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ANALYSIS OF VIDEO STREAMING OVER MOBILE WIRELESS NETWORK

N. DINESH M.Sc., M.Phil., PGDHRM.,
Assistant Professor of Computer Science,
RDB College of Arts & Science, Papanasam – 614 205.

***ABSTRACT*−*Streaming media presents the professional communicator with a whole new way to deliver information, messages, and entertainment. By leveraging the Internet, distribution costs can be much lower than the traditional media. As third generation wireless networks are rolled out, it becomes feasible to view video from mobile appliances. This paper analyzes the quality of the streaming MPEG - 4 video over a mobile wireless network using an integrated tool environment, which comprises a network simulator, a video quality streaming tools which is Evolved. Through this work I establish guidelines for the transmission of video based on the mobile and wireless networks and leads to a conclusion of that is the link bandwidth must be greater than the video Streaming rate to viewing a good quality streamed video by the end user.*

1, INTRODUCTION

Mobile cellular telecommunication networks have been growing continuously. Users of modern telecommunication systems will expect the support of sophisticated services over wireless transmission. And one of these services is video. Recently more and more telecommunication systems are supporting different kinds of real-time transmission, video transmission being one of the most important applications. So streaming media is becoming prominent over current generation of mobile wireless network. Since bandwidth availability to the end user still has severe limitations even with the current high-end technology, such streaming are still limited to low-quality video. And to deliver a fair quality of video over this modern telecommunication system different types of video compression format have been invented. And most of these video compression formats are a loss compression due to the limitation of the bandwidth of the end users. So the delivering a good quality streamed video to the end user does not only depend upon medium of the transmission and is also depends upon the compression formats.

2, VIDEO COMPRESSION

Since the size and streaming rate of raw format of video file which is YUV format is huge enough and is not feasible to transmit this format over any wired and wireless network, so the raw format needs to be compressed. Compression reduces the number of bits used to represent each pixel in the image.
Compression systems exploit the mechanism of human perception to remove redundant information, but still produce a compelling viewing experience. As a result redundant data can be eliminated if the raw video file is compressed. Redundant data may consists of like by reducing the total numbers of colors, amplitude of neighboring pixel are often correlated, consecutive frames often having same object perhaps undergoing some movements. So a lower compression ratio results in less data being discarded and higher compression ratio results in higher data being discarded. Hence if the compression is increased more artifacts become apparent. That is why it needs to trade-off the level of artifacts of the video and the bandwidth if transmission medium.

2.1 Compression Algorithm

Compression can be lossless or lossy. If all the original information is preserved, the codec is called lossless. But for streaming video over mobile wireless network, data should be more reduced. That is why lossy compression is generally being chosen for streaming video over wireless networks.Compression algorithms aim at lowering the total number of parameters required to represent the signal, while delivering a reasonable quality picture to the player. These parameters are then coded into data packets for streaming. There are four main redundancies present in the video signal.
- Spatial
- Temporal
- Perceptual
- Statistical

2.2 Spatial

Spatial redundancy occurs where neighboring pixels in a frame of a video signal are related; it could be an object of a single color.

2.3 Temporal

Video is a sequence of similar images, with step changes at scene boundaries. In many sequences there is virtually no change from one frame to the next. In scenes with subject motion, or where the camera is moving, there will be differences from one frame to the next, but there are many areas of the picture that do not change. This redundancy of information from one frame to the next can be exploited to lower the data rate. The basis of the compression is to transmit only the difference between frames – ***frame differencing***. The player stores the entire picture in a frame store, and then reconstructs the picture from the previous frame and the difference information. Since most of the difference between frames is from moving objects, there is further potential to reduce the data.

2.4 Perceptual

Perceptual redundancy takes advantage of the varying sensitivities of the human visual system. The human eye is much more discriminating regarding changes in luminance than chrominance, for example, so a system with this feature can discard some color-depth information, and viewers do not recognize the difference.

2.5 Statistical

Statistical redundancy uses a more compact representation for elements that frequently recur in a video, thus reducing the overall size of the compressed signal.

2.6 International Standards of Video Codec

The video codec we use today come from two backgrounds: the first is the telecommunications industry and the second is multimedia. These are some of the most used codec.
- H.261
- AVC (Advanced Video Codec H.264)
- WMV
- Real Video
- MPEG-1, MPEG-2, MPEG-4
- And many more

2.7 MPEG-4 Codec

MPEG compression divides a video sequence into groups of pictures. The temporal compression is arranged in short sequence of frames called a group of pictures (GOP). MPEG defines three types of frame within the group.
- I-frame (Intra-frame)
- P-frame (Predicted frame)
- B-Frame (Bidirectional frame)

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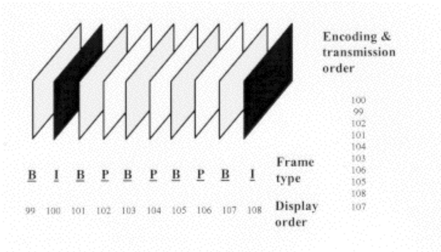

I-frame (Intra-frame)

These are coded spatially; solely from information contained within the frame. I-frames provide reference points for random access to a stream. The number of pictures between I-frames is set by the encoder, and can be varied to suit subject material.

P-frame (Predictive frame)

These are coded from previous I- or P-frame pictures. The decoder uses motion vectors to predict the content from the previous frames. The data in a typical P-frame are one-third of that in an I-frame.

B-frame (Bidirectional frame)

These pictures use past and future I and P pictures as a reference, effectively interpolating an intermediate picture. The B frames however, are coded based on a forward prediction from a previous I or P- frames, as well as a backward prediction from a succeeding I or P frame. B-frames are half that of a P-frame.

3, Characteristics of Mobile and Wireless Networks

IEEE 802.11

Wireless local area networks (WLANs) based on the IEEE 802.11 standard are a significant and viable alternative to wireless connectivity. The standard has currently three variations widely deployed. The 802.11b operates on the 2.4GHz band and has a maximum theoretical data rate of 11Mbps, but operates also on 1, 2 and 5Mbps. The 802.11a and g operate on the 5GHz and 2.4GHz bands respectively and both have a maximum theoretical data rate of 54Mbps. Using different modulation schemes they can also operate on the lower scales of 6, 10,12, 18, 36, and 48 Mbps. Based on CSMA/CA, a common resource sharing MAC protocol, 802.11 also adheres to the characteristic that the data rate allocated to each user is inversely proportional to the number of users in the local network. Therefore, the practical data rates are usually lower than those mentioned above.

4, VIDEOQUALITY ASSESSMENT

Objective QoS Measures

In an optimal case, the quality of video is monitored during transmission. According to measurements, adjustment of parameters and possible retransmission of the data is

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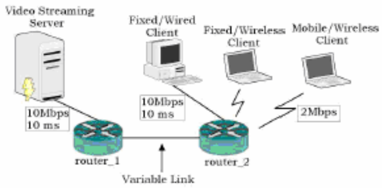
carried out. Objective quality assessment methods of digital video can be classified into three categories. In the first category, the quality is evaluated by comparing the decoded video sequence to the original. The objectivity of this method is owed to the fact that there is no human interaction; the original video sequence and the impaired one are fed to a computer algorithm that calculates the distortion between the two. The second category contains methods that compare features calculated from the original and the decoded video sequences. The methods of the third category make observations only on decoded video and estimate the quality using only that information. The Video Quality Experts Group (VQEG) calls these groups the full, the reduced and the no reference methods. Traditional signal distortion measures use an error signal to determine the quality of a system. The error signal is the absolute difference between the original and processed signal. The traditional quality metrics are the Root Mean Square Error (RMSE), the Signal-to- Noise Ratio (SNR), and the Peak Signal-to-Noise Ratio (PSNR) in dB. In this work I employ a Full reference method and use the PSNR as the objective quality metric. So in this work PSNR is calculated by the comparison of the sender side (original) raw YUV format video file with receiver side (processed) raw YUV format of video file. The receiver end (processed) raw YUV video file is being decomposed by the receiver end MPEG-4 codec files which are already missing of some redundant information due to compression of MPEG-4 encoder.

5, EVALUATION SETUP AND SCENARIOS

5.1 Topology



The evaluation topology consists of one Video Streaming Server, two backbone routers and video clients of variable types and connectivity methods (fixed, mo bile, wired, wireless) The video streaming server is attached to the first backbone router with a link which has 10Mbps bandwidth and 10ms propagation delay. These values remain constant during all scenarios.

This router is connected to a second router using a link with unspecified and variable bandwidth, propagation delay, and packet loss. The different parameter values used to characterize this variable link are shown in Table. Using this topology, I conducted several experiments for two different sample sequences and with fixed-wired clients, fixed-wireless clients and mobile-wireless clients.

5.2 Variable Test Parameters

The choice of the parameters used in the video quality evaluations was based on the typical characteristics of mobile and wireless networks, as these are described in Section 3. For example, the Link Bandwidth can be considered as either the last hop access link BW or the available BW to the user.

Video Stream Bit Rate	Link Bandwidth	Propagation Delay
64Kbps	64Kbps	10ms
128kbps	128kbps	50ms
256kbps	256kbps	100ms
512kbps	512kbps	200ms
768kbps	1Mbps	400ms

5.3 Test sequences

The test sequences used in this work were the sample sequences Foreman. The sequences were chosen because of their different characteristics. The first is a stream with a fair amount of movement and change of background, whereas the second is a more static sequence. Each sequence was encoded at the rates shown in Table 1. The video stream bit rate1 varies from 64Kbps to 768Kbps. This rate is the average produced by the encoder. Since the encoding of the sample video sequences is based on MPEG4, individual frames have variable sizes.

5.4 Data Collection

All the aforementioned experiments were conducted with an open source network simulator tool NS2. Based on the open source framework called EvalVid, I was able to collect all the necessary information needed for the objective video quality evaluation like PSNR values, frames lost, packet end to end delay and packet jitter. Some new functionality was implemented in NS2 from in order to support EvalVid. The whole data collection procedure and PSNR evaluation.

6, Appendix

List of Acronyms with abbreviations

- BW Bandwidth
- GoP Group of Pictures
- MAC Medium Access Protocol
- MPEG Motion Picture Expert Group
- MPEG-4 Motion Picture Expert Group Layer 4

- NS-2 Network Simulator 2
- PSNR Peak Signal to Noise Ratio
- QoS Quality of Service
- WLAN Wireless Local Area Network

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