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Subjective and Objective Evaluation of the Effect of Packet Loss and Delay on Video Streaming Quality

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Abstract — **Video streaming is commonly used to deliver videos over IP based networks such as the internet. Unlike video downloading, in video streaming, the video is played out while parts of it are being received and decoded. Network congestion is a common problem in such networks. It can result in the loss of video packets or a delay in their arrival at the destination which can affect the received video quality. Packet losses and delays can happen in different possible patterns and magnitudes each of which has a different effect on the quality of the decoded video. That is why, it is important to identify the different patterns and magnitudes in which packet losses and delays can happen so that strategies can be formed to counter their effects on the video quality. In this paper, the effect of packet loss and delay on the objective and subjective quality of the received video is assessed. The experimental results show that the video streaming quality suffers because of packet loss and variable delay. It also shows that, overall, there is a strong correlation between the results of the objective and subjective tests.**

Keywords: video streaming; video quality; packet loss; delay.

I. INTRODUCTION

Nowadays, real-time video communication over the internet has become very popular. Video compression and high speed networks are the technologies which have helped to make it possible. Real-time video, as its name suggests, has rigid timing requirements e.g., a video must be played out continuously. If the required data does not arrive in time, then the video will experience delay which can be annoying to human eyes.

Real-time video communication can be divided into two categories. These are: communication of live video and communication of stored video. In this paper, we consider the scenario of the communication of stored video. Two common modes of transmitting stored videos over the internet are video streaming and download modes [1]. In download mode, a video file must be downloaded completely before it can be watched. In contrast, in streaming mode, the video can be played back while parts of it are being downloaded. A common issue with video streaming is that it has timing constraint. For example, if the video data is not received in time, the video playback might stop temporarily until the required data is received. This delay in playback is generally undesirable and can be annoying.

Another common issue is that of the loss of video packets during transmission. When the required video packets do not arrive at the receiver, the quality of the video play out at the receiver suffers. Both these factors, delay and packet loss, can happen in a number of ways. For example, delay can be of different lengths while packets can be lost either individually or in a burst (a sequence of consecutive packets). Moreover, the percentage of packets lost can also wary. All these possible scenarios will lead to different effects on the quality of the received video. It is important to understand the effect of each of these scenarios on the quality of the received video.

Video quality can be evaluated using mathematical models (objective evaluation criteria) or using human judgement (subjective evaluation criteria). Both criteria have their strengths and weaknesses and may provide unique insights. That is why, in this paper, we use both objective and subjective video quality evaluation criteria to assess the effect of delay and packet losses on the quality of the received video.

The rest of the paper consists of the following four sections: Section II contains a study of the fundamental concepts of video streaming, including its typical architecture, important metrics for the evaluation of QoE in video streaming etc. A detailed literature survey is presented in Section III. Section IV focuses on the experimental part of this paper. It outlines the experimental setup, the results obtained from the tests and a detailed discussion of the results. Finally, we present our conclusions in Section V.

II. FUNDAMENTALS OF VIDEO STREAMING

A. Video Streaming Architecture

A common video streaming architecture is shown in Fig. 1. It has three main components: a streaming server, a client or a receiver, and the internet. The raw audio and video are compressed using a suitable video compression algorithm. The Application Layer QoS Control then adapts the compression video bit stream according to the network status and QoS requirements. The compressed and adapted video bit stream is then packetized using a suitable transport protocol. After packetization, the packets are sent over the internet. Inside the internet, a packet may be dropped or it may experience long delay due to network congestion. The packets that successfully reach the receiver have to go through the transport layer where they are de-packetized into a compressed bit stream. Finally, after going through the application layer, the bit stream is decoded and the video is made available to the user.

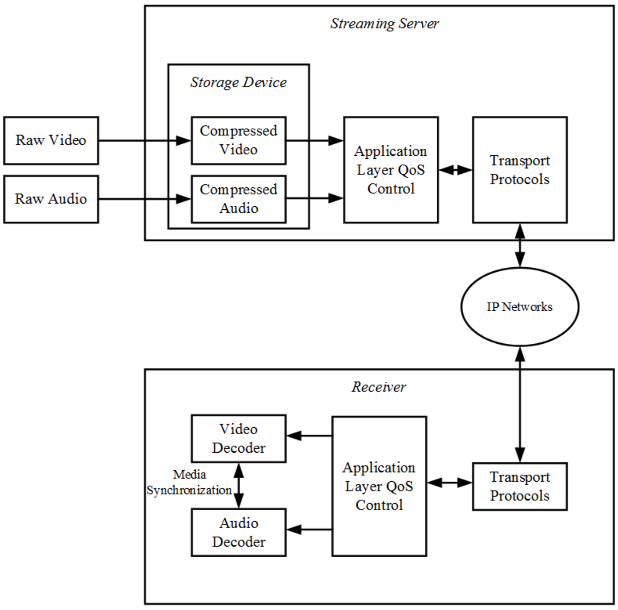

Figure 1: A typical video streaming architecture

It can be seen from Fig. 1 that there are four essential and related components in video streaming. These are briefly described here.

1. *Video compression:* Transmitting a raw video over the internet can be both challenging and expensive. Therefore, it is necessary to compress a raw video before transmission. Video compression methods can be broken down into two types. These are: scalable video compression and non-scalable video compression.

2. *Application layer QoS control:* A user may request any level of quality of experience. Similarly, the network conditions may vary. That is why, a number of application-layer QoS control methods have been proposed [2], [3], [4]. Congestion control and error control are two common types of application layer QoS control schemes. They work in the following manner: congestion control is used to minimize delays and to avoid losing packets. On the other hand, error control is used to improve the quality of the video in case that the video packets suffer delay or packet loss. Some important error control techniques are: forward error correction (FEC), retransmission, error-resilient encoding, and error concealment.

3. *Streaming servers:* A streaming server is one of the most important components of the video streaming architecture. It plays an essential part in enabling streaming services. Advanced streaming servers are expected to process videos in a timely fashion. Moreover, they are required to provide additional features such as pause/resume, fast forward, fast backward etc. Additionally, a streaming server may be required to allow the synchronous retrieval of a media component. Generally, a streaming server is made up of three components. These are: a communicator (e.g., transport protocols), an operating system, and a storage system.

4. *Protocols for streaming media:* Before the client and the server can communicate, they are required to set rules for their communication. These sets of rules are called network protocols. A number of standard protocols exist which can be used for the communication of client and servers. These protocols set rules for services such as network addresses, transport and session control. Network protocols can be generally split into three types:

   a. network-layer protocol such as Internet protocol (IP),

   b. transport protocol such as user datagram protocol (UDP), and

   c. Session control protocol such as real-time streaming protocol (RTSP).

B. Video Streaming Tools

A number of publicly available tools can be used to stream videos over the internet. The tools that were used during our experiments are briefly introduced here.

Video LAN Client (VLC) media player [5] is an open source application software which is available under GNU General Public License. It has support available for a number of common video file formats (e.g., WMV, AVI, etc.) and encoding formats (e.g., H.264, MPEG-4 etc.). VLC can also be configured to be used as a streaming server. As a streaming server, it is compatible with a number of protocols such as HTTP, UDP, RTP, RTSP etc. VLC player provides the option to compress the video before streaming. It provides several ways of compressing the video. In this paper, we use H.264/MPEG-4 codec for compression. Some of the advantages of MPEG-4 are: In comparison with the previous standards, it gives about 50% improved bit-rate efficiency. It has also been used in several applications such as HD DVD, HD-DTV etc. It is flexible and handles a number of tools and also well designed for applications ranging from low bit rate and low resolution to high bit rate and high resolution [6] [7]. Similarly, VLC provides a number of protocols to stream the video. In this paper, we use UDP as the transport layer protocol while RTP is used an application layer protocol. Since UDP is a real time protocol, it is used for streaming videos over the network. The complexity of using UDP is very low and at the client side as well as at the server side, additional memory is not required. Moreover, it is possible to achieve an acceptable subjective quality using UDP [8]. On the other hand, an important limitation of UDP is that it does not support reordering of packets.

Clumsy [9, 10] is a utility which uses the WinDivert library to stop network packets and perform a number of operations on them e.g., lag/drop/tamper etc. according to the user demand, and then release the packets [9]. It is a useful tool to track down unknown problems related to broken networks, or to evaluate the performance of an application under different network conditions. In this paper, Clumsy was used to simulate video packet losses and delay.

C. Quality of Experience in Video Streaming

1) Methods of evaluation

In video streaming, quality of experience (QoE) provides an assessment of human perception of the quality of the video [11]. The metrics that are used for evaluating the QoE in video streaming can be classified into two categories: objective and subjective.

Objective Assessment: In objective assessment, a mathematical formula is used to evaluate the quality of a video. Objective metrics include Mean Squared Error (MSE), Peak Signal to Noise Ratio (PSNR) etc. For a reference frame A = $\{a_1 \ldots a_M\}$, and a distorted frame B = $\{b_1 \ldots b_M\}$, each containing M-number of 8-bit pixels, these metrics can be calculated as follows:

$$MSE(A,B) = 1/M \sum_{i=1}^{M} (a_i - b_i)^2 \quad (1)$$

$$PSNR(A,B) = 10 log_{10}\left(\frac{255^2}{MSE(A,B)}\right) \quad (2)$$

Subjective Assessment: In subjective assessment, human judgment is used to evaluate the quality of a video. Mean Opinion Score (MOS) is commonly used as a subjective metric. In MOS, first, the original and the distorted videos are shown to a group of viewers. Next, the opinion of the viewers are recorded and averaged to evaluate the quality of the video. Each viewer gives a score in the range 1 – 5 to each video according to the criteria specified in Table 1.

Subjective video tests can be conducted using a number of standard approaches. These approaches include: Single Stimulus (SS), Double Stimulus Impairment Scale (DSIS), Double Stimulus Continuous Quality Scale (DSCQS), and Single Stimulus Continuous Quality Evaluation (SSCQE), Simultaneous Double Stimulus for Continuous Evaluation (SDSCE), and Stimulus Comparison Adjectival Categorical Judgement (SCACJ) [12]. All these approaches are very similar with slight differences in the scale used for evaluation, the length of the video sequence, the number of viewers, and the number of trials etc.

2) Important Factors

Bandwidth: Video streaming applications normally have a minimum bandwidth requirement. When the minimum bandwidth requirement is not met, the quality of video streaming is affected. One of the solutions to this problem is a technique called rate-control which modifies the sending rate of the video according to the available bandwidth.

Delay: Delay is an important factor in video streaming as video streaming applications require a reasonable end-to-end delay limit so that at any time, the required video packets can arrive in time at the receiver, and can be decoded and displayed. When the end-to-end delay limit is passed while the packets have not arrived, the playout is affected in the shape of pauses in the playout which is annoying to viewers. If a video packet does not arrive in time, the playout process will pause, which is annoying to human eyes. That is why, when a packet does not arrive in time, it should be considered lost. One of the solutions to this problem is the introduction of a buffer at the receiver [13].

There are two important aspects of delay: duration of delay and delay jitter (or variation in end-to-end delay).

Loss: Video packets can be lost during transmission. When this happens, at the receiver side, the video cannot be decoded properly i.e., the resultant video content is of poor quality. Thus, it is important that a video streaming application can deal with packet losses i.e., it should have a mechanism which can help in decoding acceptable quality video even in the event of packet loss.

Losses may occur in different ways. For example, in wireless channels, typically, losses happen in the form of bit errors or burst errors whereas, in wired networks, losses typically happen in the form of packet loss where an entire packet is lost [14]. Typically there are two important aspects of loss: the amount of loss and the burst size (i.e., the number of consecutive lost packets). Commonly, for distributed losses, a uniform packet loss model is considered while for burst losses, Gilbert-Elliot (GE) model is used [22].

III. RELATED WORK

To achieve good quality streaming, video streaming applications commonly impose two important types of constraints on a video encoder and decoder namely delay and losses. Delay, because video packets must arrive in time at the receiver so that it can be decoded and displayed. Therefore, end-to-end delay is important. In addition, packet delay variation (jitter) also plays an important role in the perceptual quality of the video [15]. The effect of packet delay can be overcome using different methods including adaptive playout [16]. Video packets can be lost during the transmission which can damage the video quality. Therefore, packet losses play an important role in the quality of the received video [17]. They effect the user's quality of experience (QoE) [18].

Shmueli et al. [19] analysed the perceived quality of a compressed video stream transmitted over a lossy IP network with the help of a quality of service mechanism and found it possible to assess the subjective quality of the video stream from a set of known parameters. They called this method the Canonical Discriminant Analysis (CDA).

Khalifa and El Mahdy [20] analysed the impact of error propagation on videos and found that the impact of error propagation on video streaming can be mitigated by using Mixed Streaming (MS). In MS, a video file is transmitted over both reliable and best effort connections. Moreover, this method makes sure that the errors in reference frames are corrected on a priority basis.

Fiedler et al. [21] proposed the use of throughput histograms as a measure of the Quality-of- Service and concluded that when such histograms are exchanged between a sender and a receiver, they can help in identifying the types of performance bottlenecks in the network (if any). Moreover, Fiedler et al. concluded that further research was required to obtain thresholds for acceptable quality of experience to the users.

The effect of packet loss video streaming quality was assessed by Lin et al [22]. In this work, the effects of the distribution (distributed and burst) of packet losses and the packet size were considered. It was found that for the same packet loss rate, video quality suffers more if packets are lost in burst rather than distributed/isolated. Moreover, it was concluded that the smaller the packet size is, the worse is the impact of packet loss on video streaming quality.

Lee and Kim [23] and Tsai and Leou [24] analysed the effect of channel characteristics on the quality of video streaming. While the former proposed to adjust the video frame rate according to the channel characteristics in a time-varying rate channel, the latter, proposed a similar method for a constant bit rate (CBR) transmission channel. Both the studies concluded that the quality of video streaming can be determined by modelling of the error characteristics of the link layer.

Error resilient video streaming techniques for mobile networks were studied by Nemethova et al. [25]. The videos were compressed using the H.264 video compression standard. In particular, the impact of residual redundancies in video streams was observed. It was found that redundancy helps in error localization and impairment detection.

To mitigate the effect of jitter, Sen et al. [26] proposed the introduction of a few seconds of start-up delay and a client buffer. However, this scheme requires a dedicated smoothing server and is suitable only for wired networks with guaranteed bandwidth. Similarly, Varsa and Cursio [27] considered having two buffers at the receiver: a delay jitter buffer and a decoder buffer. Stockhammer et al. [28] compared the above single and double buffer approaches and found that a single buffer can perform at least as good as two separate buffers.

Liang et al. [29] studied the effect of burst length for estimating the expected distortion occurring due to the loss of video packets. The focus of the study was low bit rate videos. It concludes that pattern plays an important role in determining the quality of the distorted video and that burst losses have a more profound effect on the quality of the distorted video compared to isolated losses.

However, these studies use either only subjective evaluation or only objective evaluation of the video streaming quality. Observing both the objective and subjective video streaming quality brings new insights about the effect of delay and packet loss. That is why, in this paper, we use both objective and subjective video quality evaluation criteria to assess the effect of delay and packet losses on the quality of the received video.

IV. RESULTS

A. Test Videos

Four test videos were used in the experiments. These are shown in Table 1 and Fig. 2.

TABLE 1: VIDEO SEQUENCES AND THEIR PROPERTIES

Video Sequence	***News***	***Foreman***	***Soccer***	***Crew***
Resolution	352x288 (CIF)	352x288 (CIF)	176x144 (QCIF)	176x144 (QCIF)
Frames per second (fps)	25	25	15	15
Total Frames	300	300	150	150

Figure 2. The original test video sequences: (a) Crew, (b) Soccer, (c) Foreman, and (d) News

B. Experimental Setup

The videos were compressed using the H.264 video compression standard. During compression, the bitrates were set to 128 kbps, 256 kbps, 512 kbps and 1024 kbps. In the experiments performed for testing the effect of packet loss rates, both the client and server were on the same machine.

C. Delay Settings

To check the effect of delay on video streaming quality, the four test video sequences were streamed while introducing different amounts of delays using the Clumsy software. The software accepts values of delays in milliseconds. Different delay values were used to analyse the impact of delay on the viewer's perceived video quality. After conducting many experiments, the following delay values were finalized for the experiments: 20 ms, 40 ms, 60 ms, 80 ms, and 100 ms.

It is important to state that the effect of delay on video quality was evaluated only using the subjective tests.

D. Packet Loss Settings

To check the effect of packet loss on video streaming quality, two types of experiments were performed. The first type of experiment was conducted using standalone H.264 encoder and decoder based on the same machine. In these experiments, rtp_loss.exe tool was used to introduce packet loss. The results of these experiments were evaluated objectively. The second type of experiment was conducted using the client and server approach in which one machine was configured as a client and another was configured as a server. Clumsy was installed on the server machine. Before transmitting each of the videos to the client, Clumsy was used to introduce different amounts of packet loss. The packet loss values that were used in these experiments are: 0% (no packet loss), 1%, 2%, 3%, 4%, 5%, 6%, 7% 8%, and 9%. An important point which should be noted is that both the rtp_loss.exe tool and Clumsy lose packets randomly.

E. Objective Tests

The tools used for the objective tests are: H.264 Encoder, H.264 Decoder, and RTP_Loss tool of H.264. In the objective tests, videos were encoded using the H.264 encoder. To simulate packet loss, the RTP_Loss tool of H.264 was used to simulate the loss of 1%, 2%, 3%, 4%, 5%, 6%, 7%, 8%, and 9% packets. The encoded bit stream file (after simulating packet loss) was decoded to obtain the reconstructed video at the receiver side.

The results of objective tests are shown in Fig. 3 – Fig. 6 and are discussed below.

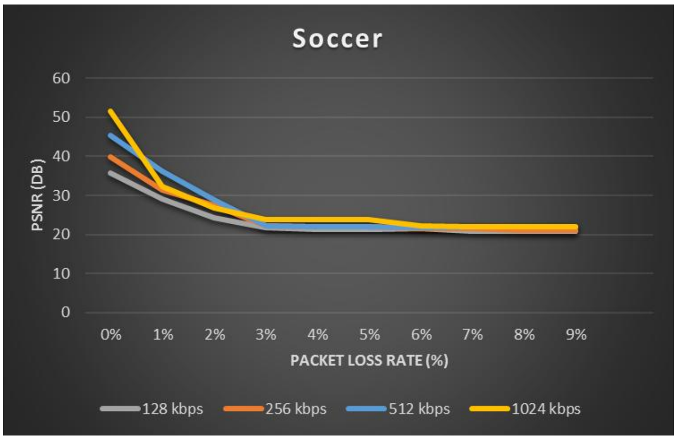

Figure 3. The effect of different packet loss rates on the PSNR (quality) of Soccer video at different bit rates

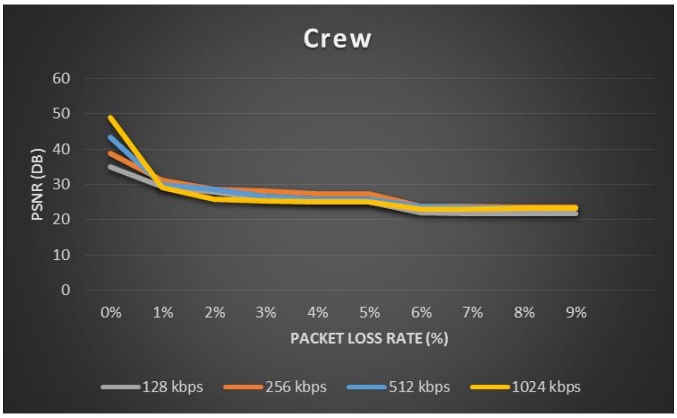

Figure 4. The effect of different packet loss rates on the PSNR (quality) of Crew video at different bit rates

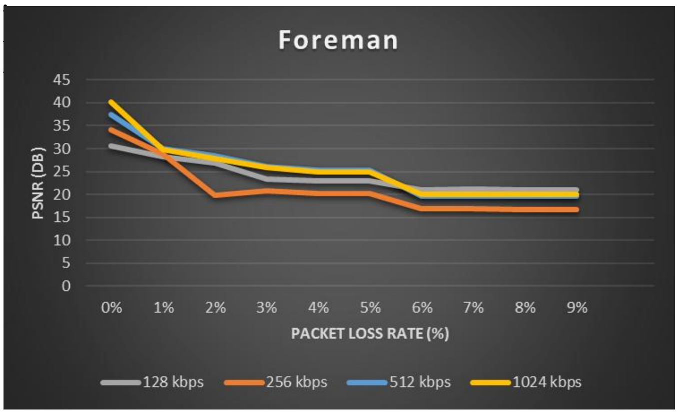

Figure 5. The effect of different packet loss rates on the PSNR (quality) of Foreman video at different bit rates

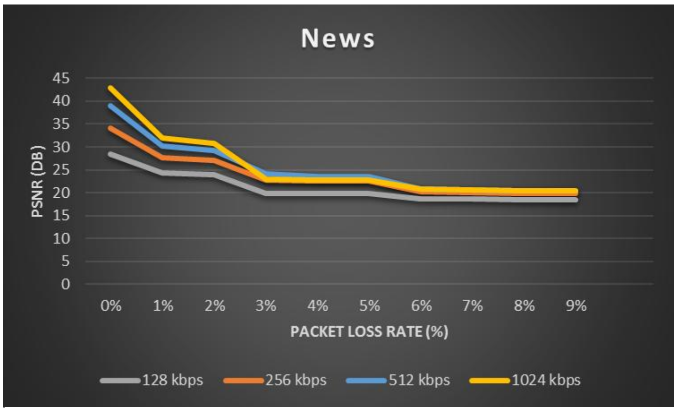

Figure 6. The effect of different packet loss rates on the PSNR (quality) of News video at different bit rates

1)Effect of encoding at different bit rates: The results in Fig. 3 – Fig. 6 show that the objective quality of a video depends on the encoding bit rate. Low bit rates correspond to low quality video while high bit rates correspond to high quality video. For example, when there is no loss (0% packet loss rate), the soccer video has a PSNR of 35.84 dB at 128 kbps, 39.87 dB at 256 kbps, 44.43 dB at 512 kbps, and 51.64 dB at 1024 kbps. A similar trend is observed for the remaining three videos as well.

2)Effect of packet loss: The results in Fig. 3 – Fig. 6 show that packet loss affects the quality of a video. It can be seen in these results that, generally, as the packet loss increases, the quality of the video decreases. For example, the uncorrupted crew video (without any packet loss) has an average PSNR of 35.06 dB at 128 kbps. The PSNR drops to 29.41 dB, 27.84 dB and 25.75 dB respectively for packet loss rates of 1%, 2%, and 3% respectively. Similarly, the drop in PSNR is maximum for the first 1% lost packets and it is less sharp as the packet loss rate increases further. For example, for the crew sequence, at 128 kbps, at 1% packet loss rate, the drop in PSNR compared to error-free transmission is approximately 5.5 dB. For a further 1% loss (2% packet loss rate), the drop in PSNR is only approximately 1.5 dB. It can also be said that, for some videos, when the packet loss rate increases, the difference between the qualities (dB) of the videos at higher bit rates and lower bit rates decreases. This trend can be very easily seen in Fig. 3 – Fig. 6. For example, the chart for soccer and crew videos show that the lines corresponding to the different bit rates superimpose each other for packet loss rates greater than 3% and 1% respectively.

3)Effect of packet loss at different bit rates: The results in Fig. 3 – Fig. 6 show that the effect of packet loss is more prominent at higher bit rates. For example, for the Foreman sequence, the loss in PSNR (compared to the error-free transmission) at 1% packet loss rate is ~2.5 dB at 128 kbps, ~5.5 dB at 256 kbps, ~7.5 dB at 512 kbps, and ~10.5 dB at 1024 kbps. The same trend can be observed for the other three videos as well.

4) Effect of packet loss for videos of different resolutions: It can be seen from Fig. 3 – Fig. 6 that compared to the CIF sequences Foreman and News, the QCIF sequences Soccer and Crew suffer more in case of packet loss. For example, at 8% packet loss the Soccer and Crew sequences lose approximately 23 dB and 20 dB respectively compared to the scenario where no packet loss occurs. On the other hand, the CIF sequences Foreman and News lose approximately 17 dB and 19 dB respectively. It is interesting to note that most of the quality degradation happens at the first 2% packet loss. For example, the Soccer, Crew, Foreman, and News sequences lose approximately 17 dB, 15 dB, 9 dB, and 10 dB respectively. For the next 2% packet loss (overall 4%), the losses are approximately 7 dB, 3 dB, 3 dB, and 6 dB respectively. Similarly, a further 2% packet loss (overall 6%) causes the loss of 0.5 dB, 2 dB, 6 dB, and 3 dB respectively.

F. Subjective Tests

The tools (and equipment) used for the objective tests are: Clumsy, VLC player, and two PCs. The tests were performed by setting up VLC on two different machines. One of the machines was configured to work as a server while the other was configured to act as a client. Clumsy was installed on the server machine. The delay and packet loss settings were set using the Clumsy software before transmitting a video from the server to the client. At the client side, the received videos were saved and shown to the participants of the subjective tests for evaluation.

1)Effect of packet loss: The results of subjective tests for the evaluation of the effect of packet loss on video streaming quality are given in Fig. 7. From these results, it can be observed that as the packet loss rate increases, the average MOS decreases. There are some exceptions to this trend as well. For example, for the crew video, the average MOS at 3% packet loss rate is 3.2 while that at 4% packet loss rate is 3.3. It indicates that the users are not equally sensitive to all types of errors. It is possible that at 3% packet loss rate of the crew video, the areas affected in the video are more important to users than the areas affected by 4% packet loss rate.

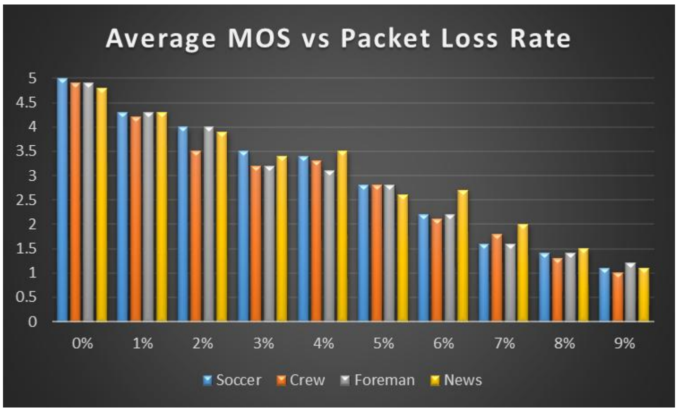

Figure 7. Average Mean Opinion Scores for different videos at different packet loss rates

2) Effect of delay: The subjective results of the effect of delay on video quality are shown in Fig. 8. The results show that as the delay is increased, the average MOS decreases. It should be noted that the delay is randomly inserted between packets. The delay values given in Fig. 8 correspond to the mean of the randomly inserted delays. For example the mean delay value of 20 ms does not mean that each packet experienced a delay of 20 ms but that each packet experienced a random delay and that the mean or average value of all the random delays was 20 ms.

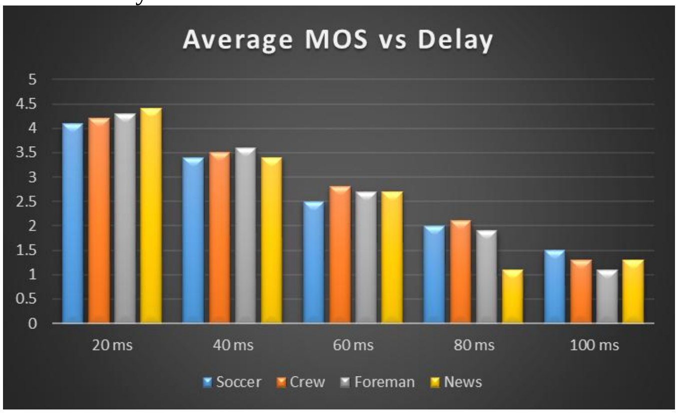

Figure 8. Average Mean Opinion Scores for different videos with different amounts of delay

G. Correlation between the results of objective and subjective tests

Both the objective and subjective tests were performed in the case of packet loss. That is why, it is important to find if there is any correlation between these results. When the results of these tests are compared against each other, it can be seen that for some of the results, there is a high correlation while for some other results the correlation is low. For example, both the objective and subjective tests reveal that an increase in packet loss rate decreases the MOS which means that increasing the packet loss rate decreases the objective as well as subjective video quality. On the other hand, unlike the results of the objective tests, no direct relation between the resolution of the video and the effect of packet loss was found in the subjective tests.

V. Conclusion

In this paper, the effect of packet loss and packet delay on the quality of video streaming was studied. It was found that the effect of constant packet delay on the quality of video streaming is much smaller than that of variable delay. Moreover, variable delay can significantly reduce the users' quality of experience. It was also found that the user's MOS for a video is affected by the amount of variable delay. As the delay increases, the MOS decreases and vice versa. Similarly, packet loss can severely affect the quality of video streaming. The user's MOS for a video is affected by the amount of packet loss. As the amount of packet loss increases, the MOS decreases and vice versa.

Overall, a strong correlation was found between the objective and subjective test results for the effect of packet loss on video quality while no correlation was found between the objective and subjective test results for the relationship between the resolution of a video and the effect of packet loss on it. Lastly, it was found that the effect of packet loss is more prominent at higher bit rates.

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