THE JOURNAL OF

COMPUTER SCIENCE AND ITS APPLICATIONS

nigeria computer society
ncs
www.ncs.org.ng

Vol. 19 No. 2, December, 2012

ISSN: 2006-5523

AN INTERNATIONAL JOURNAL OF THE NIGERIA COMPUTER SOCIETY (NCS)

Title and Authors | Page

Editor-in-Chief: DR. A. S. SODIYA

THE JOURNAL OF COMPUTER SCIENCE AND ITS APPLICATIONS
AN INTERNATIONAL JOURNAL OF THE NIGERIA COMPUTER SOCIETY (NCS)
Vol. 19 No.2, December, 2012

nigeria computer society
ncs
www.ncs.org.ng.

Ensuring QoS with Adaptive Frame Rate and Feedback Control Mechanism in Video Streaming

Olufade F.W. ONIFADE,[1] and Adebola, K. OJO [2]

[1 2]Department of Computer Science
University of Ibadan
Ibadan, Nigeria

[1]fadowilly@yahoo.com
[2]adebola_ojo@yahoo.co.uk

ABSTRACT

Video over best-effort packet networks is cumbered by a number of factors including unknown and time-varying bandwidth, delay and losses, as well as many additional issues such as how to fairly share the network resources amongst many flows and how to efficiently perform one-to-many communication for popular content. This research investigates video streaming formats, encoding and compression techniques towards the development and simulation of a rate adaptation model to reduce packet loss. The thrust of this research aimed at enriching and enhancing the quality of video streaming over the wireless network. We developed both mathematical models which were thereafter simulated to depict the need for advancing the existing solution for packet scheduling towards recovery from packet loss and error handling in video streaming.

(Keywords - Quantity of Service (QOS), Adaptive Frame Rate, Video Streaming)

1.0 INTRODUCTION

Streaming media technology is an integrated part of Internet media service providers such as Yahoo, MSN, AOL, YouTube and many others [1]. Video is becoming the most important media for communication and entertainment all over the world. Initially video was captured and transmitted in analog form. The advancement in integrated circuits and the miniaturization of computers with the advent of Internet have led to the continuous digitalization of video, leading to the possibility of transporting digital video in compressed format to facilitate ease of transportation and less stress on available bandwidth.

Video compression became an important area of research in the late 1980's and 1990's [2] thus enabling variety 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), and video conferencing and videophone over circuit-switched networks. The growth and popularity of the Internet in the mid-1990 motivated video communication over best-effort packet networks. Video over best-effort packet networks is cumbered by a number of factors including unknown and time-varying bandwidth, delay and losses, as well as many additional issues such as how to fairly share the network resources amongst many flows and how to efficiently perform one-to-many communication for popular content.

Most popular Internet applications are web-based audio and video playback, where stored video is streamed from the server to a client on-demand. Rigid playback deadlines coupled with constraint on resources such as network bandwidth and client buffer make video delivery a challenging task [3].Video streaming known to be one of the many exciting applications on the Internet [4]. It already created a new business known as internet broadcast, or intercast/webcast, although Internet broadcast is still advancing; there is a glimpse of the future of broadcast but also on-demand and personalized programs. Real-time transport of live video or stored video is the predominant part of real-time multimedia. In this research, we are concerned with only video streaming, which refer to real-time transmission of stored video on the wireless network. There are basically two modes for transmission of stored video over wireless network or the Internet, namely the download mode of transmission and the streaming mode (i.e., video streaming).

In the download mode, users download the entire video file and then play back the video file. Full file transfer in the download mode usually suffers long and perhaps unacceptable transfer time. In contrast, in the streaming mode, the video content need not be downloaded in full, but is being played out while parts of the content are being received and decoded. Due to its real-time nature, video streaming typically has bandwidth, delay and loss requirements. The above mentioned is a typical occurrence whenever attempt is made to view YouTube video or watch popular sites like the CNN. The effect of the inadequate bandwidth is more pronounced for bursty traffic like video where part-download is performed to allow for continuous playback.

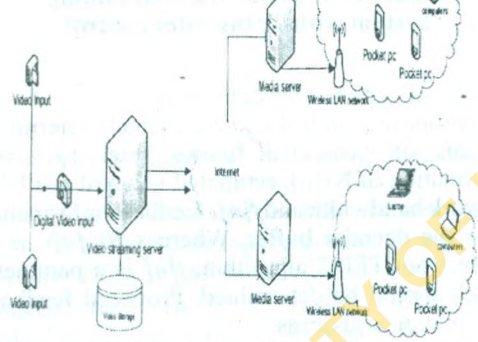

Figure 1: Hierarchical Architecture for Wireless Video Streaming

Video files in video streaming are transmitted by a media server application as shown in figure 1, and is processed and played back by a client player application, as it is received. A client application, known as player, can start playing back streaming media as soon as enough data has been received without having to wait for the entire file to have arrived. As data is transferred, it is temporarily stored in a buffer until enough data has accumulated to be properly assembled into the next sequence of the media stream. When streaming technology was first available, the ability to begin playback before the entire file had been transferred was a distinct advantage. Now, however, pseudo-streaming techniques, such as progressive download, allow some other formats to begin to play before file download is completed, while the ability to begin playback prior to completing file transfer is a characteristic of streaming, it is not, in and of itself, a differentiating factor. [5] said a streamed file is received, processed, and played simultaneously and immediately, leaving behind no residual copy of the content on the receiving device. An important advantage of streaming media (unlike either traditional or progressive download) technology is the copyright protection it provides. No copy of the content is left on the receiving device. Therefore, the recipient can neither alter nor redistribute the content in an unauthorized manner.

This research investigates video streaming formats, encoding and compression techniques, and simulate a rate adaptation model to reduce packet loss. The focus of this research is a bid to enhance the quality of video streaming over the wireless network by advancing the existing solution for packet scheduling to recover packet loss and error handling on video streaming to improve and produce video quality. In the rest of this paper, section 2 review of related literature and insight into video streaming and TCP friendly rate control (TFRC) mechanism. Section 3 presents both the framework and the mathematical model which were simulated and discussed in section 4. We conclude the work in section 5 and also give our future direction.

2.0 REVIEW OF RELATED WORKS

Video streaming typically has bandwidth, delay and loss requirements. However the current best effort Internet does not offer any quality of service (QoS) guarantee to streaming video over the Internet or wireless network as stated by [6]. There have been several selective dropping schemes proposed by [7], [8], [9], [10], [8] and [11]. [9] stated the problem of streaming packetized media in a rate distortion optimized way and gives a rigorous analysis of it. The scheduling algorithm decides which packets will be transmitted under the rate constraint while minimize the end-to-end distortion. [12] modeled the streaming system as a queuing system. An optimal sub-stream is selected based on the decoding failure probability of the frame and the effective network bandwidth. A probability dropping mechanism is proposed by [10] to calculate the dropping probability for each layer. [8] presented a streaming framework centered on the concept of priority drop. It combines the scalable compression and adaptive streaming to provide a graceful degradation of the quality. [13] also presents an end-to-end based approach to facilitate streaming over wireless which was discussed in [14]. Their approach is based on two observations; first, relative one way delay increases monotonically if there is congestion; second, inter-arrival time is expected to increase if there is packet loss caused by wireless channel errors. Extensive researches conducted by many researchers aim at the dissemination of different areas of contributions into the field of streaming media (video streaming) over the internet.

A. TCP Friendly Rate Control (TFRC)

A widely popular rate control scheme over networks is equation-based rate control, also known as TCP friendly rate control (TFRC) [15]. There are basically three main advantages for rate control using TFRC: firstly, it does not cause network instability, which means that congestion collapse is avoided. More specifically, TFRC mechanism monitors the status of the network and every time that congestion is detected, it adjusts the sending rates of the streaming applications. Secondly, it is as regards the TCP flows, which are the dominant source of traffic on the Internet. Thirdly, the TFRC's rate fluctuation is lower than TCP, making it more appropriate for streaming applications which require constant video quality. A widely popular model for TFRC [15] is described as

$$T=kS/(RTT\sqrt{p}) \qquad (1)$$

Where T is the sending rate, S is the packet size, RTT is the end-to-end round trip time, p is the end-to-end packet loss rate, and k is a constant factor between 0.7 and 1.3, depending on the particular derivation of (1).

The use of TFRC for Streaming video over wireless network surpasses most protocols for streaming. It resolves two TFRC difficulties: firstly, TFRC assumes that packet loss in wired networks is primarily due to congestion, and as such it is not applicable to wireless networks in which the bulk of packet loss is due to error at the physical layer. Secondly, TFRC does not fully utilize the network channel. It resolves it by multiple TRFC connections that results in full utilization of the network channel.

3.0 PROPOSED SYSTEM MODEL

Streaming media is known to be faced with packet loss, delay jitter and bandwidth problem and these reduces the video quality and irritate playback for the viewer or user. For example, any late frame resulting from delay jitter can produce problems in reconstructed video, accurately estimating the available bandwidth to match the pre-encoded video to the estimated channel bandwidth is another problem, packet loss can lead to total erasure of a particular frame and when Buffer constraint is introduced into the system there will be no guarantee that the encoder and decoder buffer will not overflow or underflow. Wireless video system with a transcoder control and feedback is illustrated in Figure 2 below.

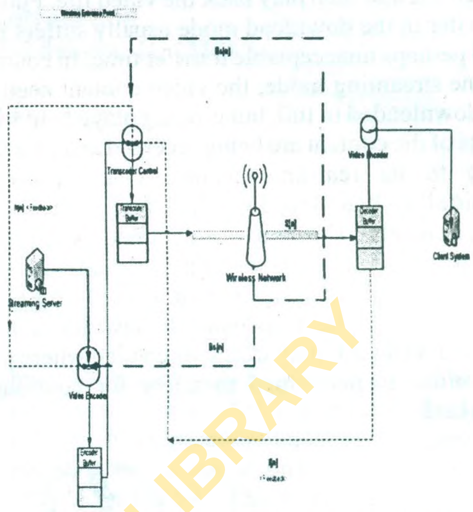

Figure 2: Wireless video streaming system with Transcoder control

A. Feedback Information

The proposed control equation which determines bitrates of generated frames refer to status information of Ns[n], estimated value of available network bandwidth and *f[n]*, feedback information from the decoder buffer. Whereas *Ns [n]* is as given from TFRC algorithm, *f[n]* is a parameter which should be determined. Proposed feedback information is given as:

This equation depicts that the feedback information *f[n]*is an error term between the current backlog of the decoder buffer $DQ_{back}[n]$ and the current target value of the decoder buffer $DQ_{target}[n]$. Actually, $DQ_{back}[n]$ can be another candidate for feedback information. However, adopting an error term as Feedback information allowing feedback packets of smaller size. The streaming server should maintain $DQ_{target}[n]$ when $DQ_{back}[n]$ is used as feedback information. Calculating $DQ_{target}[n]$ and maintaining it is made easier at the playback client than at the streaming server. If the value of $DQ_{target}[n]$ is not too big to cause overflow of the decoder buffer and not too small to cause underflow of the decoder buffer, this might be very good. Knowing how much amount of video in the decoder buffer that does guarantee a good knowledge of how long the video is playable. It is equally expedient to note that every frame has different size and different bitrate to each other. Controlling the Frame rate *Fr[n]* will maintain the playable time of the video in the decoder buffer at the target value rather than to control *Fr[n]* just for constant backlog. Estimated playable time *Pt [n]* of the current video in the decoder buffer and this can be represented thus:

$$Pt[n] = \frac{DQtarget[1]}{Fr[1]} + \frac{DQtarget[2]}{Fr[2]} + \ldots + \frac{DQtarget[N]}{Fr[N]} = \sum_{n=1}^{N} \frac{DQtarget[n]}{Fr[n]} \quad (3)$$

We can divide the video in the decoder buffer into several classes each of which has a different bitrate as shown in figure 3.

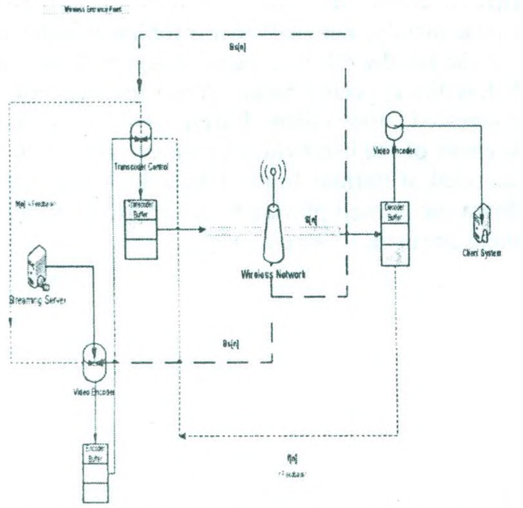

Figure 3: Model of the Decoder buffer

Suppose that there are N such classes in the decoder buffer. Knowing the bitrate *Fr[n]* and the amount $DQ_{targ\ d}[n]$ of each *N* class, total playable time of the video in the decoder buffer can be calculated compared to (3). However, it is not easy to know bitrate of each of video classes and so some approximation will be needed. Intuitively, sending rate *Sr[n]* can be an approximation of *Fr[n]*. Thus, we can safely assume that each sending rate of different classes in the decoder buffer may be similar and thus represented as follows:

If we can determine the total amount of the video in the decoder buffer and the sending of the latest packet, we can approximately calculate playable time of the video in the decoder buffer giving rough approximation. Since, the knowledge of the exact playable time is not of utmost importance, we only need feedback information which reflects time-based information of the decoder buffer, not just backlog-based information. Maintaining playable time of the video in the decoder buffer at the initial playback delay IP_d, implies that every instance of playable time should be equal to IP_d.

Cross multiplying (5) with IP_d should have the following form.

Furthermore, target value $DQ_{targ\ d}[n]$ should not be greater than the size of the decoder buffer (DB).

Feedback information of the proposed streaming system gotten from (2)

$f[n]=DQ_{b\ ok}[n] - DQ_{targ\ d}[n]$ now substituting (7) we have the following

4.0 SIMULATION AND DISCUSSION OF RESULT

Sequel to the mathematical model developed in section 3, we need to simulate this result in a bid to ascertain the efficacy and establish our research findings. Consequently, the model was developed in Omnet++.

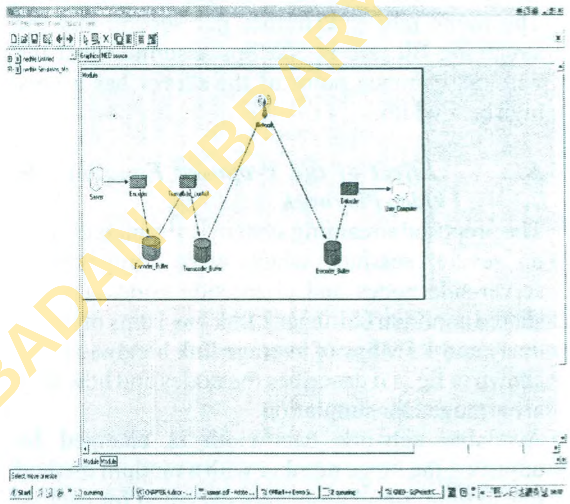

Figure 4: Node Arrangements for the Simulation (using OMnet ++)

A. Parameters definition

In this research, we identified four main parameters amongst others: Average sending rate, Buffer capacity, Feedback interval, and Frame rate. Although there are others like the propagation delay and many more which can be assumed and calculated during the simulation. Table 1 below describes the four main parameters used for the simulation

Table 1: SIMULATION PARAMETERS

Parameter	Value
Average Link Bandwidth	1.30Mbps
Buffer Capacity	1000Kbyte
Feedback Interval	5/10 seconds
Frame rate	15.00fps
Data packet size	1000bytes
Startup Delay	10ms
Streaming Bit rate	50Mbps
End-to-End Delay	100ms

Each module can have different parameters. The parameters used are numeric which includes expressions using other parameters and calling different functions, random variable from different distributions where used for the bandwidth and the initial delay and buffering time with the processing time that will be initialized by the user. The time was measured in milliseconds (ms) during the simulation and in seconds when plotting the graph. Frame rate is 15fps. Generally video frame rate of between 10fps and 20fps compared with that for non-real-time playback. The frame rate is 15frames per second and the streaming Bit rate is 50Mbps assuming that the wireless entrance point of the server has a very high bandwidth.

A. *Effect of the Proposed Equation On Video Playback*

The proposed streaming system is simulated based on several sessions which were composed of server-side nodes and client-side nodes and the shared common bottleneck link has 10ms of initial delay and 1.5Mbps of average link bandwidth. As shown in fig. 4 it describes the nodes and how they are arranged for simulation.

Available network bandwidth is changed by adjusting the range number with a random method due to time, simulation were executed for each of fixed network bandwidth, variable network bandwidth, and gradually fluctuating the bandwidth. Feedback information to affect the change of the frame rate is sent every 15 seconds. With this, the bandwidth cannot maintain a fixed bandwidth during the simulation, it maintained between 1000000bytes/sec and 1300000bytes/sec. Note that the throughput has wide range of variation because TFRC algorithm which calculates available bandwidth cannot guarantee constant value, and the playable time is maintained at constant rate. Also note that if the bit rate of Fine Granularity Scalability (FGS) video does not change, playable time and decoder buffer exhibited similar behavior.

In evaluating the effectiveness of the rate adaptation transcoder with Feedback information, it behooves that the transcoder will transcode the incoming video and the previous feedback information will be taken into consideration by checking if either to increase the frame rate or to reduce the frame rate of the frame transmitted to the decoder buffer. The proposed equation for frame rate is used in the transcoder control and based on the feedback information provided, the transcoder generate appropriate frame rate that will concur to the bandwidth's strength.

A graphical representation of the simulated result is given in figure 5. The playback rate of the video streamed across the network against the time used to stream the video, knowing fully-well that the bandwidth varies with time, in this result the only thing that was removed was the feedback information to the transcoder control. Consequently, there will be no problem whether or not the bandwidth is consistent and will not fall below the expected range. When the network is congested, the client buffer will underflow because of the bandwidth limitation, the video is decoded at normal frame rate, which will slow down the normal playback rate and there will be breakage in the video playback.

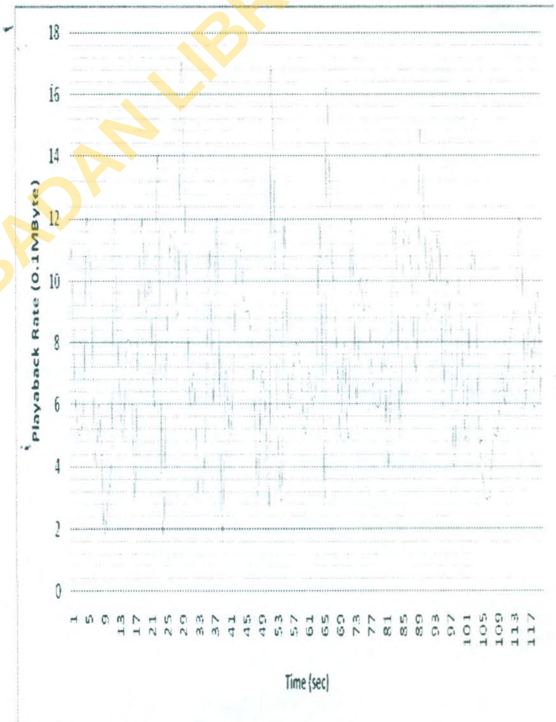

Figure 5: Result of Simulated Transcoder System without Feedback Information

Considering the Feedback Information with the Transcoder control (Fig. 6 represent the result of the simulation) the proposed equation will be triggered based on the bandwidth and other information necessary to calculate the frame rate. The new frame rate will work according to the network bandwidth and the drain rate will be proportional to the frame rate.

The buffer overflow and underflow was considered and during the simulation buffer underflow was greatly reduced, although we assumed that the buffer overflow at the client side (Decoder Buffer) depends on the memory capacity of the client system.

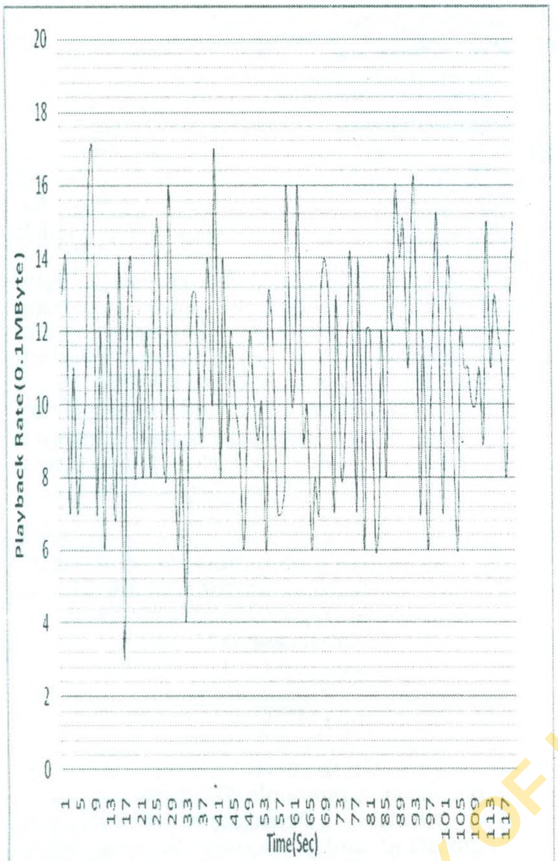

Figure 6: Result of Simulated Transcoder System with Feedback Information

In a bid to have a vivid demonstration of the discussions hitherto, we present interleaved result of the Transcoder Control with Feedback information and without Feedback information, comparing the effectiveness of the Feedback Information on Transcoder control. There is great different between the two systems, since the feedback information is sent back to the transcoder giving the requirement of the size and rate of frame to send the next Drain rate did not decrease below a certain manageable level on the playable rate axis (this is designated during the simulation experiment as '6'). It thus became apparent that the inclusion of the feedback information has increased the reliability of transmission and added value of the streamed video.

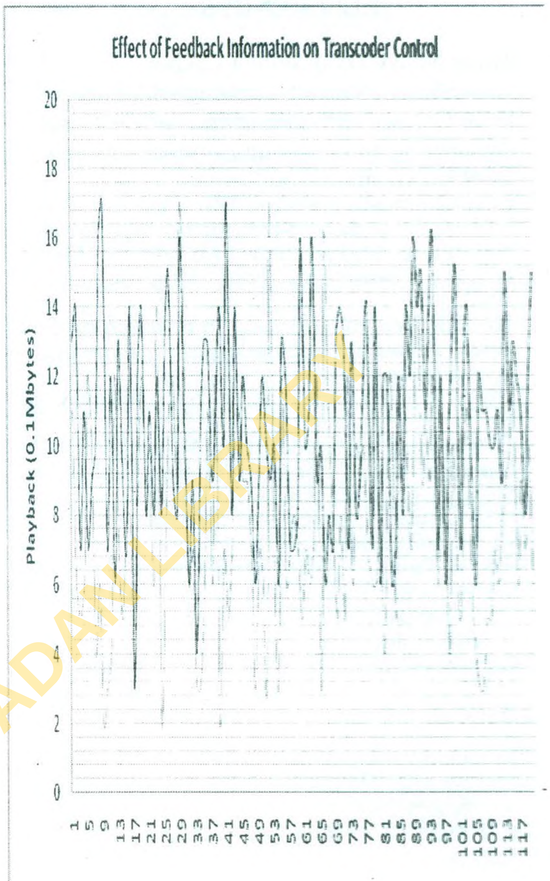

Figure 7: Comparative analysis of both results.

V. CONCLUSION AND FUTURE DIRECTION

This research has investigated video streaming formats, encoding and compression techniques, and simulate a rate adaptation model aimed at reducing the packet loss resulting from the fluctuation in network bandwidth. We developed both mathematical and conceptual framework for the proposed model and our research result showed clearly that our model fared well with the introduction of the feedback control mechanism. In the future, we hope to demonstrate the effect of buffering on the overall performance of the system.

REFERENCES

1. Thinh Nguyen and Avideh Zakhor "Distribution Video Streaming with Forward Error Correction " Department of Electrical engineering and Computer sciences, March 1999 University of California.
2. John G. Apostolopoulos, Wai - tai Tan, Susie J.Wee " Video Streaming: Concepts, Algorithms, and Systems" Mobile and Media systems laboratories Palo Alto, September 2002.
3. Jaromil, "Video streaming: a research on free video streaming technologies " dyne.org / rastasoft.org http://frost.htu.tuwien.ac.at/~roman/nuppelvideo
4. G. Conklin, G. Greenbaum, K. Lillevold, A. Lippman, and Y. Reznik, "Video Coding for Streaming Media Delivery on the Internet," IEEE Trans. Circuits and Systems for Video Technology, March 2001.
5. Serhat Tural " Video streaming for wireless messaging " Dokuz Eylul university, department of computer Engineering 2007.
6. J. Apostolopoulos and S. Wee, ``Video Compression Standards", Wiley Encyclopedia of Electrical and Electronics Engineering, John Wiley & Sons, Inc., New York, 1999.
7. S.H. Kang and A. Zakhor, "Effective Bandwidth based Scheduling for Streaming Multimedia," IEEE ICIP'03,Barcelona, Spain, September 2003.
8. C. Krasic, J. Walpole, and W. Feng, "Quality-Adaptive Media Streaming by. Priority Drop," NOSSDAV 2003, June 2003.
9. P.A. Chou and Z. Miao, "Rate-distortion Optimized Streaming of Packetized Media," Technical Report, MSR-TR-2001-35, Microsoft Research, February 2001.
10. I.V. Bajic, O. Tickoo, A.S. Kalyanaraman, and J.W. Woods, "Integrated End-to-End Buffer Management and Congestion Control for Scalable Video Communications," IEEE ICIP'03,Barcelona, Spain, September 2003.
11. Zhijun Lei, and Nicolas D. Georganas, "Rate adaptation transcoding for video streaming over wireless channels" Distribution and collaborative virtual environment research laboratory, University of Ottawa 2000.
12. Song Chong, Sangho Lee, and Sungho Kang, "A simple, Saclable Explicit rate allocation algorithm for MAX-MIN Flow Control with Minimum rate guarantee," IEEE/ACM Trans . on Networking, vol 9, pp 320-335, jun 2001.
13. Cen Yao, "Real-time Internet video using error resilient scalable compression and TCP-friendly transport protocol," IEEE Trans. Multimedia, vol.1, pp. 172186, June 1999.
14. S. Cen, P.C. Cosman, and G.M. Voelker, " End-to-end differentiation of congestion and wireless losses", Proc. Multimedia Computing and Networking (MMCN) conf. 2002, pp. 1-15, San Jose, CA, Jan 23-25, 2002
15. Minghua Chen and Avideh Zakhor, "Transmission protocols for streaming video over wireless", Department of Electrical Engineering and Computer Science 2001, Univerisity of California. Pg 2-4