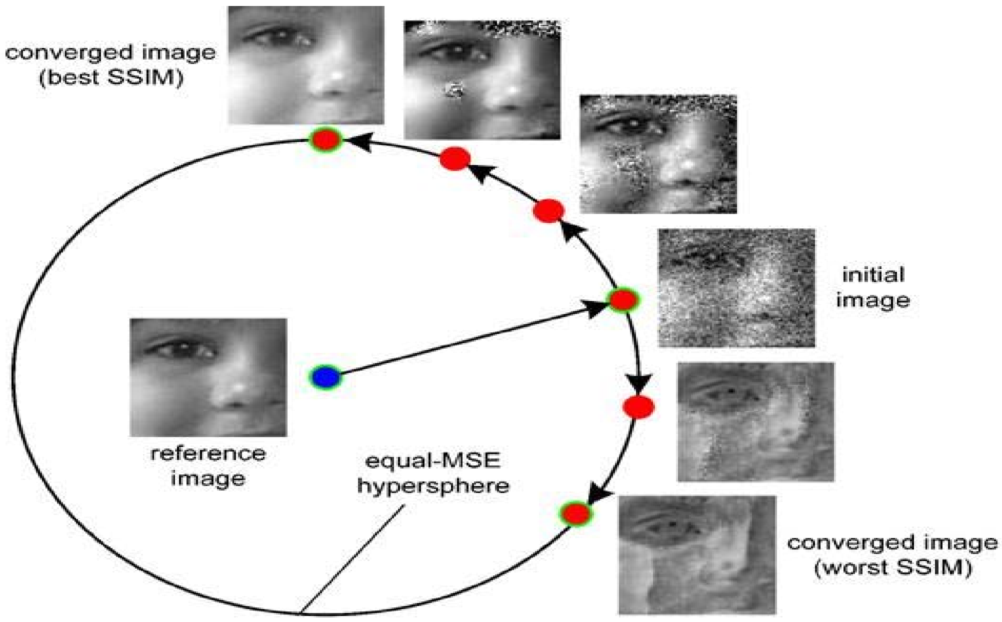


Figure 5-2 SSIM vs. MSE [48]

The SSIM index is calculated between two image patches extracted from the same spatial location from the reference *R* and distorted *I* image as a function of a luminance comparison term *l*, contrast comparison term *c,* and structure comparison term *s,* as;

$$SSIM[R,I] = l[R(t,x,y), I(t,x,y)]c[R(t,x,y), I(t,x,y)]s[R(t,x,y), I(t,x,y)], \quad (5.3)$$

where luminance comparison term *l* is given by;

$$l[R(t,x,y), I(t,x,y)] = \frac{2\mu_{R(t,x,y)}\mu_{I(t,x,y)} + C_1}{\mu^2_{R(t,x,y)} + \mu^2_{I(t,x,y)} + C_1}, \quad (5.4)$$

and contrast comparison term *c* is given by;

$$c[R(t,x,y), I(t,x,y)] = \frac{2\sigma_{R(t,x,y)}\sigma_{I(t,x,y)} + C_2}{\sigma^2_{R(t,x,y)} + \sigma^2_{I(t,x,y)} + C_2}, \quad (5.5)$$

and structure comparison term *s* is given by;

$$s[R(t,x,y), I(t,x,y)] = \frac{\sigma_{R(t,x,y)I(t,x,y)} + C_3}{cov_{R(t,x,y)I(t,x,y)} + C_3}, \quad (5.6)$$

where $\mu_{R(t,x,y)}$ and $\mu_{I(t,x,y)}$ are the sample means of image patches *R(t,x,y)* and *I(t,x,y)*; $\sigma^2_{R(t,x,y)}$ and $\sigma^2_{I(t,x,y)}$ are their sample variances and $cov_{R(t,x,y)I(t,x,y)}$ is the sample covariance. The constants C_1, C_2 and C_3 were included in SSIM index as an improvement to the original Universal Quality Index (UQI) to avoid instability when the denominators of the luminance, contrast and structure terms were too small.

5.2.2 Delay metric: Inter-Packet Delay Variation

In order to measure the effect of delay introduced by the link on video performance, we calculate the inter-arrival delay between packets of a single flow at receiving end, and compare this to the inter-arrival delay between packets at transmitting end. This helps us to estimate the jitter introduced by the link that affects the transmitted video packet flow. RFC 3393 defines jitter in two ways, the second of which is relevant to this study. It is defined in the following excerpt;

“The second meaning has to do with the variation of a metric (e.g., delay) with respect to some reference metric (e.g., average delay or minimum delay). This meaning is frequently used by computer scientists and frequently (but not always) refers to variation in delay. In this document we will avoid the term "jitter" whenever possible and stick to delay variation which is more precise.”

Delay variation quantifies a path’s ability to transfer packets with consistent delay [50]. One important use of delay variation is the sizing of playout or de-jitter buffer for applications requiring the regular delivery of packets (for example, video playout buffer).

The Inter-Packet Delay Variation (IPDV) metric provides a means to compare the difference in one-way delay profiles for video flows as transmitted and received in the following manner. For packets in a stream consecutively numbered i = 1; 2; 3; within a particular test interval, IPDV is given by [49] [50]:

$$IPDV_i = D_i - D_{i-1} \tag{5.7}$$

where D_i denotes the one-way delay of the *i*th packet. The one-way delay is equal to the difference between timestamps applied at the ends of the path, or the receiver time minus the transmission time. The IPDV can take on positive, negative and zero values. It can be shown that IPDV also represents the change in inter-packet spacing between transmission and reception [50]:

$$IPDV_i = (U_i - V_i) - (U_{i-1} - V_{i-1}) = (U_i - V_{i-1}) - (U_i - V_{i-1}), \tag{5.8}$$

where U_i and V represent the *i*th packet received and transmission times respectively.

5.3 Tools

This section lists the software tools and resources used in the experiment test bed setup. Along with a short description of each, reasoning behind the choice of tool is also presented.

5.3.1 QualNet

QualNet is a high-fidelity network modelling and evaluation tool that can be used to simulate mixed platform networks and networking devices [35]. This serves as the link simulator for our scenario. The HITL (Hardware-in-the-loop) capability of QualNet is utilised to interface external machines with the QualNet simulated link. This allows the simulator to be fed with external input of live video traffic in an emulation scenario. Emulation-based testing and analysis can provide a more accurate prediction of real-life network performance [36].

5.3.2 MGEN

The 'Multi-Generator' is open-source software that uses scripts in order to model different kinds of real-time network traffic patterns. The generated traffic can be received and logged for post-analysis using the companion DREC (Dynamic Receiver) software which also uses scripts to drive reception model over time [37]. MGEN is used to verify our emulation against simulation to ensure consistency in the link's performance in terms of bandwidth and delay. It is used to benchmark the link's performance for UDP-based constant bit-rate (CBR) generated using simple models. This benchmark can then be used as reference to compare link performance for UDP-based video streams.

5.3.3 TRPR

'TRace Plot Real time' is an analysis tool that can take MGEN/DREC generated logs as well as Tcpdump logs as input and generate time-based plot data as output. The default output is rate vs. time plot data. TRPR allows very flexible use of filters to extract data of interest from the collected logs [38]. TRPR is used to analyse all traffic traces and extract traffic metrics from these, to ensure consistency across results obtained regardless of packet sniffer/generator used (that is, MGEN, iperf or TCPdump). The output of TRPR is used to plot the various graphs in this work using MATLAB instead of gnuplot, to allow manipulation/conversion of collected data for convenient plotting.

5.3.4 Tcpdump

Tcpdump is a packet-sniffing software program that 'dumps' traffic seen by the host running Tcpdump on the network. Tcpdump allows flexible usage of filters to collect traffic of interest, that is, by capturing packets with headers that match the specified flow. Furthermore, it can also write these logs to file for post-analysis [39]. Tcpdump is used to collect traces of video streams from which traffic metrics such as bandwidth consumption and packet inter-arrival times are extracted using TRPR. Hence, it allows fine control over network performance monitoring.

5.3.5 VLC media player

The VLC (formerly known as VideoLAN Client) media player is open source software that can support a wide variety of codecs. It can be used as unicast or multicast streaming server with a host of streaming control options [40]. VLC is used both for streaming and receiving video in our scenario. It is also used to encode and decode the transmitted and received video at the respective ends. VLC is chosen for our purpose because it provides all three functions of video coding/decoding and display, a variety of codecs and video streaming, in one package. Moreover, it can save the transmitted and received video streams to file in .mp4 format for post-streaming quality comparison.

5.3.6 EvalVid

EvalVid is a video quality evaluation tool-kit developed by researchers at TKN Berlin. It is targeted at research work that requires network performance evaluation in terms of end-user perceived video quality. The development and extension of EvalVid has been the subject of thesis papers [42]. Its use in other studies involving video quality evaluation further cements its usefulness to the research community. Although we do not utilize the full toolbox of the EvalVid, it was chosen from other available quality evaluation tools for its simple command line interface that gives fast and precise results. The supported video quality metrics are Peak-signal-to-noise-ratio (PSNR) and Structural similarity (SSIM).

5.3.7 FFmpeg

FFmpeg is a tool that provides a complete cross-platform solution to record, convert and stream video. Its supports a wide variety of codecs and includes the audio/video codec library libavcodec. It sports an intuitive command line interface in the sense that it tries to figure out the default settings for video parameters that may be derived automatically, whatever it is asked to do whether converting, capturing from source, etc. We use the tool to convert encoded video to raw video (.yuv) format for input to EvalVid.

5.4 Testing approach summarized

We use three different pre-recorded clips, hereafter video A, B and C in order to identify whether the received video quality is also affected by the video content or not. The video B is the most stable video with almost no scene changes, while the background remains fairly constant. On the other hand, the videos A and C contain frequent scene changes and rapid panning in and out. All the three videos are initially available at the transmitter's side in Windows Media Video (WMV) format. We use VLC to encode them at various rates using H.264/AVC and transmit them over the simulated TEDS link. Then, we inject the video packets back into the real network. The figure below provides a recap of the test-bed scenario employed in our experiments; detailing the tools used and their corresponding metrics.

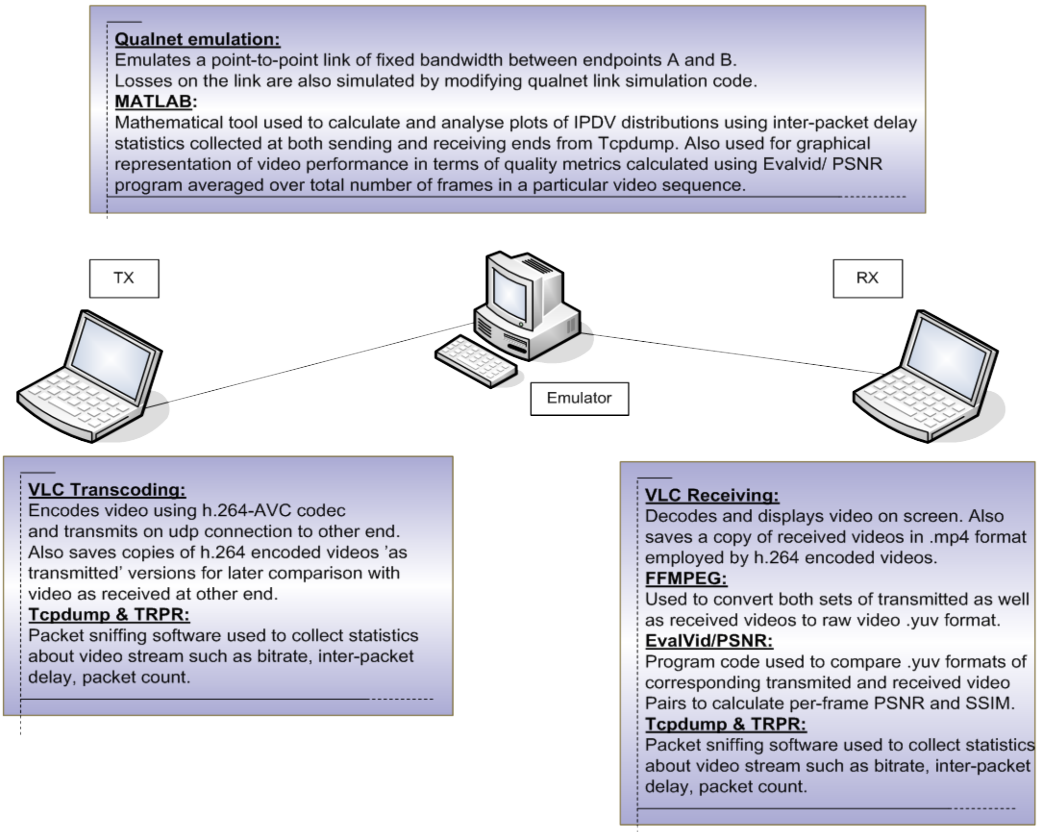


Figure 5-3 Experimental setup (Functional)

To realistically model the TEDS link we append a 79-bit header at the IP packet after being sniffed by the simulator. The size of the header is equal to the aggregate sum of the 2-byte SNDCP, 3-bit MLE, 8-bit LLC, 36-bit MAC header and the 16-bit cyclic redundancy check (CRC), as shown in TETRA PDU encapsulation Figure 3-6. The quality comparison of the videos saved at transmitting and receiving end for each codec rate and link bandwidth combination is carried out using Evalvid. Prior to Evalvid measurements, the received and the transmitted video (both MPEG-4 format) are converted to YUV format by using FFmpeg software. While streaming video, we use Tcpdump to sniff and capture packets on the sending and receiving end. These Tcpdump logs are parsed using the TRPR program to calculate per-packet, instantaneous throughput and inter-packet delays. These metrics are then imported in Matlab to calculate IPDV.

The data rates selected for the simulated TEDS link are 150 kbps and 300 kbps. These roughly correspond to the peak uplink data rates for 50 kHz and 100 kHz channel bandwidth respectively (Table 3-2). In both cases the modulation scheme is QAM. Using average MER values for the uplink SCH-Q channels in Table 3-3 as reference, we simulate a uniformly distributed 5% and 10% packet loss on the Qualnet emulated link for packet loss scenarios. Assuming no packet segmentation in layer 2 and above, and with the use of a basic unacknowledged link, the MER is considered to be equivalent to packet loss probability.

Our approach, however, excludes modelling of RF environment in terms of signal modulation schemes. We also do not take the effect of over-the-air-retransmissions due to errors in transmission into account. Futhermore, we implement a uniformly distributed packet loss model on the link to simulate a lossy environment, as opposed to the bursty nature of real loss environment.

Chapter 6 Verifying Emulation Behaviour

Before conducting any tests we need to determine that our emulated link functions as expected. To this end, we conduct some trials using MGEN traffic generator to generate CBR traffic at ,transmitting end and send over the emulated link (through the machine running QualNet) to the receiving laptop. The MGEN and QualNet emulation settings are described in appendix A.

6.1 Throughput profile

Figure 6-1 illustrates the average UDP throughput at different data rates for link bandwidths of 150 kbps and 300 kbps plotted against analytically calculated throughput curve corresponding to the input flow bit rate. UDP flows were generated at controlled data rates by the MGEN packet generator. This test aims to verify trend of the UDP-based flows’ performance on the link.

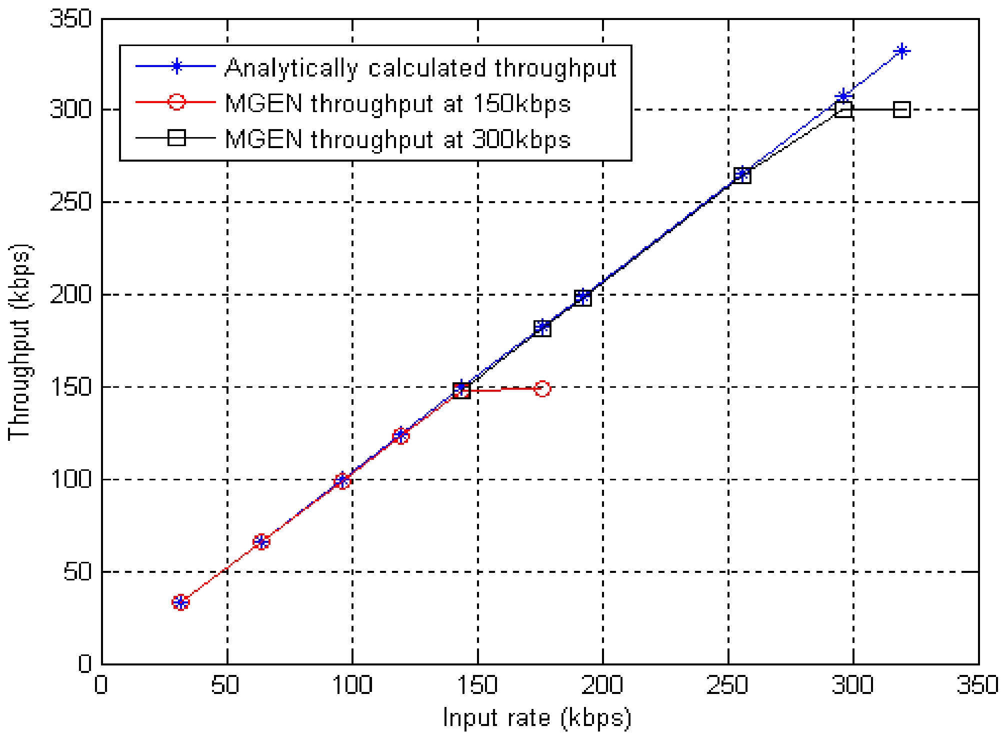


Figure 6-1 UDP flow throughput vs. link bandwidth

MGEN generates packets of fixed size of 1000 bytes. The different input bit rates are generated by assigning number of packets *p* to be sent per second. So the throughput *S* (vertical axis in Fig. 16) is analytically calculated as;

$$S = p \cdot (N_u + N_h), \qquad (6.1)$$

where N_u represents packet size and N_h represents header size. In our tests, 1000 bytes is the packet size, 28 bytes make up the UDP and IP headers and 79 bits make up the TETRA lower layer headers. The plot shows a consistent trend in the behaviour of the link at the different bandwidths. At UDP input data rates close but below the link bandwidth, the achieved data rate on the link matches the analytical curve. Note that the Tcpdump tool collects statistics at link layer so the readings plotted here are inclusive of headers, which is why the data rate achieved on the link is slightly higher than input rate.

The delay experienced by each packet is due to propagation delay in transmission of packet plus overhead. Therefore delay T_d is given by;

$$T_d = \frac{N_u + N_h}{B}, \qquad (6.2)$$

where *B* represents the link bandwidth. For *B* equal to 150kbps and 300kbps, the calculated delays are 0.0554s and 0.0277s respectively.

The UDP flow data rates are selected close to the available link bandwidth for both 150 and 300 kbps especially to illustrate the maximum achievable application level throughput at the various link bandwidths. According to the delay values calculated above, the theoretically achievable maximum application level throughput for each of the links is $\frac{N_u}{T_d}$, when N_u is 1000 bytes. For 150kbps link bandwidth, the maximum throughput thus calculated amounts to 144.53kbps, whereas for 300kbps link bandwidth, it is 289.05kbps. Thus, we see that a link does not allow data flows passing through it to use bitrates higher than nominal link bandwidth, in fact actual utilized link capacity will always be below the nominal value.

These link bandwidth imposed throughput limits can be seen in the Fig.16 as the slope of the curve levelling out. These upper limits are defined by the propagation delay experienced by packets on the links. This propagation delay is dependent on link bandwidth and packet size inclusive of headers. At input data rates higher than link bandwidth, packet loss is observed due to excess packets that the link cannot carry being dropped. It is to be noted that the values plotted here are averages of instantaneous, per-packet throughput measurements by Tcpdump, hence they tend to be slightly greater than the analytically calculated limit. Nevertheless, it is verified that the simulated link bandwidths are limited to their nominal value.

6.2 Inter-packet delay profile

Figure 6-2 shows a plot of average one-way inter-packet delay at the receiving node for different input rates generated using MGEN. The plot indicates that the average inter-packet delay decreases with increasing input data rates. This is to be expected as increasing data rate implies sending more packets per second on the link which naturally reduces spacing between packets in a stream.

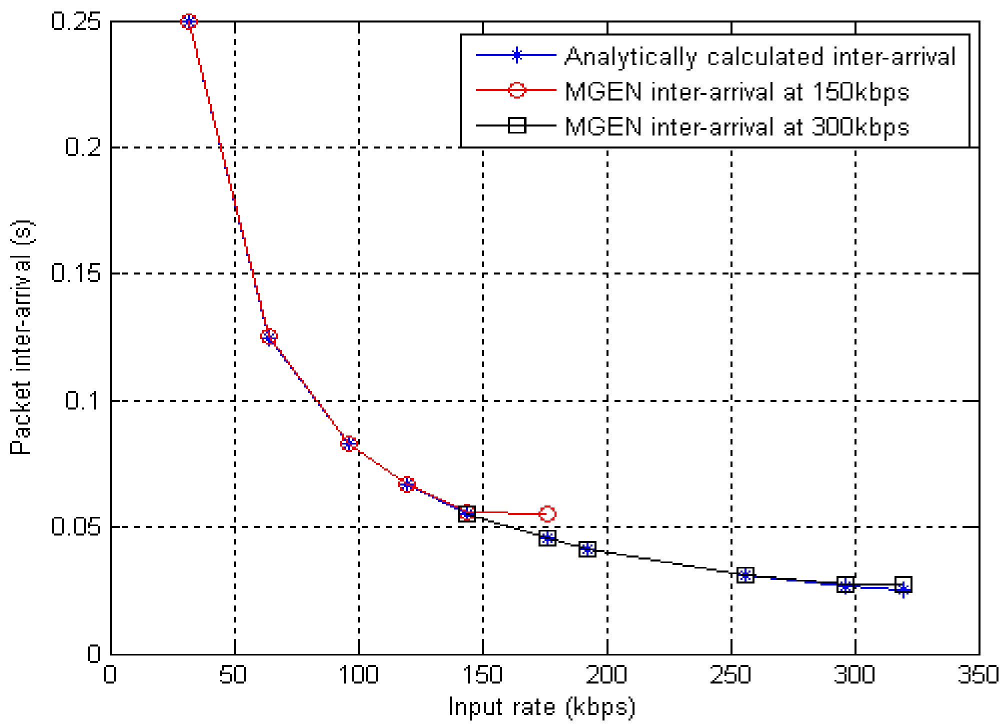


Figure 6-2 Average inter-packet delay at the receiver using MGEN at the transmitter

The inter-packet arrival time is analytically calculated as inverse ratio of number of packets sent per second, i.e. $\frac{1}{p}$. Packet inter-arrival time can be taken as a measure of inter-packet spacing in seconds. The bandwidth induced limitations of the links are again visible as deviations from the analytical curve, implying that for input flows at bitrates higher than link bandwidth, the link is unable to support the shorter inter-packet spacing required.

Chapter 7 Performance Evaluation Results

Video streaming performance can be measured from a link layer perspective as well as application layer perspective. To indicate performance at link layer, the used metrics include throughput in terms of stream bitrate required to deliver videos transcoded at particular codec rates, as well as corresponding IPDV. From an application layer or end-user perspective, video quality measures are more important. We use averages of per-frame PSNR and SSIM values as well as CDF plots of the same to depict video quality performance.

It is important to note that for calculating video quality metrics, the received video is compared with its transmitted version at a particular codec rate, instead of being compared with a reference video at a fixed (high) codec rate. The logic behind this manner of testing is to allow for choice of codec rate taking into account bandwidth consumed on the link along with fidelity of transmitted video (codec performance). The general rule is that the usage of higher codec rate at the transmission end results in better visual quality at the receiver. However, this approach results in large bandwidth consumption on the link and evaluates solely the codec performance without taking into account the link constraints. To sum up, the PSNR and the SSIM values in our experiments depend on the comparison between the transmitted and the received video.

The results are presented in the following order; first, we take a look at effect of video content on achieved throughput on a link bandwidth of 150 and 300 kbps at different codec rates. Then we proceed to measure video quality for increasing codec rates and repeat the same observations for link losses of 5% and 10%. Next, we take a look at link layer characteristics of video streaming at different codec rates on different link bandwidths by IPDV measurements. This leads us to investigate the effect of play-out buffer size on video quality. We conclude the results section by observing effect of prioritizing streams in the presence of a competing traffic flow on received video quality.

7.1 Video Streaming Throughput

Now that link behaviour has been determined for controlled UDP flows, we test the link using traffic generated by VLC media player during transmission of video from node 1 to 2. The objective is to record the data rates employed by VLC using the H.264 codec to transport three different videos named A, B and C at different codec rates respectively.

It is observed in Figure 7-1 that the data rates achieved on the link by particular codec rates remain roughly constant for all three videos. This implies that video content does not impact significantly on utilized link capacity. In addition the output data rate for a particular codec rate does not depend on the link bandwidth. It should be noted that the codec rate depicted on x-axis is proportional but does not correspond to input data rate. The codec rate determines the desired or target output rate of video. However, the actual achieved rate on the link is usually higher. The rate control mechanism of a codec (which is codec-specific) dictates how well the achieved output matches target bit rate as set by codec rate.

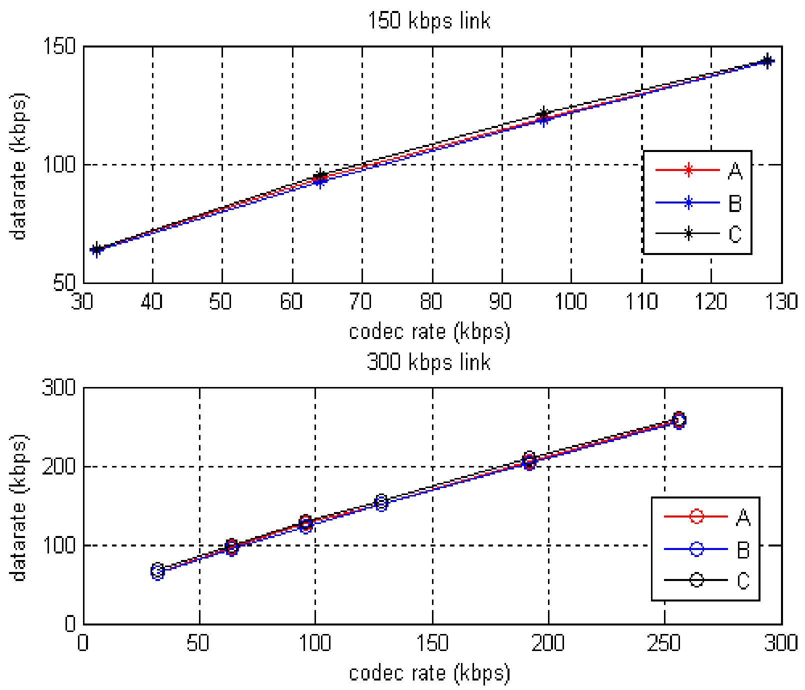


Figure 7-1 Achieved datarates on link at various codec rates

A plot of average packet inter-arrival times for the three videos (not included) also indicates that video content does not affect packet inter-arrival times at a particular codec rate. The packet inter-arrival trend (decreasing with increasing codec rate) is similar to that observed for MGEN generated traffic, as expected.

7.2 Video Quality Performance

In this section, performance analyses of video transmission at particular link bandwidth in terms of objective video quality metrics PSNR and SSIM are presented. There are three different cases of observation. The first case considers an ideal link with no packet drops. The rest consider the effects of 5% and 10% packet loss on the link on the received video quality.

7.2.1 No Packet Loss

Figure 7-2 depicts the mean PSNR and SSIM values at different link bandwidths achieved by different codec rates of H.264. The mean values are obtained by averaging the per-frame PSNR and SSIM values over the entire duration of the video sequence. The average value is taken as a general valuation of the overall quality of received video.

It can be seen that at codec rates upto 128 kbps, the average PSNR and SSIM are roughly the same at link rates, 150 and 300 kbps. This is due to the testing methodology explained previously Observe that for videos B and C at 128 kbps codec rate and link bandwidth 150 kbps, the performance metrics degrade. This is to be expected as the output data rate approaches the link capacity. That is, however, not the case for video A, so it seems that while video content does not affect the output data rates, it plays a part in how video quality is affected.

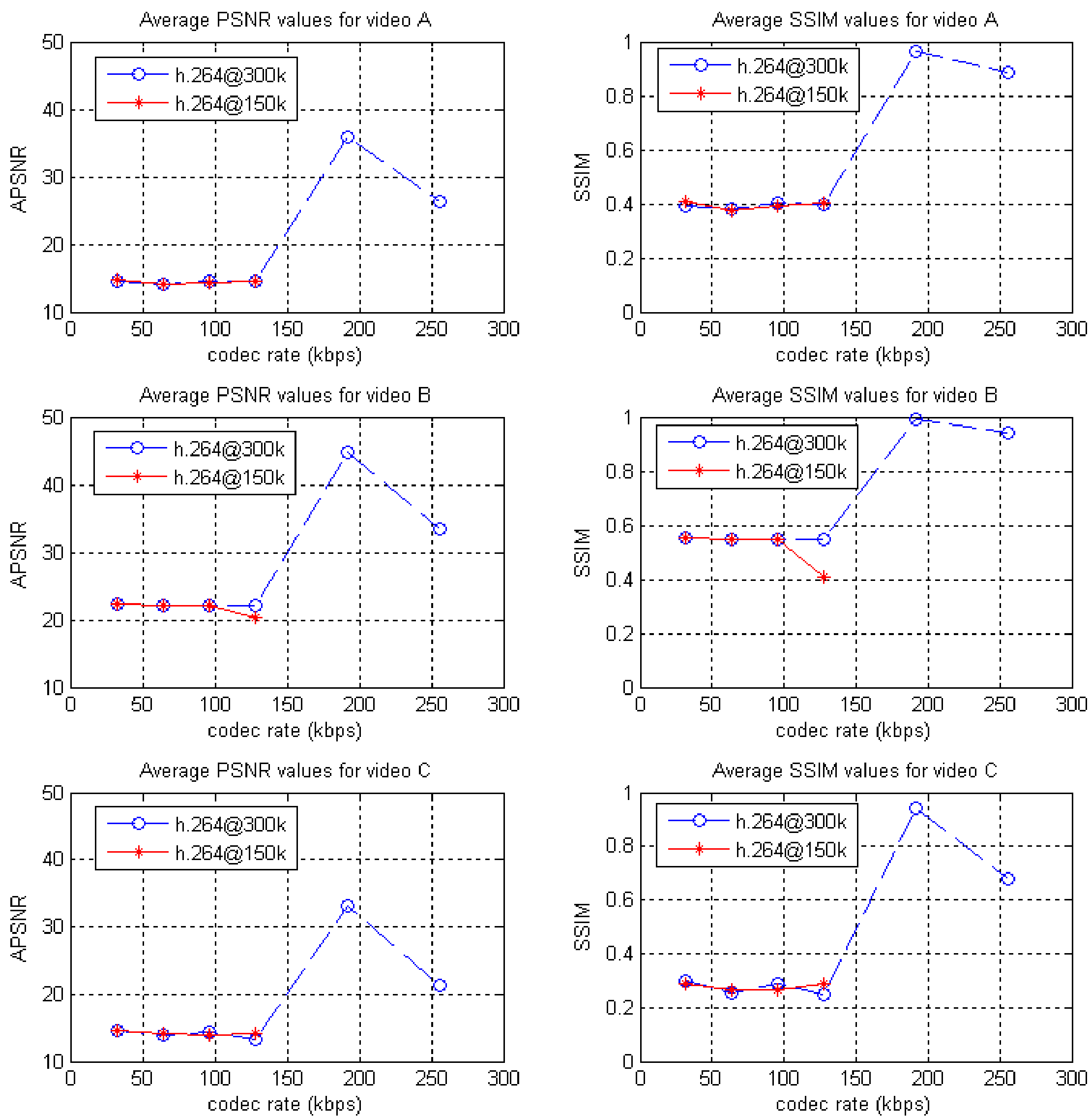


Figure 7-2 Video quality without packet loss at different link rates

Both SSIM and PSNR plots in Figure 7-2 show an agreement with each other regarding video quality trends for all three videos. However, no direct relationship can be claimed between a particular SSIM and PSNR average value, that is, they are essentially independent metrics. It is immediately clear from the graphs that the best performance results from the use of H.264 at 192 kbps codec rate on a 300 kbps link. This result is consistently observed for all three videos.

It is observed that for a particular link bandwidth, switching to higher codec rates does not necessarily result in an improvement in PSNR or SSIM (visual quality) as might be expected. This can be explained by the method of video comparison used (explained above). We compare the received to the transmitted video. Therefore the performance of higher codec rates in terms of calculated PSNR/SSIM can be the same while the link utilization is increased.

a) 64kbps b)96kbps c)128 kbps d)192 kbps

Figure 7-3 Illustrating visual quality at different codec rates

As can be seen in the screenshots of Video A encoded at different codec rates in Figure 7-3, visual quality gradually improves as codec rate is increased. At 96 kbps codec rate, both background and foreground is blurred while colours are sharper than at 64 kbps. At 128 kbps, the foreground sharpens while at 192 kbps both background and foreground exhibit improved sharpness as compared to the other videos. Note that the quality depicted in the stills is continuously varying in actual video sequence.

7.2.2 With Packet Loss

In this section, received video quality is compared against transmitted video quality with and without packet loss on the link, for both link rates of 150 kbps and 300 kbps, separately. Figure 7-4 shows comparison of video performance at packet loss rates of 5% and 10% against the case with no packet loss for 150 kbps link bandwidth and there appears to be some slight but no major variation at most codec rates. However these slight variations represent disturbances in the video signal that are plainly visible and obvious to the human eye. A higher loss percentage is seen to result in slightly more visual quality degradation, as expected.

a) Video without loss b) Video with 5% loss c) Video with 10% loss

Figure 7-4 Video B coded at 96 kbps at different channel loss conditions

Implementing packet loss on the link means incurring some frame loss at the application layer. Consider the case where only the first few frames are received and the rest are lost. The absence of some frames at the receiver can mean a skewed computation of mean PSNR and SSIM values unless the missing frames are considered in the calculations. However even with packet loss, it is possible to reconstruct the entire video sequence such that there are no missing frames, using error concealment (EC) techniques. These techniques may involve either reconstruction of a damaged frame or repeating previous frames in place of missing frames (look for a reference on EC). In the calculations of PSNR/SSIM we ensure that the number of frames compared remains equal.

For our study, we rely on the FFMPEG tool's EC method. The EC techniques are applied by FFMPEG tool during conversion to raw video format for video performance metric computation. These are similar to the ones employed by VLC during video display and hence do not represent any alteration of received video, rather it mimics video player behaviour. In cases where complete frame recovery is not possible due to key frame loss, we eliminate all such points in the plots for APSNR. For SSIM, we compute the average value by adding zeroes equivalent to the number of missing frames, under the assumption that the comparison of missing frames with the corresponding frame would result in a 'no-match' or zero value.

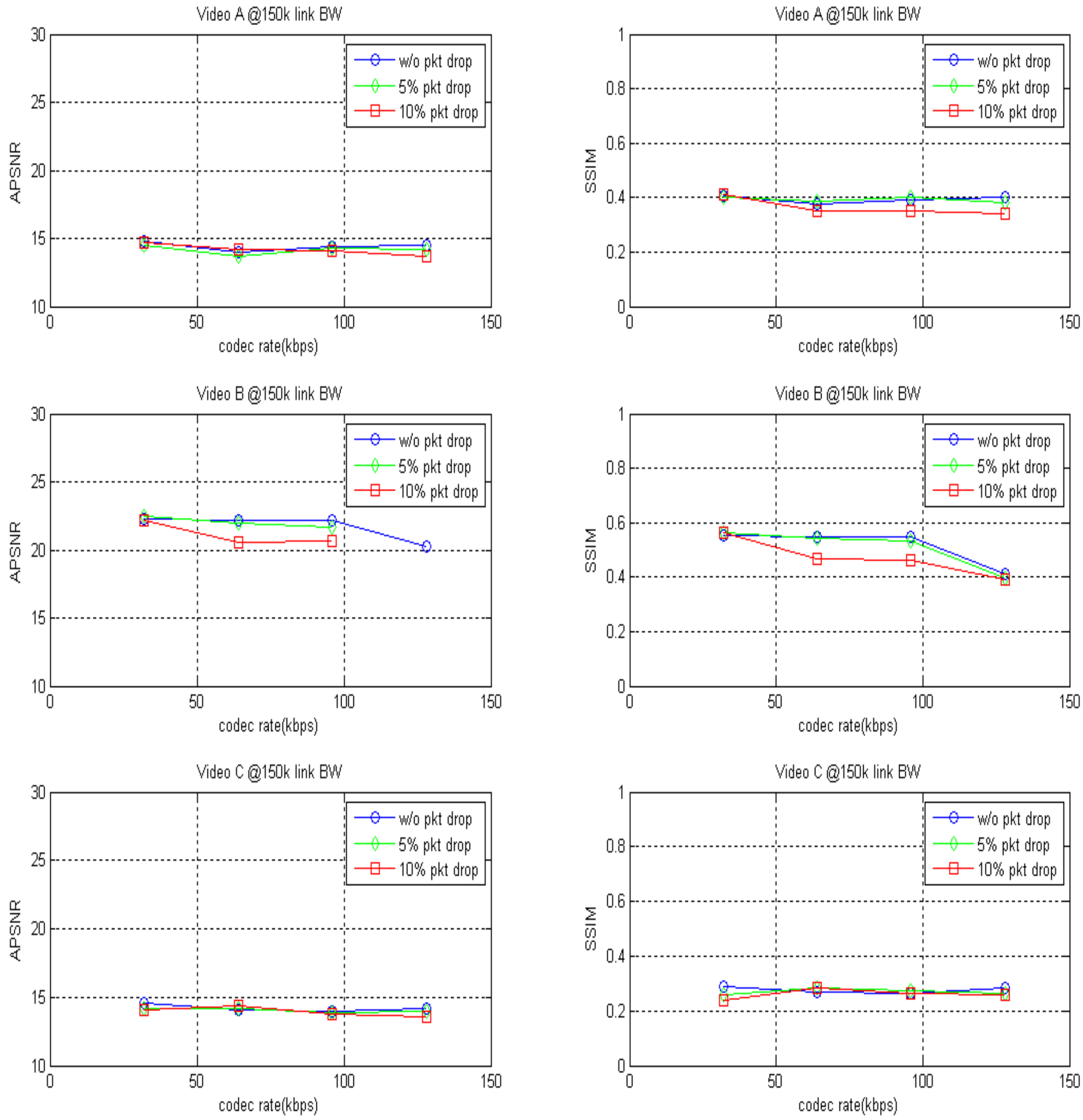


Figure 7-5 Video quality with and without packet loss at 150 kbps link rate

For video B at 128 kbps codec rate and both 5% and 10% packet loss on the link, frame loss is inevitable. It is observed in Figure 7-5 that introducing packet loss on the link results in degraded visual quality. However, the degradation is not discernable as the achievable visual quality on 150 kbps link is already quite low.

Since the achievable video quality even without packet loss at a link rate of 150 kbps is mediocre, the same observations for 300 kbps reveal visual quality degradation more clearly.

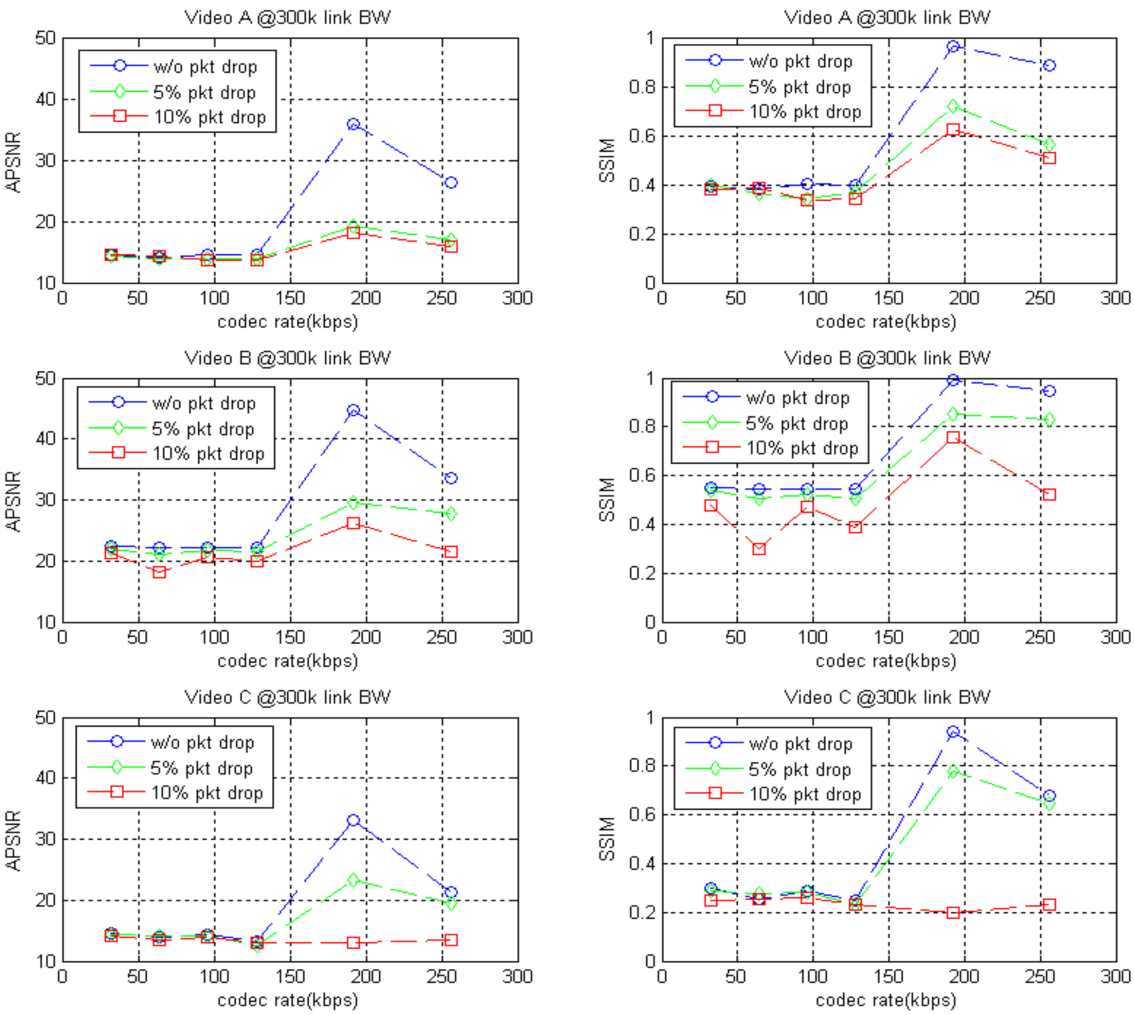


Figure 7-6 Video quality with and without packet loss at 300 kbps link rate

In Figure 7-6, with respect to the packet loss only slight variation in visual quality is observed for lower codec rates. However at higher codec rates, the quality degradation is more pronounced, with increasing packet loss percentage, markedly at 192 kbps codec rate.

At 256 kbps codec rate, there is a noticeable degradation in quality for all videos, with or without loss introduced on the link. This can be explained by the link behaviour close to link bandwidth limit as described previously for the benchmarking tests using MGEN traffic. In this case, the erratic stream behaviour characterized by high variation (as compared to the variation that would be seen at higher link bandwidths) is seen to affect achieved decoded video quality adversely.

7.3 Effect of video content

We have shown that video content does not affect achieved throughput rates on the link for particular codec rates. However, the difference in performance for the different videos indicates a reliance of the performance on the video content. This conclusion can also be inferred from the fact that even though all three videos exhibit best possible performance at 192 kbps codec rate, the particular PSNR or SSIM average value achieved is not the same for all. This is further elaborated in a CDF plot of PSNR and SSIM values for all three videos at 192 kbps codec rate at 300 kbps link bandwidth as shown in Figure 7-7.

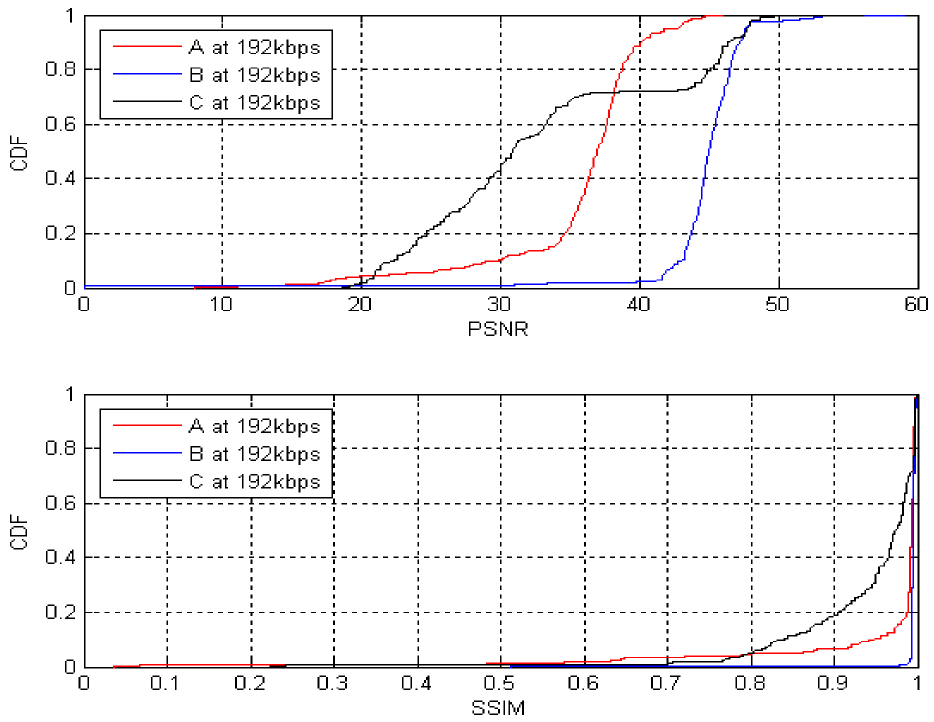


Figure 7-7 Video quality metric distribution at 300 kbps link rate

The curves for the different videos correspond to the video content with video B exhibiting the highest SSIM and PSNR values. Video C is a busy video sequence with frequent scene changes and rapid panning in and out. Video B is the most stable video with almost no scene changes and localised changes within frames with background remaining mostly constant. Video A is similar to video C.

7.4 Inter-packet delay variation analysis

In this section, we take a look at inter-packet delay variation (IPDV) for all three videos in order to find a relationship between inter-packet delay variation and the resulting video quality. The aim in investigating the inter-packet delay variation profiles is to identify trends of behaviour with using increasing codec rates on a particular link as well as the difference in the profiles for particular codec rates on different link bandwidths.

Figure 7-8 plots IPDV histogram profiles for video A at different codec rates on different link bandwidths.

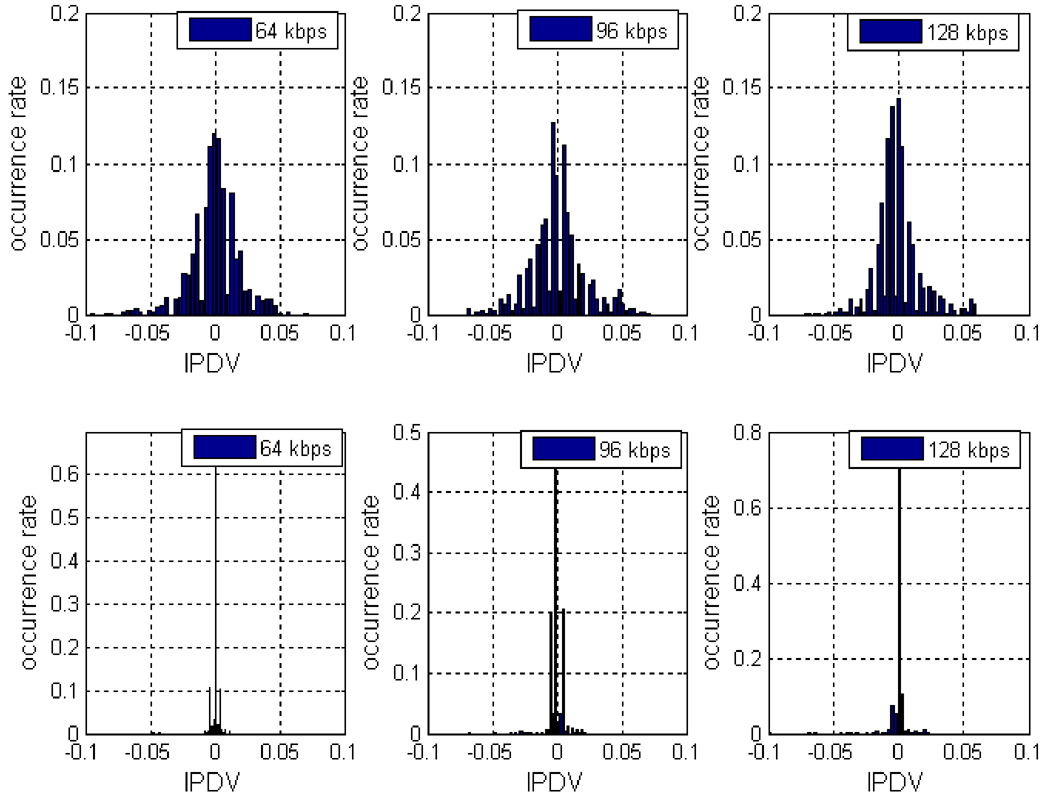


Figure 7-8 Video A IPDV histogram plot

Three things apparent from Figure 7-8 are:

I. For any particular codec rate, the IPDV histogram will be wider (implying a higher delay variation) at lower bandwidth than for the same codec rate (or input rate) at higher bandwidth.

II. Increasing codec rate results in increasing variation, characterised by widening of the histogram. This indicates a greater difference between transmitted and received inter-packet spacing. Video at 150 kbps however, does not depict this trend as well as at 300 kbps link bandwidth, implying the link's inability to cope with/support inter-packet spacing variation associated with VBR traffic at transmitting end, even at low codec rates.

III. The PSNR and SSIM values remain roughly the same at both link bandwidths (Fig.22). This means that the application buffer can accommodate the resulting IPDV. We show later that selecting a smaller application play-out buffer degrades the performance in terms of PSNR and SSIM.

The same trend can be shown for videos B & C at 300 kbps link bandwidth as follows:

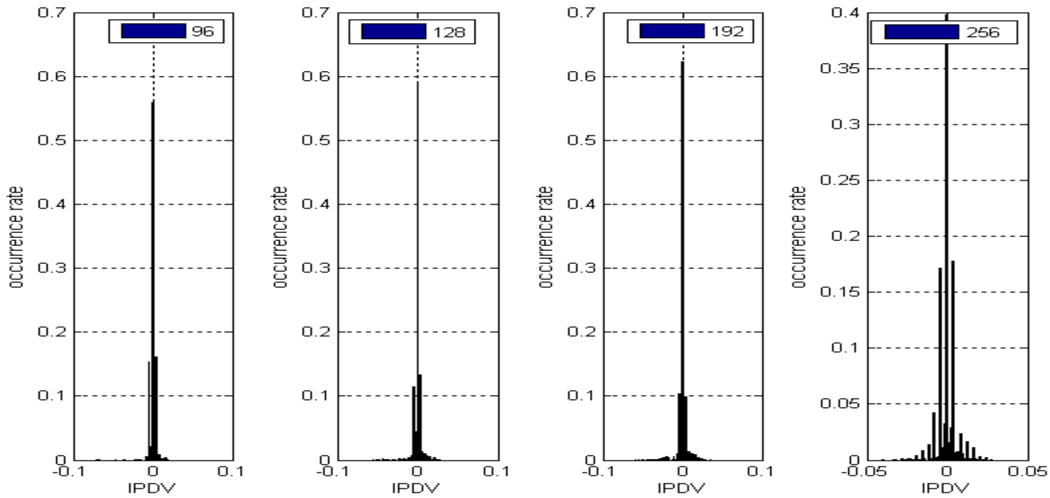


Figure 7-9 Video B IPDV profile at 300kbps link bandwidth

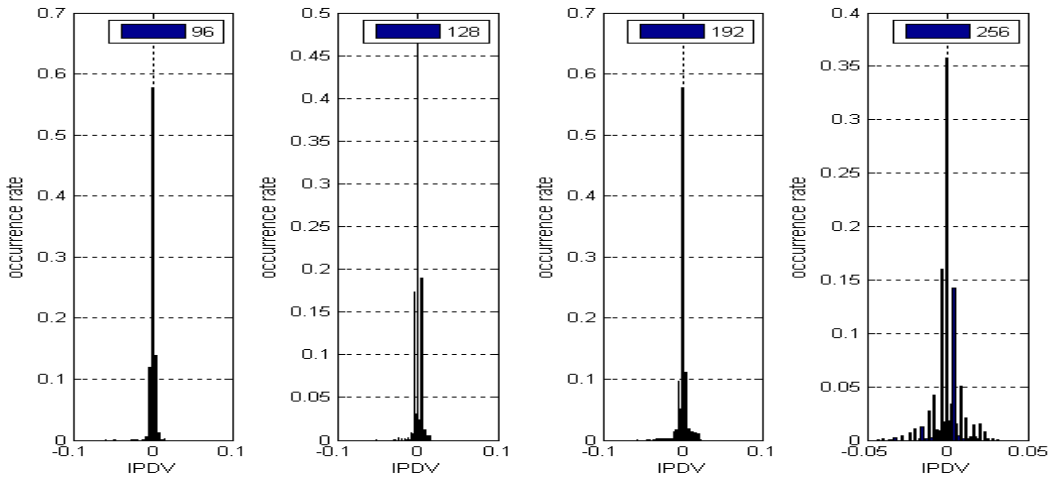


Figure 7-10 Video C IPDV profile at 300kbps link bandwidth

It has been observed for MGEN traffic that at input bitrates close to link bandwidth, the inter-packet delay variation increases. Similar is the case for 256 kbps codec rate on 300 kbps link and 128 kbps codec rate at 150 kbps link. It has also been shown that as codec rate increases, inter-packet delay variation increases. The two factors in combination, cause visual quality impairment at higher codec rates due to decreased tolerance to delay variation at playout buffer. The progressive increase in delay variation with codec rates is shown in Table 7-1 in terms of absolute mean values of IPDV for codec rates transmitted across 300 kbps link rate. The jump in mean value at 256 kbps is indicative of both factors mentioned super-imposing.

Table 7-1 IPDV vs. codec rate

Video B	IPDV abs. Mean	Video C	IPDV abs. mean
96kbps	0.0020 s	96kbps	0.0017 s
128kbps	0.0022 s	128kbps	0.0024 s
192kbps	0.0023 s	192kbps	0.0027 s
256kbps	0.0034 s	256kbps	0.0047 s

A comparison of performance in terms of achieved decoded video quality between 300 kbps link and a higher link rate of 400 kbps reveals the same downward sloping trend at operating bit rates close to link bandwidth limit, as shown in Figure 7-11. This supports our assumptions above.

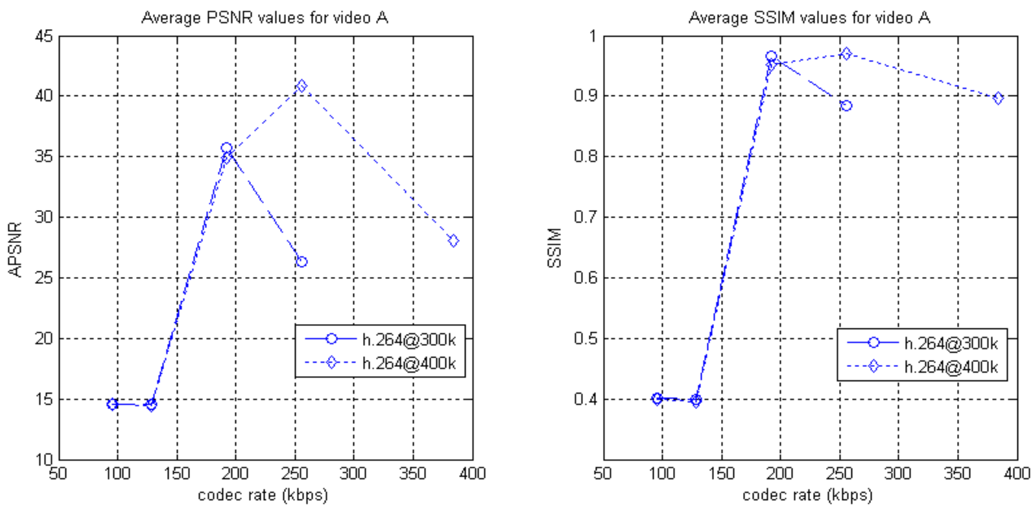


Figure 7-11 300kbps vs. 400kbps link rate performance

7.5 Effect of playout buffer size

The play-out buffer serves to remove the effect of jitter, introduced by delay variation experienced by a flow on the link, by collecting a certain number of frames before presenting on the screen, instead of having to display each frame as it is decoded. Thus, the buffer smoothens the play-back process in case of back-end decoding delays. The amount of frames a buffer can store is determined by its size. The fact that there is no difference in video performance at lower codec rates for both link rates (ref. Fig. 15) implies that the jitter buffer, in our case the VLC play-out buffer, can withstand the delay variation introduced at link layer by the link (this delay variation is assumed to be due to bandwidth constraint only, in the case of no packet loss). This can be shown by changing buffer size and observing affect on resulting received PSNR/SSIM. The default size of VLC play-out buffer is 300ms. When reduced to 100ms, some degradation in video quality is observed, as shown by the CDF plots in Figure 7-12 (line curves), depicting per-frame PSNR and SSIM values at both 300ms and 100ms play-out buffer size at 300kbps link rate, for different codec rates. We mind that the degradation in de-jitter buffer does not cause any frame loss.

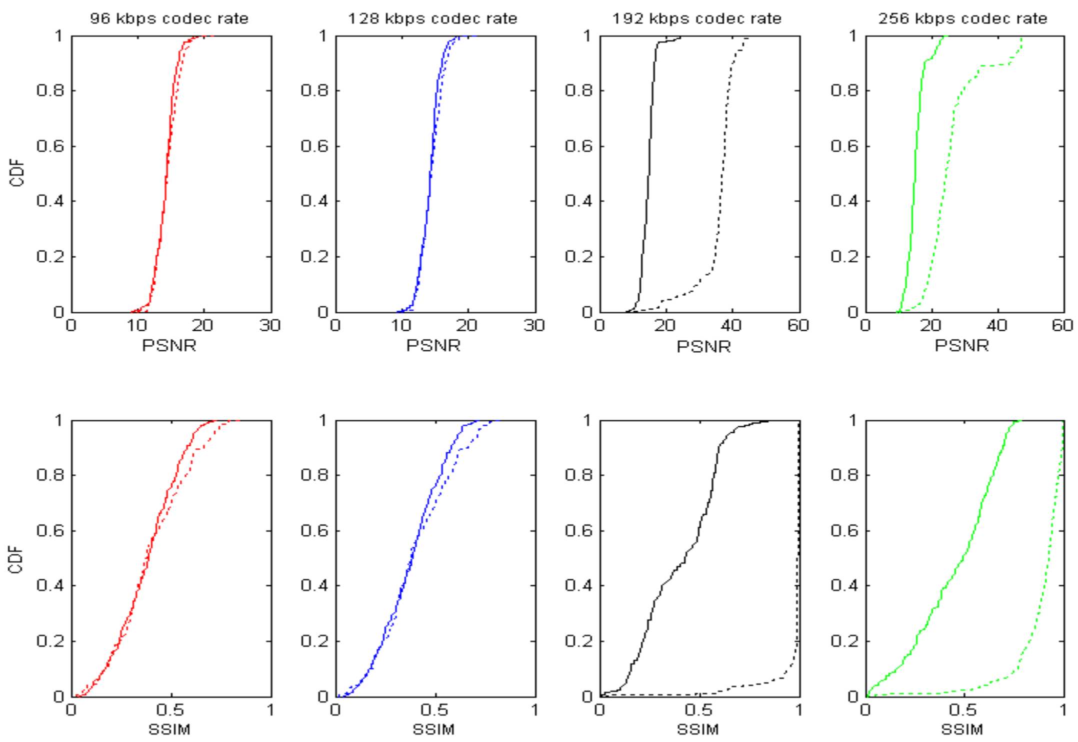


Figure 7-12 Video A PSNR & SSIM CDF plot with different buffer sizes at 300kbps link bandwidth.
Dotted curve: 300ms buffer size, line curve: 100ms buffer size.

A drop in visual quality is observed at codec rates that employs bitrates on the link close to link bandwidth limit (ref. Fig. 26, videos at 256kbps codec rate on 300 kbps link). This can be explained as follows. At lower codec rates, the play-out buffer fills up at lower speed than at higher codec rates. This can allow for more tolerance toward high delay variation at lower codec rates. Lower codec rate means that the packets are not buffered for long time in the FIFO queue before transmission on the link. Therefore the IPDV is lower and thus the play-out buffer is able to empty more smoothly.

However, as codec rate is increased, the tolerance to delay variation reduces. The buffer is filled more quickly and must also be emptied more quickly to keep up with the codec rate at display. This can be seen also from Figure 7-12, where degradation in PSNR/SSIM for lower codec rates is only slight for lower codec rates (despite increasing trend of IPDV with increasing codec rate as shown before) but becomes more prominent for higher codec rates when buffer size is changed. A smaller buffer will be more sensitive to these changes.

7.6 Effect of flow prioritization

In this section, we look at how video traffic prioritisation can affect decoded video quality. For the test scenario, we use MGEN to generate CBR traffic at transmitting end, to compete with VLC generated video stream on the same link (300 kbps bandwidth) with different prioritisations. The QualNet IPNE library allows setting up precedence levels for streams sniffed by IPNE module, identified by unique endpoints.

We concentrate on high video codec rates that resulted in better decoded visual quality; therefore we conduct the prioritisation tests only for 300 kbps link bandwidth. Also, for this test we only use the SSIM metric to determine achieved decoded visual quality. This is because some frame loss is observed for video A encoded at 256 kbps codec rate. Since the SSIM index grades visual quality on an absolute scale from 0 to 1, we insert 0s in place of missing frames equivalent to the number of missing frames in received video to get an estimate of visual quality. Where we have added this constant value for any number of frames, a discontinuity can be observed in the CDF plot.

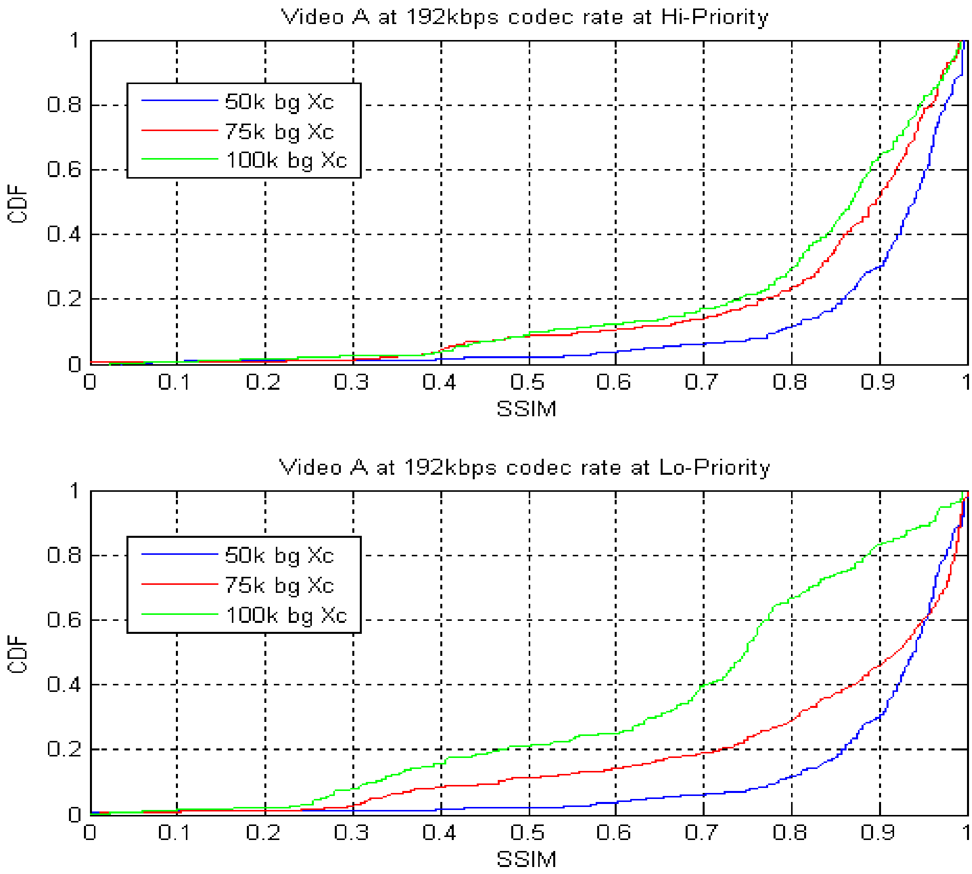


Figure 7-13 Prioritised video encoded at 192 kbps codec rate

The MGEN generates the CBR traffic used as background traffic (referred to as 'bg Xc' in figure labels) at different datarates. The corresponding resulting visual quality at receiving is shown in the figure above. The observations can be two divided into cases;

a) Video stream has higher priority than background traffic

b) Video stream has lower priority than background traffic

It is observed in Figure 7-13 that visual quality degrades with increasing background data traffic volume in terms on bandwidth occupancy. whether it is assigned higher priority or not. Note that the bandwidth occupied by video encoded at 192 kbps does not change for different background traffic data rates. However, when video is assigned higher priority the degradation is more graceful. This is true particularly for the case where background traffic reaches 100 kbps throughput on the link. We do not test for higher background traffic rates as they result in substantial frame loss with correspondingly high loss in visual quality.

Also to be noted is the fact that video encoded at 192 kbps remains unaffected by background traffic upto 50 kbps on the link, as the visual quality at both high and low prioritisation remains unchanged.

Next, we take a look at Video A encoded at 256 kbps codec rate. Figure 7-14 shows that video encoded at 256 kbps codec rate cannot withstand background traffic much more than 50 kbps either at high or low prioritisation. At lower prioritisation, even 50 kbps traffic results in some visual quality degradation.

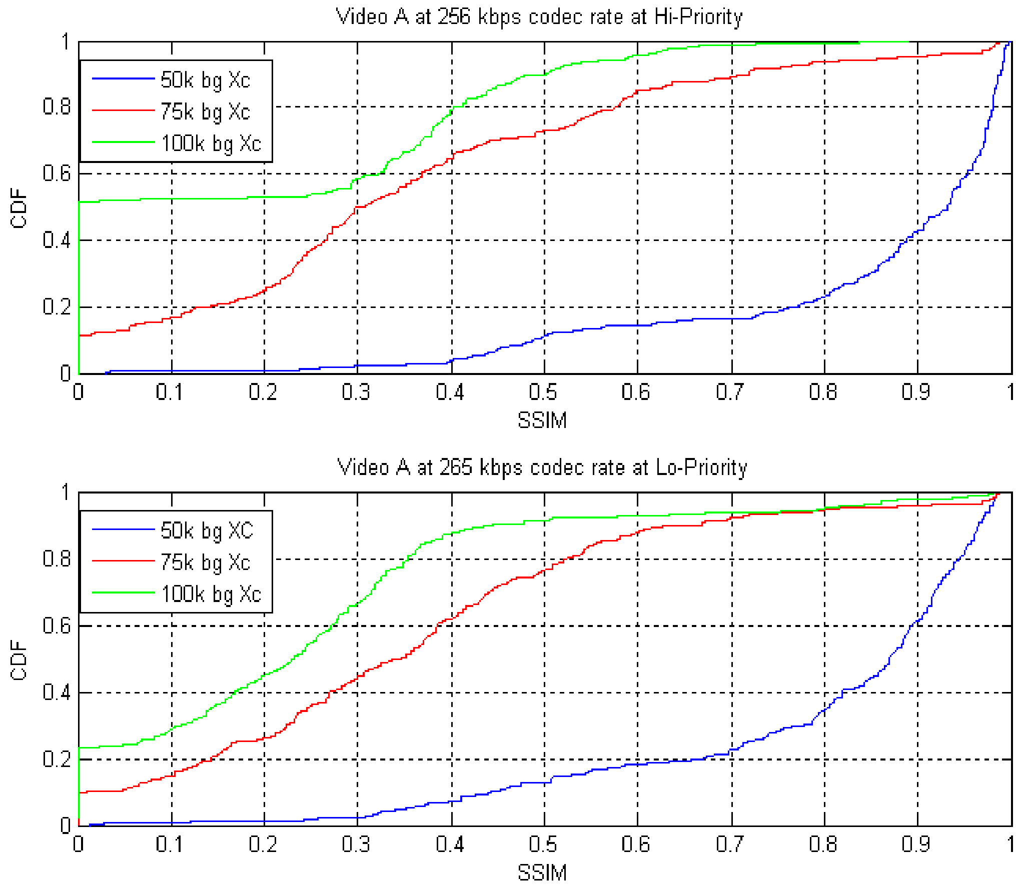


Figure 7-14 Prioritised video encoded at 256 kbps codec rate

In Figure 7-14, we see that frame loss is greater when video has higher priority when background traffic reaches 100 kbps. This is not so significant, as the number of frames lost in any case depends upon which frames were lost. If a key frame is lost, it means that a number of frames depending on the key frame cannot be reconstructed, even by error correction (EC) techniques. A key frame is one from which other frames can be reconstructed, so a key frame generally refers to I or P frame.

Chapter 8 Summary of Results

In this thesis, an investigation of H.264/AVC encoded video streams' performance over an emulated TEDS link was conducted, with the aim to demonstrate the feasibility of video streaming services over various TEDS link conditions. To conclude this study, following is a summary of the significant results:

- PSNR and SSIM are found to correspond to each other, with SSIM giving a finer picture of visual quality performance variation from case-to-case.
- The highest usable codec rate on a link depends on the link bandwidth. Codec rates higher than link bandwidth are unworkable (ref. Figure 7-2).
    - Visual quality is seen to improve at least in subjective terms as codec rates are increased (ref. Figure 7-3).
    - A high visual quality performance in terms of objective metrics requires use of high link bandwidth. As shown, a good quality reproduction is first achieved when using video encoded at 192 kbps codec rate on 300 kbps link. (ref. Figure 7-2).
- Video streaming is still possible under packet loss of 5% on the link, with some degradation (ref. Figure 7-6).
    - At 10% packet loss, the amount of visual quality degradation is dependent on video content and whether EC techniques are able to cope with the resulting frame loss. For less demanding videos, successful video streaming, albeit degraded, may still be possible (ref. Figure 7-6).
    - Packet loss on the link is seen to have a more pronounced impact on videos encoded at high codec rates of 128 kbps onwards (ref. Figure 7-6).
- Video content is seen to affect video streaming performance in terms of achievable visual quality, while employed bitrates on the link for particular codec rates are similar. Less demanding videos, which have more redundancy

between frames, result in better performance in terms of visual quality (ref. Figure 7-7, Figure 7-1).

- For videos encoded at a particular codec rate, the delay variation introduced by the link is greater at lower link bandwidths. Also, results seem to indicate an increasing delay variation trend for higher codec rates on a particular link bandwidth (ref. Figure 7-8, 7-9, 7-10).
- Videos encoded at high codec rates close to the link bandwidth are prone to degradation, as the overwhelmed link cannot cater for the increased delay variation (ref. Figure 7-11 & Table 7-1).
- Video play-out buffer size at receiving side is also observed to have an impact on visual quality performance (ref. Figure 7-12). It should be enough to cater for delay variation on the link.
- In competing traffic scenario, video streaming is unaffected as long as background traffic does not exceed bandwidth leftover; both in the case where video stream is prioritised and when it is not (ref. Figure 7-13, 7-14).
    - In case background traffic exceeds leftover bandwidth, prioritisation of video stream helps to prevent visual quality degradation, but only up to a certain level. Once the background traffic's bandwidth occupancy exceeds this level (dependent on codec rate employed; 100 kbps for video encoded at 192 kbps), visual quality degradation is inevitable (ref. Figure 7-13, 7-14).

Conclusion & Future Work

The results of this thesis work, as summarised above, demonstrate the feasibility of video streaming over wideband TEDS link using the H.264/AVC codec. We have shown that adequate levels of visual quality can be expected in such streaming scenario, despite bandwidth constraints. Also, we have determined video streaming parameters such as employed codec rate in relation to available link bandwidth and play-out buffer size in relation to delay variation introduced by bandwidth constrained link, as some of the influential factors affecting achieved decoded visual quality.

However, user requirement studies are still required to define an acceptable level of video quality, or in other words, a tolerable level of visual degradation in terms of objective visual quality metrics, for different usage scenarios such as real-time tele-medicine, fire incidents, on-site personnel/equipment monitoring, etc. It is important to have objective quality metrics as these translate well to system dimensioning parameters. This information will enable fine tuning of the video streaming system parameters identified in this thesis for an efficient TETRA/TEDS system provisioning for different applications.

Initial TEDS systems deployments are to make use of a 50kHz bandwidth due to available radio spectrum constraints. There is currently an ongoing debate as to whether TETRA systems should be allocated spectrum leftover from the analog services (TV, cellular & military communication) phase-out in the European countries in particular, instead of being auctioned off to private operators. Among other benefits, such a measure has important security implications, as it would create greater uniformity among operators in different countries in terms of used frequencies for TETRA, allowing for smoother inter-operability should the need arise. In terms of functionality, it is clear that a good video streaming performance requires a move to higher bandwidth TEDS deployment. Therefore, this study also contributes to the argument to allocate a greater share of the scarce spectrum resources to PSS, to harness the full potential of TEDS.

In the literature review, we pointed out that very little work has been done regarding transmission of video over wideband systems like TEDS; therefore, there is a lot of room for further work. Future work can include implementing a more accurate model of the link loss to better simulate radio channel conditions, such as a bursty loss model. This would give a clearer picture of the robustness of the H.264 codec for video transmission over TEDS. Moreover, different codec profiles with or without error resilience mechanisms in use can be tested to see which can best cope with lossy streaming over wideband link.

In the same vein, a similar performance evaluation can also be carried out for the newer scalable version of the H.264 codec (H.264/SVC) in order to identify the feasibility and limitations of its use for video over TEDS. The H.264/SVC is intended for use in video broadcasting to multiple end-points of different configurations and capabilities, from one source. It would be interesting to see if and how the technology translates to a constrained bandwidth environment.

The effectiveness of the prioritisation mechanism for video in the presence of other types of traffic can also be further investigated. It would be useful to see how concurrent activities such as web browsing (HTTP/TCP based traffic) affect video streaming, and if using prioritisation mechanisms as defined in TETRA/TEDS standards can help mitigate the problems.

Appendix A

QUALNET IPNE (IP Network Emulator)

In an IPNE scenario, the QualNet simulator takes real network packets as input to the simulated scenario and then outputs the resulting packets back into the real network. In this case, QualNet is used to simulate a 100kbps link over which we stream real, live video from a webcam using VLC media player at both end points.

The setup of this scenario requires an .ipne file in addition to .config and .nodes files, which defines IPNE mode used and other parameters. For this scenario, two virtual nodes are created in .config file representing the two real-world ‘operational nodes’, one of which acts as transmitter and the other receiver of video stream. The link between them is defined as a 100 kbps wired interface. No application protocol is defined in .config file. The scenario is as depicted below.

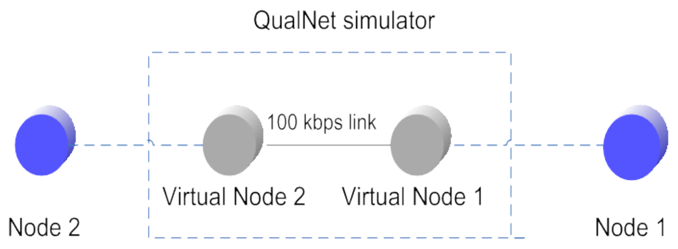


Figure A-1 QualNet IP Network Emulator Model

The IPNE modes used are ‘NatNo’ and ‘TrueEmulation’. Both require that all nodes have IP addresses in the same subnet. The machine running QualNet also has to have an IP address on the same subnet. The virtual nodes are assigned the same IPs as their real-world counterparts. In addition, a packet sniffing device which is the interface that will sniff for packets on the real network and inject these into the simulated network (and vice versa) is to be specified. All these parameters are specified in the .ipne file.

Physical connectivity arrangements are as follows:

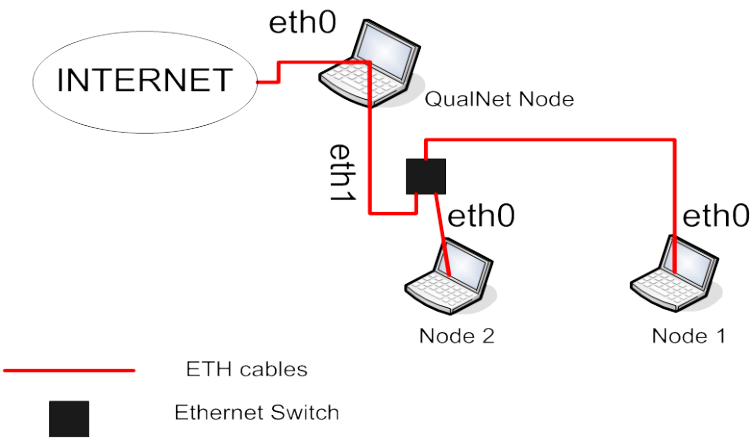


Figure A-2 Physical connectivity diagram

Node 1, 2 and the QualNet machine are connected via simple Ethernet cables through a switch/hub.

The QualNet machine also requires Internet connectivity to enable license checkout for running the simulation. For this purpose, we introduce an additional Ethernet port via USB.

The following manual system configurations are required on the operational nodes:

Node 1: route add –host 192.168.0.2 gw 192.168.0.3

Node 2: route add –host 192.168.0.1 gw 192.168.0.3

The following table shows the network settings at each node/interface.

Table A-1: IPNE network settings

	Node 1	**QualNet Node**	**Node 2**
Wired interface	eth0	eth0 eth1	eth0
Wired address	eth0:192.168.0.1	eth0:130.233.158.194 eth1:192.168.0.3	eth0:192.168.0.2
Subnet mask	eth0:255.255.255.0	eth0:255.255.255.0 eth1:255.255.255.0	eth0:255.255.255.0
Default gateway	eth0:192.168.0.3	eth0:130.233.158.254 eth1:130.233.158.194	eth0:192.168.0.3
DNS address		130.233.160.131 130.233.160.132	

NOTE: Packet sniffing device is specified as eth1 on the QualNet machine in the .ipne file, in this case.

Running the Simulation

The command to run the qualnet simulation from the CLI should be given in super-user mode i.e. ‘sudo ./qualnet ipne.config’.

This starts the configuration running and stats are printed on-the-fly as simulation progresses so packet input/forwarding/output can be monitored.

Start by pinging from node 1 to 2 and vice versa to check for connectivity via the QualNet node. The stats printing on screen provide confirmation of network traffic transfer. Next, run the VLC player on Node 2, our transmitter for this scenario, to stream video from the attached webcam over UDP to Node 1. Setup the VLC media player to receive the video stream on Node 1 making sure port numbers at both ends are the same.

VLC Settings

- Go to File and choose video file to be streamed.
- Next, under Network tab, check udp/rtp, set port to 1234 and check Stream/Save option.
- Go to Stream settings and check ‘play locally’ as well as ‘udp’ and enter the IP address of the machine to be streamed to (in this case, Node 1’s IP).
- It is possible to change video codec and bitrate settings here as well.
- At Node 1, simply go to File->Open Network Stream and check udp/rtp and corresponding port number which should be the same as that on Node 2.
- The stream received can be saved by checking stream/save option at receiver and specifying file name and format to save in (.mp4 for H.264 encoded video) before start of streaming.

Of the two IPNE mode options available for emulation of our scenario, i.e. NatNo or TrueEmulation, comparison of emulation performance results with simulation results showed that TrueEmulation mode was better able to closely model the link layer as desired. In TrueEmulation, the incoming packets are treated only at link layer level (i.e. adding 28 byte header required for point-to-point transmission only). This contrasted with the case in NatNo where the link received incoming packets at network layer and appeared to add its own IP headers on top of existing ones which lead to an increase in observed delay on the link. This in turn affected link behaviour, causing it to deviate from expected results based on simulations.

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