Guaranteed QoS for Selective Video Retransmission

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Abstract

Video Streaming presents viewers with the opportunity of viewing video on the internet without necessarily downloading it. This provision however comes with a lot of challenges amongst which are packet loss and propagation error. The above mentioned can drastically reduce the video quality at the receiving end. Consequently, error resilience and effective management of packet loss are critical issues while handling streaming media.

This research examines the challenges that make simultaneous delivery and playback, or streaming of video difficult over packet switched networks with an attempt to improve the drastic effect of packet loss on video streaming and propose a model that enable appropriate streaming of pre-encoded or live video over packet networks such as the Internet. The above was accomplished by extending the operation of existing transmission mechanism to involve selective retransmission in other to conserve the bandwidth utilization.

Keywords: *Video Streaming, QoS parameters, Propagation error, Video encoding.*

1. INTRODUCTION

Video has been an important medium for communication and entertainment for many decades. Initially video was captured and transmitted in analog form. The advent of digital integrated circuits (DIC) and computers has led to the digitization of video and digital video thereby prompting a revolution in the compression and communication of video [6]. Video compression became an important area of research in the late 1980’s and 1990’s due to the sporadic growth and popularity of the internet, thus enabling/accommodation varieties of applications including video storage on DVD’s and Video-CD’s, video broadcast over digital cable, satellite and terrestrial (over-the-air) digital television (DTV)[11], and video conferencing and videophone over circuit-switched networks.[6]

Generally, streaming involves sending media (e.g., audio and video) from a server to clients over a packet-based network such as the Internet. The server breaks the media into packets that are routed and delivered over the network. At the receiving end, the packets are reassembled by the client and the media is played as the packets arrive. A series of time-stamped packets is referred to as *stream*. From a user’s perspective, streaming differs from simple file transfer in that the client plays the media as it comes in over the network, rather than waiting for the entire media to be downloaded before it can be played. In fact, a client may never actually download a streaming video; it may simply reassemble the video’s packets and play the video as the packets come in, and then discard them [2].

Most multimedia traffic does not provide error handling or congestion-control [5, 9]. Providing error resilience and error concealment to this time sensitive traffic as well as recovering from packet loss due to congestion in the network has been an area of intense research in recent past. This problem is further compounded when the data is transmitted using MPEG frames. In such cases, loss or corruption of a single frame may propagate the effect to all the remaining frames thus rendering a sizeable length of the stream useless [5]. The process of source coding involves converting the original picture into a sequence of bits and compressing the same. The diagram shows the basic communication process that takes place in video streaming. The information from the original video is being encoded by the transmitter and later it will be decoded and sent out through the receiver to produce a reconstructed video that will be displayed to the viewer. [16]

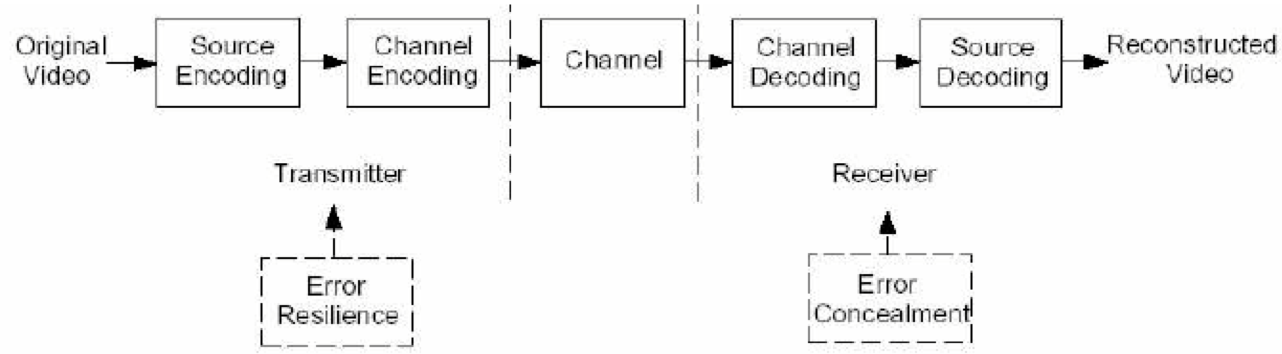

Fig. 1. Basic Structure of a communication system

Congestion is a common phenomenon in communication networks that occurs when the offered load exceeds the designed limit, causing degradation in network performance metrics such as throughput. Other symptoms of congestion in packet networks may include packet losses, higher delay and delay jitter. To avoid the undesirable symptoms of congestion, control procedures are often employed to limit the amount of network load. Such control procedures are called rate control, sometimes also known as congestion control. It should be noted that different network technologies may implement rate control in different levels, such as hop-to-hop level or network level [12, 14].

Large numbers of alternatives have been suggested for the recovery of packet loss due to congestion. This research will use selective retransmission technique to recover lost packet so as to enhance and improve the video quality. This is due to the fact that most internet protocols rely on retransmission to recover lost packets. And by this technique it is believed that vital information will not be lost due to time constraint. The rest of the paper is outlined thus: section 2 present an in-depth overview of existing research work, video streaming methods and challenges associated with streaming. Section 3 presents an overview of the new model. In section 4, we discuss our simulation results with the existing schemes and point out the remarkable difference in the QoS delivered to the users in our new scheme as opposed to the existing ones. Section 5 concludes the paper with useful recommendations.

2. REVIEW OF EXISTING LITERATURE

Review of previous researches depicted that error concealment are achieved in dependent frames by replacing missing macro blocks with a motion-compensated block from a different frame [5]. However, other researchers have shown that retransmission can be a feasible option for error recovery. M. Yajnik. et. al. proposes a retransmission-based error control technique without incurring additional latency by rearranging the temporal dependency of frames so that a reference frame is referenced by its dependent frames much later than its display time, thereby masking the delay in recovering lost packets.

Other schemes employ play out buffering, conditional retransmission requests, and various other techniques to alleviate the effects of packet loss. Previous work also analyzed MPEG-4's built-in error resilience capabilities and examined propagation of errors on an inter-frame video bit stream when bit errors occur [12].

Forward error correction (FEC) has been proposed in several projects as a means for providing error recovery and packet reconstruction [2, 10, 15]. However, these schemes rely on the fact that the FEC information contained in one packet itself is not lost. FEC-based schemes add redundant information, which can potentially worsen network traffic and aggravate existing packet loss which is bound to result from impending congestion.

Prioritization ideas were proposed to protect data of high importance on loss channels [14, 15, 16]. Quality assurance layering (QAL) protects high priority data with Forward Error Correction. One approach is to use the priorities associated with a bit stream to provide error protection at the time of encoding. This approach allocates more bits to more important information, such as header and motion information, while allocating fewer bits to texture information, within a particular video packet. Priority encoding transmission (PET) is an approach for sending messages over a lossy network based on a specified prioritization scheme. It has been used to provide a mechanism for using PET to create a hierarchical encoding of MPEG-1 video bit streams [4, 10].

Another approach to error concealment is multiple description coding (MDC), a joint sender-receiver approach for designing transforms. This scheme divides the bit stream into equally important “descriptions”, so that each additional description is useful in enhancing the quality of the received video.

Critical analysis of the above has revealed that all the approaches are similar due to the division of video frames into layers to improve QoS (quality of service) and they all depend on the play out time/deadline. Also, all of the approaches described above do not support retransmission due to the time constraint. Therefore, the goal of this research will be making use of the selective retransmission method so as to overcome the undue stress on the bandwidth.

From the above, the shortcomings of the existing approaches in improving the video quality can be summarised thus:

i. Any loss of a particular layer will affect the remaining frames.
ii. If more layers are dropped, the amount of the detailed picture will degrade considerably, thereby contradicting the video delivered to the viewers.
iii. Some use larger number of bits for similar video quality.
iv. Any packet that does not meet the deadline will be discarded.

Sequel to the above, a mechanism for recovering data in reference frames using selective retransmission method is proposed. It was observed that as more layers are dropped, some vital information/connecting frames were lost, thus resulting into jerk or jitter in the video. The new technique will allow the exact lost packet to be recovered without the problem of overloading the channel. This is based on our submissions that in providing a means for recovering from packet loss, a video streaming system for the Internet should adapt its sending rate and the quality of the video stream it sends in accordance with the available bandwidth. A video streaming system should deliver video at the highest possible quality for the available bandwidth and share bandwidth fairly with TCP flows.

2.1 Video Streaming Methods

There are Four Basic Steps to stream video. The following is a vastly outline of how to stream video. [7]

Step 1: Creating Content. Video content can be created through traditional means such as a camera, camcorder or VCR. However, the type of film and the filming technique will greatly affect the clarity of the resulting video stream.

Step 2: Digitizing the Video. The resulting video must be transformed into a digital file. This is accomplished by using a computer equipped with a video capture card installed. The VCR, camcorder or other video device is connected directly to the capture card via standard RCA cords or S-video cords. An S-Video connection usually results into better quality. Video editing software such as Adobe Premiere can be used to run the capture card, and edit the resulting file.

Step 3: Compressing/encoding the Video. The resulting digitized video files are usually too big for immediate transmission over the Internet, thus the file must be altered to reduce its size. There are many software packages that use different algorithms to accomplish this task known as video encoding. Once encoded, the files are only accessible on the player associated with the encoding software vendor. Some of the leading encoders include; Real Video, Media Player, Movie maker e.t.c.

There is no "best" way to encode video. It depends on content of the video (animation, fast motion, and talking head) and the upper limit of transmission speed that is available. Most articles recommend encoding several ways, and viewing the results on a control PC.

Step 4: Serving the Video. Once the video files are created, digitized and encoded, the next step is to serve the resultant stream to interested clients. The simplest method is to download the entire video file, and then view it with a plug-in or helper application. As opposed to "Raw" video files that are characteristically large, the effect of which is that users are not able to begin viewing the video until the entire file has been downloaded.

2.2 Major Challenges In Video Streaming

Video streaming over the Internet is difficult because the Internet only offers best effort service. That is, it provides no guarantees on bandwidth, delay jitter, or loss rate. Specifically, these characteristics are unknown and dynamic. Therefore, the major expectation of video streaming is to present a system that reliably delivers high-quality video over the Internet when dealing with unknown and dynamic factors which include:

o Bandwidth
o Delay jitter
o Loss rate

3 OVERVIEW OF THE NEW MODEL

As depicted above, video streaming is basically faced with packet loss, delay jitters and limited bandwidth which usually reduces the video quality. For example, any late frames resulting from the delay jitter can produce problems in the reconstructed video, accurately estimating the available bandwidth to match the pre-encoded video to the estimated channel bandwidth is another problem and additional considerations that make the video quality to be afflicted is the packet loss which can lead to the total erasure of a particular frame. Various approaches have been developed to improve the video quality but all seems to have one draw back or the other and these lead to the selective retransmission technique in which two frames are being sent at a time with the hope that at least one will get to the receiver and image will be displayed, otherwise, only the low quality frame is retransmitted. However, transmitting the dual frame will result into excessive bandwidth consumption, which can further escalate the problem of congestion. We present a diagrammatic representation of the dual transmission scheme below showing the low quality type as the dotted lines.

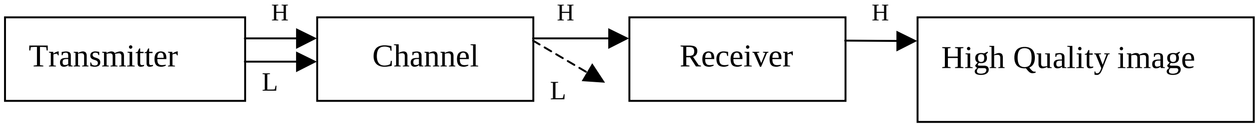

Image with high quality is being received while Low quality image is being discarded.
H = image with high quality
L = image with low quality.

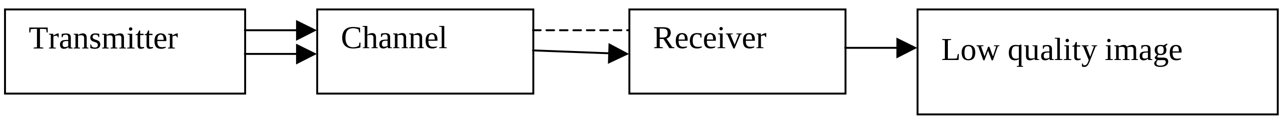

Image with low quality will be received due to loss or corruption of H

Fig. 2: Dual transmission method for video streaming

Below is the employed algorithm aimed at improving the video/image quality at retransmission.
1: We defined λ as the waiting time before retransmission)
2: Let T = t + λ (where T is the total time required to transmit while t is the time it will take a images to be displayed)
4: Divide the frames in to two that is, *H* and *L* (where *H* is the image with high quality and *L* is the image with Low quality)
5: Let H = 1 to N   Ɏ H {HєF э H=1…N}
6: Let L = 1 to N   Ɏ L {LєF э L=1…N}
7: F = H+L (where F is the frames to be sent)
8: Assuming the two gets to the receiver, discard L and display image with high quality
7: Otherwise display image with low quality
8: In a case where by both H and L are lost go back to step 1 above
9: Continue until the whole packet is delivered.

In this research, a total lost packet (i.e H and L) is replaced by the redundancy (λ) transmitted within the next packet. When the redundancy fails to repair the lost packet, a repetition occurs. The original video frame is being encoded into two versions, one with high quality and one with low quality. The high quality frames are sent as primary frames, while the low quality ones are considered secondary frames and piggy-backed with the next primary frame to repair any loss of a primary frame.

4 RESULT DISCUSSION

4.1 Multiple Descriptions Coding (MDC)

This method suggests transmitting more than one description in the hope that at least one of them reaches the destination. The effect of this is to ensure that quality images are received by the user but sending more packets at a time will surely consume more bandwidth and will lead to congestion over the network.

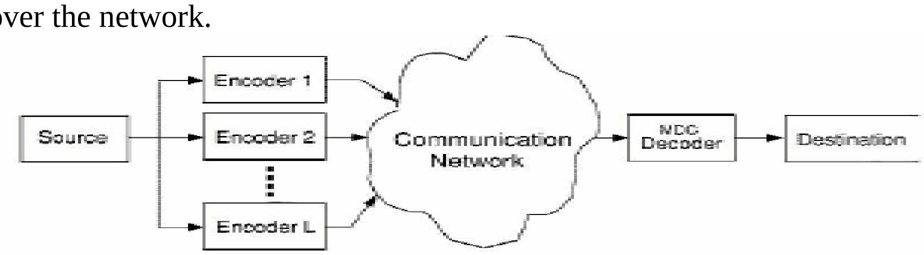

Figure 3: Multiple description coding

FORWARD ERROR CORRECTION (FEC) - This method adds specialized redundancy that can be used to recover from errors but it does not give room for retransmission and hence it will be difficult to recover any loss packet.

MOVING PICTURE EXPERT GROUP (MPEG) - This is also another method which divides the video streams into frames so as to improve the packet delivery and reduce loss but it is also not effective because some of the frames depend on one another and any loss of one frame will surely affect the other and this will affect the whole packet reception after transmission.

4.2 Major Contributions of the New Model

(a) <u>Bandwidth Utilization</u>

Bandwidth is one of the major challenges of video streaming and it is expected that less bandwidth will be used to transmit the images so that there will be more bandwidth for other activities. The new model discussed above vividly puts into consideration the amount of bandwidth to be used. This is sequel to the fact that images with low quality do not consume much bandwidth as the image with high quality which was transmitted first. Therefore, despite the fact that the new model transmits two packets at the instance of request, there is consideration for bandwidth utilization so that other frames were not distorted during the simultaneous transmission.

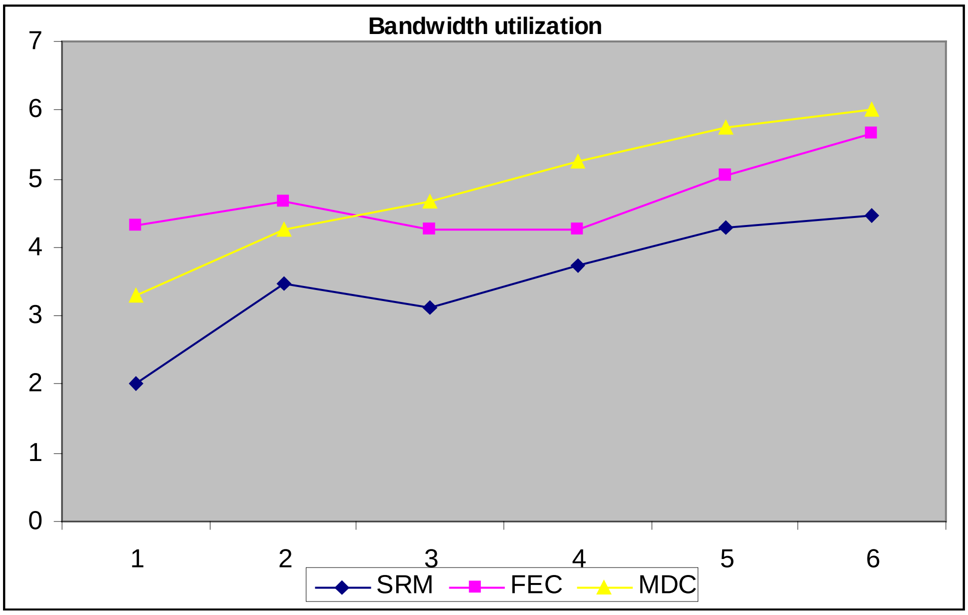

Fig 4. Bandwidth Utilization of the three schemes under consideration

The above figure depicts the bandwidth utilization from our simulation. It was found out that SRM impose the lowest bandwidth consumption as opposed to the other two schemes. Infact, it allows simultaneous transmission whenever the need arise without adversely distorting the bandwidth stress in the system.

(b) Reducing Congestion

This factor follows directly from the above, it is a common understanding that excessive bandwidth utilization normally results into congestion. The new model reduces congestion over the network by ensuring that the two packets are sent one after the other so as to avoid traffic which is the major cause of congestion. Although dual transmission scheme proves worthy in ensuring packet delivery, we deduced that loss of a pair of frame, and the request for its retransmission will surely disturb the transmission state. This results from the fact that, dual transmission employs almost twice the size of the bandwidth, and on an occasion when there is no loss, resulting into the discard of the low quality frame, this is a share waste of scarce resources which could have been employed in the transmission of another useful frame.

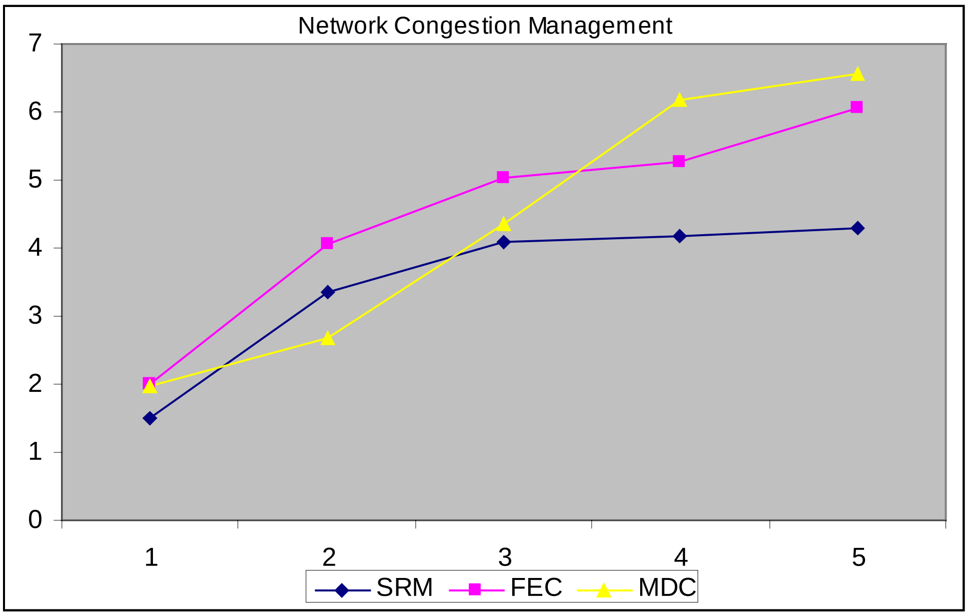

Fig. 5: Network Congestion Management for the three Schemes

Congestion management is an important issue to be considered in video streaming, the rate of its occurrence determines the QoS delivered to the users at every point in time. Thus users can only have maximum satisfaction if the quality delivered is close to negotiated quality at the point of transmission. The chart above shows the rate at which congestion occurs in the three schemes under consideration. SRM shows that after the initial congestion, it achieves a balance structure despite the need for some periodic retransmission whenever the need arise. The result is not the same for the other two schemes.

(c) <u>Reliability QoS Guaranteed for Service Delivery</u>

The new model is very reliable than other approaches in the sense that it guarantee quality of service negotiated by the user prior to the commencement of transmission. It also allows for selective retransmission in case of a loss. Retransmission technique is a very good practice in the transmission of files over a network but it takes more time and this is being conquered by the new model by the provision of extra time for retransmission to take place. This makes the new model to be unique and reliable because all other approaches did not allow retransmission and this means that any loss packet cannot be recovered.

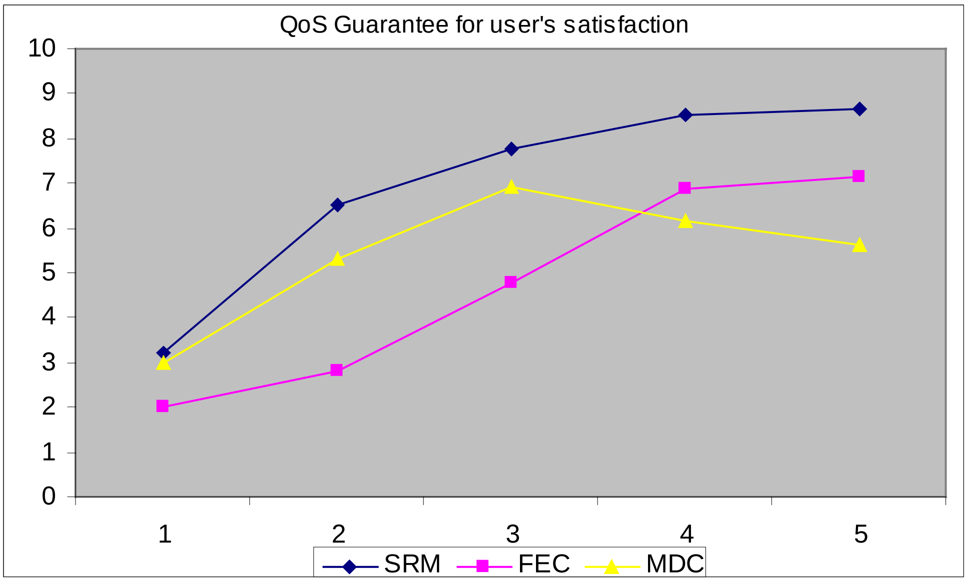

Fig.6: QoS Guarantee for user satisfaction

The new model ensures continuity because there will be no break in transmission even while the packet is being retransmitted due to the fact that the two packets are piggy packed with each other. So immediately one is missing, the other will replace it within few seconds.

5 CONCLUSION

This paper developed a model to handle the process of packet recovery and to ensure quality of service negotiated by the users at the point of transmission. It extends the dual retransmission scheme to improve bandwidth utilization, minimize congestion and also guarantee continuity in the transmission of video streams to the users.

The driving force lies in the fact that video streaming is becoming a very popular technology on the internet, however, the success of it operation in term of acceptability lies in the delivery of guaranteed quality of service (QoS) to the end users.

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