Performance Evaluation of MPEG-4 Video Streaming over UMTS Networks using an Integrated Tool Environment

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ABSTRACT

Universal Mobile Telecommunications System (UMTS) is a third-generation mobile communications system that supports wireless wideband multimedia applications. This paper investigates the video quality attained in streaming MPEG-4 video over UMTS networks using an integrated tool environment, which comprises an MPEG-4 encoder/decoder, a network simulator and video quality evaluation tools. The benefit of such an integrated tool environment is that it allows the evaluation of real video sources compressed using an MPEG-4 encoder. Simulation results show that UMTS Radio Link Control (RLC) outperforms the unacknowledged mode. The latter mode provides timely delivery but no error recovery. The acknowledged mode can deliver excellent perceived video quality for RLC block error rates up to 30% utilizing a playback buffer at the streaming client. Based on the analysis of the performance results, a self-adaptive RLC acknowledged mode protocol is proposed.

1. INTRODUCTION

Universal Mobile Telecommunications System (UMTS) [1]-[2] is a third-generation (3G) mobile communication system where the radio interface is based on code division multiple access, known as Wideband Code Division Multiple Access (WCDMA). To date, UMTS networks are in commercial service and many other UMTS networks around the world are in either pre-commercial or deployment phase. UMTS is among the first third-generation mobile system to offer wireless wideband multimedia communications over the Internet protocol [3]. As such, it allows mobile Internet users to access a variety of multimedia contents available on the Internet in a seamless fashion (i.e., always on) at data rates up to 384 kb/s in wide-area coverage and high user mobility.

Video streaming is a promising multimedia application which is recently gaining popularity and maybe a key factor to the success of 3G systems. Using a mobile device, users access online video clips (such as news, sports, etc.) by clicking on a hyperlink using their web browser, which results in the browser opening a video player to play the selected clip. With streaming, the video content need not be completely downloaded, but the client can begin playback the video a few seconds after it begins receiving parts of the content from the streaming server. Since raw video requires high transmission bit rates, video compression is usually employed to achieve transmission efficiency. MPEG-4 [4] is a video compression standard adopted by most mobile networks including UMTS. The compression technology of MPEG-4 produces good quality video at bit rates that are suitable for mobile and wireless transmission.

The paper evaluates the video quality achieved in streaming MPEG-4 coded video over UMTS networks. We employ an integrated tool environment for simulation and evaluation of the received video quality.

2. UNIVERSAL MOBILE TELECOMMUNICATIONS SYSTEM (UMTS)

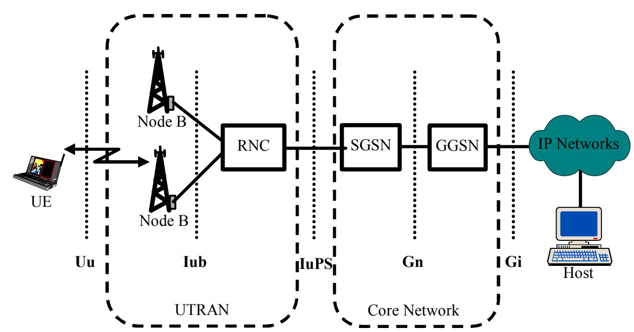

Figure 1. UMTS Reference Architecture

Fig. 1 shows a simplified architecture of UMTS for packet-switched operation [2][5], which consists of one or several User Equipments (UEs), the UMTS Terrestrial Radio Access Network (UTRAN) and the core network. The UTRAN is composed of Node Bs connected to a Radio Network Controller (RNC). The core network, which is the backbone of UMTS, comprises the Serving GPRS Support Node (SGSN) and the Gateway GPRS Support Node (GGSN). The SGSNs route packets to and from UTRAN, while GGSNs interface with external IP networks. UE, which is a mobile station, is connected to Node B over the UMTS radio interface.

Fig. 2 depicts the UMTS protocol architecture for the transmission of user data which is generated by IP-based applications. The applications as well as the Internet protocol suite are located at the end-nodes, namely, the UE and a host.

The Packet Data Convergence Protocol (PDCP) provides header compression functionality. The Radio Link Control (RLC) layer can operate in three different modes: acknowledged, unacknowledged and transparent. The acknowledged mode provides reliable data transfer over the error-prone radio interface. Both the unacknowledged and transparent modes do not guarantee data delivery. The transparent mode is targeted for the UMTS circuit-switched mode in which data are passed through the RLC unchanged. The Medium Access Control (MAC) layer can operate in either dedicated or common mode. In the dedicated mode, dedicated physical channels are allocated and used exclusively by one user (or UE), whereas in the common mode, users share common physical channels for transmitting and receiving data. The Physical (PHY) layer contains, besides all radio frequency functionality, spreading, and the signal processing including, power control, forward error-correction and interleaving.

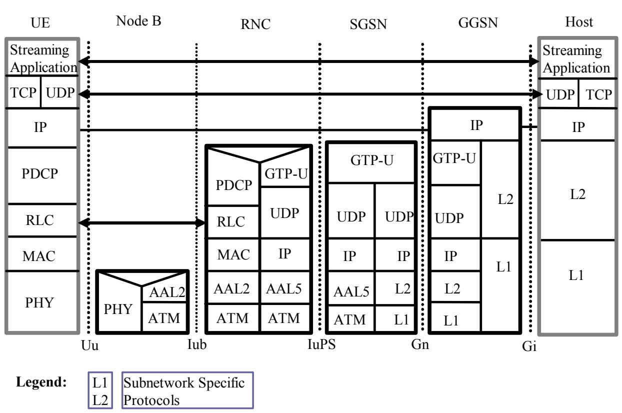

Figure 2. UMTS Protocol Architecture – U-Plane

3. MPEG-4

MPEG-4 [4], which is standardized by ISO/IEC, defines a set of tools to create, represent and distribute individual audiovisual objects, both natural and synthetic, ranging from arbitrarily shaped objects to sprites, face and body animations.

An MPEG-4 encoder generates three types of frames. Intra-frames (I-frames) contain information from encoding a still image. Predictive frames (P-frames) are encoded from the previous I-frames or P-frames. Bi-directional frames (B-frames) are encoded bi-directionally from the preceding and the following I- and P-frames. B-frames achieve the highest compression, which require the lowest transmission bandwidth, but they also take the longest time to encode. Conversely, I-frames have the lowest compression of the three frame types, but they can be encoded and decoded faster than the other frame types.

4. INTEGRATED TOOL ENVIRONMENT

The integrated tool environment consists of an *MPEG-4 encoder*, *video sender*, *Network Simulator* (*ns-2)* [6], *MPEG-4 decoder*, *evaluate trace* program, *PSNR* program and *MOS* program. Fig. 3 depicts the relationship between the different components, input video files and generated output files of the integrated tool environment. The integrated environment methodology was proposed and developed within the framework of EvalVid [7], and [8] extended the environment to include *ns-2*.

A video clip, which is usually stored in YUV format, is fed to an MPEG-4 encoder which in turn generates an encoded video stream. An open-source MPEG-4 encoder and decoder was used, i.e., *ffmpeg* [9]. The encoded video stream is read by *Video Sender* (VS) to generate a *trace* file, which contains information such as frame type, size, etc. for each video frame. The trace file is then fed to the streaming server in the *ns-2* simulator to produce video streams over UMTS. The effect of streaming video over UMTS is captured in a streaming client *log* file which is generated by *ns-2*. The log file contains information such as timestamp, size and identity of each packet. Note that, *ns-2* also generates a similar log file for the streaming server. The trace file and the log files are used by the *Evaluate Trace* (ET) program to generate possibly corrupted video files as a result of transmission over UMTS. The corrupted video file is needed by *Peak Signal to Noise Ratio* (PSNR) and *Mean Opinion Score* (MOS) programs to evaluate end-to-end video quality.

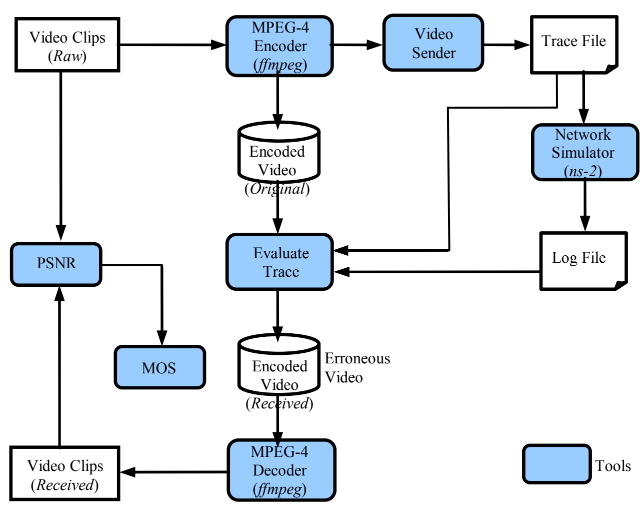

Figure 3. Structure of the Integrated Tool Environment

4.1 Network Topology

The network topology modeled in *ns-2* is illustrated in Fig. 4, which consists of a streaming client and a streaming server. In the simulation, a UE plays the role of a streaming client and a fixed host is the streaming server located in the Internet. Several extensions were made to the simulator for modeling UMTS [10]. The extensions for video traffic generation based on MPEG-4 traces are provided by [8]. With the UMTS extensions, instances of UMTS nodes, viz., UE, Node B and RNC can be instantiated.

Video streaming uses a family of transport protocols, namely, User Datagram Protocol (UDP), Real-time Transport Protocol (RTP) and Real-Time Control Protocol (RTCP). UDP is a lower-layer transport protocol while RTP and RTCP are upper-layer transport protocols. UDP is employed because it provides timely delivery of packets. However, UDP does not guarantee packet delivery. RTP runs on top of UDP, which packetizes and provides in-order delivery of video frames. RTCP is used by the video client to inform video server concerning the received video quality. In the simulation, we assume no interaction between the video client and server. Hence, RTCP is not modeled. Similarly, RTP is also not modeled, but those functions that are needed such packetization, packet sequence numbering and in-order delivery are supported by different tools in the integrated environment. For instance, packetization is implemented in VS.

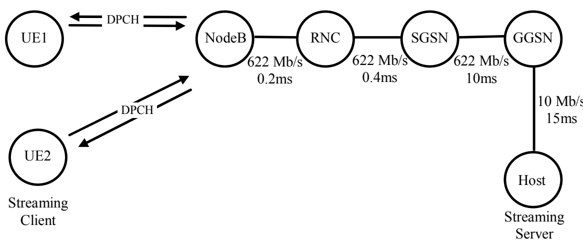

Figure 4. Network Topology

Since the primary aim of the simulation was to investigate the impact of the radio interface on the quality of MPEG-4 video streaming, we assume that no packet losses, errors or congestion occur on either the Internet or the UMTS core network. The transfer delay introduced by the Internet and the UMTS core network is constant throughout the entire video streaming duration. Hence, the quality of the streaming video is solely attributed to the radio interface. The links between two nodes are labeled with their bit rate (in bits per second) and delay (in seconds). Each link capacity was chosen so that the radio channel is the connection bottleneck. Hence, the quality of the streaming video is solely attributed to the UMTS radio interface. Consequently, the functionality of SGSN and GGSN was abstracted out and modeled as traditional *ns* nodes since generally they are wired nodes and, in many ways, mimic the behavior of IP router. Currently, no header compression technique is supported in the PDCP layer.

4.1.1 UMTS RLC Model

The RLC model supports both *acknowledged* and *unacknowledged* operation modes. The unacknowledged mode (UM) provides unreliable but timely delivery of RLC blocks. That means, no error recovery is performed even though erroneous RLC blocks are detected. An RLC block consists of a header and a payload which carries higher layer data. An erroneous block is discarded by RLC, which would result in an incomplete IP packet. An incomplete IP packet is discarded by RLC and not delivered to higher layers as soon as one of the RLC blocks composing the IP packet is erroneous.

On the other hand, the acknowledged mode (AM) guarantees delivery by retransmitting erroneous RLC blocks at the expense of transfer delay. The retransmission strategy adopted by the acknowledged mode is the Selective-Repeat ARQ (Automatic Repeat reQuest) scheme. In the simulation, the ARQ was configured to operate in a persistent manner. With Selective-Repeat ARQ, the only RLC blocks retransmitted are those that receive a negative acknowledgement. A status message is used by the receiver for notifying loss or corruption of an RLC block. The status message is in bitmap format. That is, bit_j indicates whether the jth RLC block has been correctly received or not. The frequency of sending status messages is not specified in the standard [11]. However, several mechanisms are defined, which can trigger a status message. Either the sender or the receiver can trigger the status message. Table 1 and Table 2 list the triggering mechanisms for sender and receiver, respectively. It is important to note that not all the triggering mechanisms are needed for the Selective-Repeat ARQ to operate. However, a combination of triggering mechanisms, which deliver optimum performance, is sought.

Table 1. Sender-Initiated Mechanisms

Trigger	Explanation
Last Block in buffer or retransmission buffer	status report is requested by enabling the poll flag in the RLC header
Every *m* blocks	poll flag is enabled for every *m* blocks
Every *n* service data units	poll flag is enabled for every *n* service data units
Utilization of Send Window	poll flag is enabled when the Send Window is *x*% full
Periodic Poll	poll is generated periodically based on a timer

Table 2. Receiver-Initiated Mechanisms

Trigger	Explanation
Detection of missing blocks	Status message is generated once a gap is detected in the RLC sequence number
Periodic status	Status message is sent periodically based on a timer
Estimated block Counter	Status message is generated if not all the retransmitted blocks are received within an estimated period

The advantage of receiver-initiated mechanisms is that the receiver has direct information about missing blocks. For the sender-initiated mechanisms, the sender has first to request a status message by enabling the POLL flag in the RLC header and wait for a reply, which has longer turn around time. Therefore, receiver-initiated mechanisms are preferred. Nevertheless, sender-initiated mechanisms are required to prevent deadlocks and stall conditions. Periodic mechanisms might be more robust compared to others but may result in too frequent status message. In addition, a timer is required at the sender and receiver for proper operation of the triggering mechanisms. At the sender, the timer is called *poll* timer, which is started when a request for status messages is sent to the receiver. If the status message from the receiver does not arrive before the timer expires, the sender repeats the same procedure again. The receiver is equipped with a timer called *status prohibit* timer, which controls the time interval between status messages if triggered consecutively. If the interval is too short, then bandwidth is wasted. On the other hand, if the interval is too long, bandwidth is preserved, but delay increases. The selected triggering mechanisms for the RLC model are labeled by the rows in dark grey.

4.1.2 UMTS MAC Model

The MAC model implemented the dedicated mode. It requests a number of blocks buffered at the RLC layer, which are ready for transmission, and submits to the PHY layer as transport blocks. In this case, each transport block corresponds to an RLC block since no MAC header is required in the dedicated mode as depicted in Fig. 5. The frequency at which the PHY layer can accept transport blocks from MAC is defined by the Transmission Time Interval (TTI). In the UMTS standard, the values of TTI are 10 ms, 20 ms, 40 ms and 80 ms.

4.1.3 UMTS PHY Model

The PHY model is responsible for transmitting transport blocks over the physical channels. For the MAC dedicated mode, the transport blocks are sent over the Dedicated Physical Channel (DPCH) which maintains a fixed bit rate for the duration of the connection. The channel bit rates and the TTI associated with the channel considered in the simulation are shown in Table 3. Note that the bit rates exclude the RLC headers. The other PHY layer functionality is not implemented in the model. Since the PHY layer passes the transport block to the MAC layer together with the error indication from the Cyclic Redundancy Check (CRC), the output of the PHY layer can be characterized by the overall probability of transport block error – referred to as Transport Block Error Rate (TBLER) in this paper. Thus, an error model based on uniform distribution of transport block errors, was used in the simulation. It is valid to assume that the erroneous transport blocks perceived by the RLC is independent and uniformly distributed as a result of interleaving, forward error-correction and fast power control mechanisms provided by the PHY layer [12]. The TBLER, in the range from 0 to 50%, was considered in the simulation.

4.2 Video Transmission

The transmission of video over the UMTS network is illustrated in Fig. 5. Each encoded video frame is carried by UDP over the IP protocol. If the video frame size is larger than the UDP segment size of 520 bytes then it is fragmented. The IP packets are encapsulated into PDCP packets. The RLC entity receives a PDCP packet which comprises an IP packet of 548 bytes. This PDCP packet is segmented into multiple RLC blocks of fixed size. Each of these blocks fits into a transport block in which a CRC is attached at the PHY layer. In the simulation, the RLC header for the acknowledged and unacknowledged modes is 2 bytes and 1 byte, respectively, while the payload size is 40 bytes for both modes. For this RLC payload size and, for instance, a channel bit rate of 384 kb/s, twelve transport blocks are transmitted within one TTI of 10 ms. The simulation parameters are summarized in Table 3.

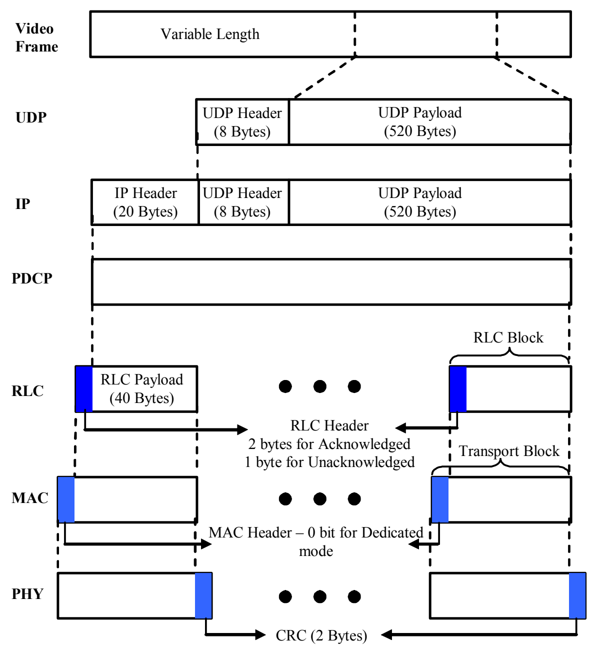

Figure 5. Video Transmission over UMTS

Table 3. Simulation Parameters

App	MPEG-4				
UDP	Maximum Segment Size (Bytes)	520			
	UDP Header Size (Bytes)	8			
IP	IP Header Size (Bytes)	20			
	IP Packet Loss Rate in the Internet	0%			
PDCP	UDP/IP Header compression	No			
RLC	RLC Mode	Acknowledged		Unacknowledged	
	Window Size (Blocks)	4096		Not Applicable	
	Payload Size (Bits)	[320, 640]		[320, 640]	
	RLC Header (Bits)	16		8	
MAC	MAC Header (Bits)	0			
	MAC Multiplexing	Not required for DPCH			
PHY	Physical Channel Type	DPCH			
		Uplink		Downlink	
		Bit Rate (kb/s)	TTI (ms)	Bit Rate (kb/s)	TTI (ms)
		64	20	384	10
	Transport Block Size (Bits)	336			
	Transport BLER	0 – 50%			
	Error Model	Uniform Distribution			

5. PERFORMANCE RESULTS

In order to assess the impact of UMTS radio interface on the video quality, three performance metrics are used, namely *Mean Opinion Score* (MOS), *cumulative inter-frame jitter,* and *video frame error rate*. MOS is the human impression of the video quality, which is given on a scale from 5 to 1 [13]. Scale 5 corresponds to the best quality and scale 1 is the worst. To obtain MOS, the PSNR value for a given video is determined using the PSNR program, which in turn maps to a corresponding MOS scale as shown in Table 4. The MOS program is used to perform the mapping. As shown in Fig. 3, both the MOS and PSNR programs are part of the integrated environment. The PSNR computes the maximum possible signal energy to noise energy, which is mathematically equivalent to the root mean squared error [14].

Table 4. ITU-R Quality and Impairment Scale and PSNR to MOS Mapping [7]

PSNR (dB)	MOS	Perceived Quality	Impairment
> 37	5	Excellent	Imperceptible
31 - 37	4	Good	Perceptible, but not annoying
25 - 30	3	Fair	Slightly annoying
20 - 24	2	Poor	Annoying
< 20	1	Bad	Very annoying

The cumulative inter-frame jitter is defined as the amount of playback delay that must be provided in order to avoid discarding of any video frames in the stream by the client [15]. In the rest of the paper, the cumulative inter-frame jitter is simply referred to as cumulative jitter. The video frame error is defined as the number of video frames with missing data parts or lost divided by the total number of frames transmitted by the server.

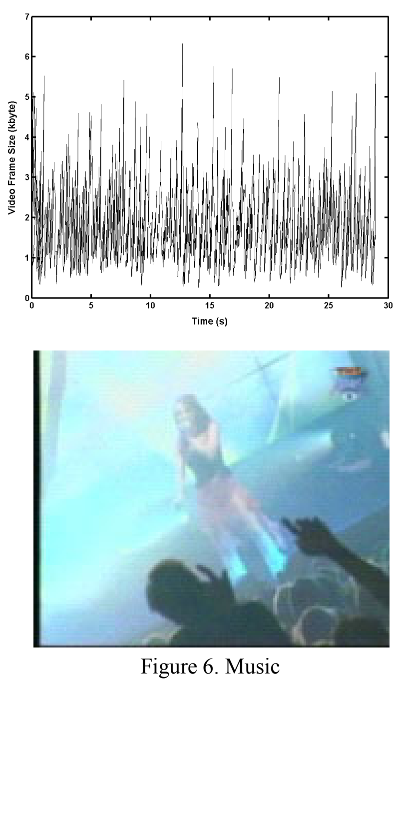

Figure 6. Music

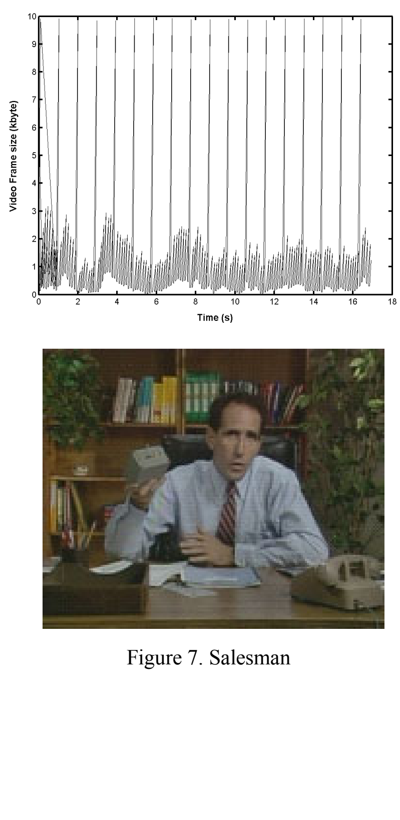

Figure 7. Salesman

Two video clips with distinct characteristics were selected for the simulations. The video clips are depicted in Figs. 6 and 7, which are called “Music” and “Salesman”, respectively. Both clips are encoded at 25 frames/s. “Music” and “Salesman” comprise 750 frames and 450 frames, respectively. The frame size of both clips is *176 × 144* pixels, which is also known as the Quarter Common Intermediate Format (QCIF). The QCIF is used because it is a typical format for streaming video over mobile networks. The encoded frame sizes for each video clip are depicted by the graphs on top of Figs 6 and 7. The peaks in both graphs correspond to the I-frames which are larger than P- and B-frames. Furthermore, the P- and B-frames of “Salesman” are smaller because it has lower temporal information as compared with “Music”.

5.1 Sensitivity to Radio Channel Losses and Frame Sizes

Fig. 8 plots the average MOS for the two video clips received at the client as a function of TBLER for RLC AM and UM modes. Under ideal radio channel, both RLC modes obtained the maximum average MOS, which indicates the best perceived video quality.

As TBLER increases, the average MOS for AM is superior to UM for both video clips. In the case of UM, the perceived video quality is rapidly deteriorating as TBLER increases. As mentioned, RLC UM does not recover erroneous RLC blocks. Instead, the erroneous RLC are discarded including the entire IP packets. As a result, the decoder at the streaming client has to decode video frames with missing data (IP packets). The decoded video quality is very sensitive to missing data. The poor video quality achieved by UM is clearly evidenced when the TBLER is mapped to video frame error rate as shown in Fig. 9. For TBLER as low as 5%, the video frame error rate is approximately 55% and 80% for “Salesman” and “Music”, respectively. Such a high video frame error rate can cause any video decoder to fail. For a given TBLER, the video frame error rate of “Music” is higher because of its encoded video frame size (in particular the P- and B-frames) is larger as observed in Fig. 7.

5.2 Effect of Cumulative Jitter

The perceived video quality for AM drops below the “fair” level only when the TBLER exceeded 15% and 35% for “Music” and “Salesman”, respectively. Unlike UM, video quality degradation can only be caused by receiving overdue video frames as a result of retransmission by RLC AM, which introduces cumulative jitter. In video streaming applications, cumulative jitter can be compensated by a playback buffer at the streaming client. A fixed playback buffer time of 300×TTI (i.e., 3s) was used throughout the simulation. The MOS obtained is directly proportional to the duration of the playback buffer. In other words, a longer buffering time can tolerate higher TBLER. Even though, a playback buffer of fixed duration was used for both video clips, the achievable MOS is different. The "Salesman" video clip can tolerate higher TBLER because it has smaller encoded video frame size than "Music". This leads to shorter cumulative jitter as illustrated in Figs. 10 and 11, which show the cumulative jitter for each encoded video frame arrived at the streaming client. For instance, for TBLER of 35%, the cumulative jitter for "Music" is approximately 4 to 5 times larger than "Salesman". Also note that, for 20% TBLER, the cumulative jitter of "Salesman" is below the playback buffer duration while the cumulative jitter of "Music" exceeded the playback buffer duration at frame number above 200. Once the cumulative jitter of a video frame exceeded the playback buffer duration, the video frame is useless and discarded. The rate at which the video frame passed the playback buffer duration is depicted in Fig. 12.

As observed in Figs. 10 and 11, the cumulative jitter increases significantly for larger TBLERs because the probability of some RLC blocks that need to be retransmitted is also higher. With the number of retransmission rises, the cumulative jitter increases by multiples of TTIs. Additionally, the in-sequence delivery of RLC AM delays all correctly received IP packets until all previous IP packets have been correctly received. This delays buffered IP packets at RLC according to the slowest IP packet.

A larger playback buffer allows more erroneous RLC blocks to be recovered, which improves video quality at the expense of longer start-up time experienced by the viewer and extra storage capacity at the streaming client. For a 384 kb/s transport channel and 3s buffering time, a buffer size of at least 144 kbytes is required.

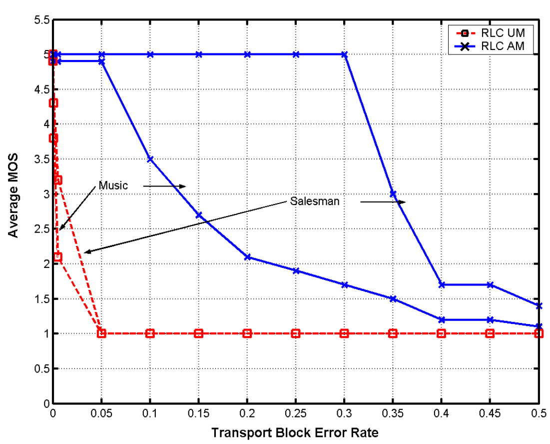

Figure 8. Average MOS as a function of Transport Block Error Rate

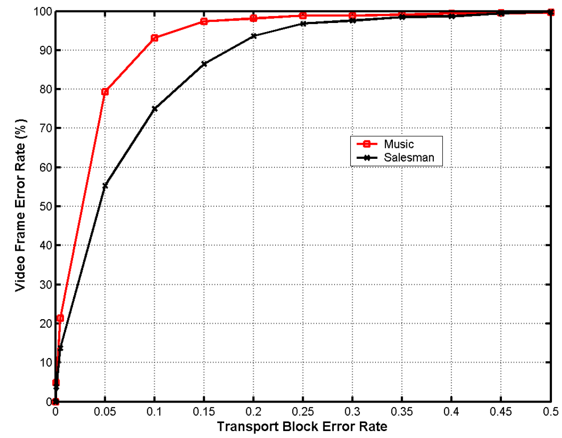

Figure 9. Video Frame Error Rate for RLC UM – "Salesman" and "Music"

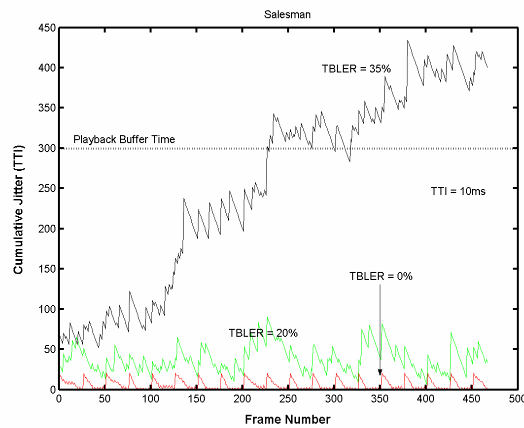

Figure 10. Cumulative Jitter for RLC AM – "Salesman"

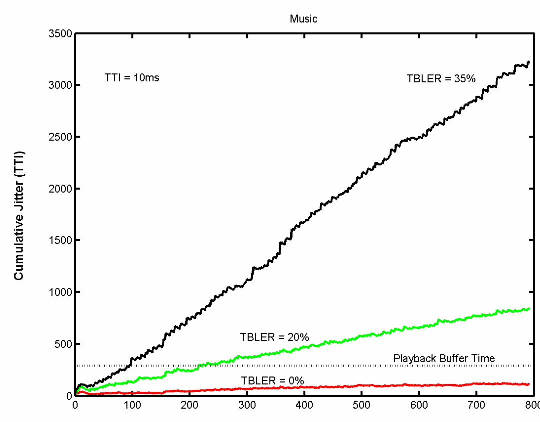

Figure 11. Cumulative Jitter for RLC AM – "Music"

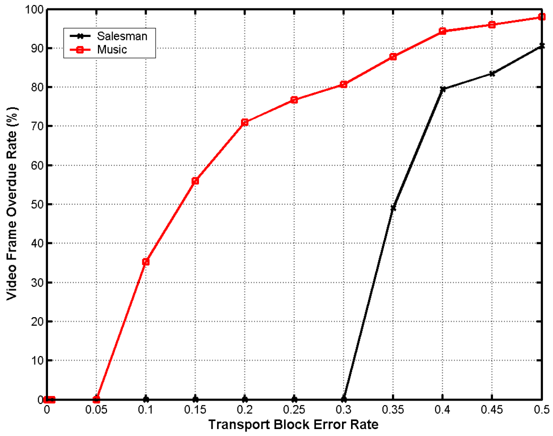

Figure 12. Video Frames which exceeded Playback Buffer Time for RLC AM – "Salesman" and "Music"

6. Proposed Self-adaptive RLC

It is clear from the simulation results in Section 5 that the RLC AM is the desired operating mode for video streaming. However, the standardized RLC AM is not fully optimized for video streaming. For instance, the cumulative jitter is increasing with streaming duration for high TBLERs because of the persistent ARQ mechanism which causes IP packets which carrying video frames to queue up at the RLC. Obviously, the increasing cumulative jitter can be reduced through a limited number of RLC block retransmissions. That means, an RLC block is discarded once the maximum allowable retransmission is reached. Even though, this approach can keep the cumulative jitter within the playback buffer time-bound, it does not necessarily improve perceived video quality. For example, discarding an RLC block which contains segments of I-frame would cause distortion of all the following P- and B-frames even they are received within the deadlines. Hence, a self-adaptive RLC AM using cross-layer design is proposed, which is aware of the MPEG-4 frames, playback buffer duration and the UMTS radio interface round-trip delay.

With the proposed self-adaptive RLC AM, the maximum allowable retransmission should be set according to the target TBLER and playback buffer time. In addition, video packets which cannot be transmitted within a given time-bound are removed from the queue at RLC. The time-bound should be derived from the playback buffer time and set according to the importance of video packets. Therefore, B-frames should be the first to be discarded followed by P-frames, and lastly, I-frames.

The UMTS radio interface round-trip delay can be reduced by using an unequal target TBLER in the uplink and downlink directions. For instance, a lower TBLER for status messages in the uplink direction.

7. Conclusion

The paper has evaluated the performance of streaming MPEG-4 video over UMTS networks using an integrated tool environment. The perceived video quality is evaluated in terms of MOS, cumulative jitter and video frame error rate. Two video clips with distinct characteristics were considered. Simulation results show that the perceived video quality achieved using UMTS RLC acknowledged mode is superior to unacknowledged mode. The performance of unacknowledged mode is worse because it does not provide error recovery as in the acknowledged mode case, and the MPEG-4 decoder is very sensitive to erroneous video frames. The poor perceived video quality of the unacknowledged mode is further aggravated by error propagation. The acknowledged mode can tolerate transport block error rates up to 30% before showing any sign of video quality degradation due to the increase in cumulative jitter which causes the playback buffer at the streaming client to underflow. Based on the simulation results, a self-adaptive RLC acknowledged mode is proposed, which is aware of MPEG-4 frames, playback buffer size and UMTS radio interface round-trip delay.

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