An Extensive Study over Vbr Image Processing Algorithm Using Headlight Prefetching and Mash Video Sensing Techniques

[a]T.S.Venkateswaran, [a]B.Sowmya, [a]R.Arun
[a]Assistant Professor in Computer Science,K.S.R College of Arts and Science (Autonomous)Tiruchengode, Tamilnadu, India.

Abstract

Today Internet is used everywhere. Video data is downloaded or watched online more among the internet users. The video is streamed which is called as Video Streaming has many techniques to stream the data. Some of the techniques are Variable Bit Rate (VBR), Constant Bit Rate (CBR) etc. Techniques like Headlight Prefetching and some of the protocols and topologies are used for video streaming.

In this research study, the video streaming technique VBR has some disadvantages over CBR i.e. VBR takes more time to encode as the process is more complex. The propagation delay in VBR is greater than CBR. The study is how to reduce the propagation delay in VBR. The propagation delay is the transmission delay (store and forward delay).

VBR Image Processing Algorithm which already exists with Headlight Prefetching Techniques and Mash Video Sensing Techniques are integrated.

Mash Sensing Technique which uses Mash Sensing Video (MSV) can be controlled by using the sampling rate. The quality of the image transmitted plays a vital role over the transmission of the video data. The video encoder is developed based on Mash sensing to record the propagation delay occurs.

The main research question is answered in this paper is how to reduce the propagation delay in VBR and what techniques can be derived or integrated to achieve the method to reduce the delay.

KEYWORDS: Video Streaming, Propagation Delay, VBR, CBR, Mash Sensing Techniques.

Introduction

Today Internet has advanced mechanisms for internet services which encourages the development of good performing networks and high demand for multimedia applications. Wireless technologies provide better performance.

In wireless networks, Quality of Service (QoS) assurance, service portability and application persistence is the key factor for good performing networks. The network communication has to be monitored. The monitoring process has an approach which focuses on the transmission of variable bit rate (VBR). The performance report shows today's internet traffic and grow day by day.

The user has experienced video streaming has been affected by many factors. Before a video is displayed, the video passes many processes between capturing the video and decoding of that video signal. The following processes include capturing, encoding, transmitting, decoding, and displaying of video images. During this steps Quality loss may occur. The general types of causes that affect video quality are Quality of Delivery

(QoD) and Quality of Presentation (QoP). The QoD plays a vital role in the wireless network video communication or transmission.

The robust statistical methods are used to detect the influence of QoS conditions on QoD of variable bit rate.

Research Questions

The research questions addressed to find the set of descriptive indicators that govern QoS in a network with focus on QoD and media buffer of streaming wireless protocols.

- What are the QoS metrics essential in the study of QoD?
- Can QoS indicators can be found?
- WLAN and W-CDMA show different behavior to outlying of QoS data.
- Do the suggested techniques deployable in a real time environment with limited resources?

Literature Review

In the field of networking, multidimensional QoS analysis has not yet received much attention with regards to quality purposes. Although, multidimensional reduction techniques are frequently applied to network related problems, linear Principle Component Analysis (PCA) seems to be a popular choice. The Mahalanobis distance is especially used in contexts where reliability plays an important role and cross-correlation of variables is of importance.

Ramirez-Velarde et al. suggested that PCA can help in reducing dimensionality in computer related modeling of, e.g., user preferences and behavior, as well as resource popularity.

A series of papers elaborate on anomaly detection in networks to identify security breaches with the help of network level QoS. Lakhina et al andHakami et al. expound on diagnosing anomalies in traffic flows in fixed networks and wireless networks, respectively.

Yoshihiro et al. suggests a mapping between application-level QoS and user-level QoS. The authors measured nine metrics for media synchronization quality for audio-video transmission and reduced its dimensions first with PCA. A Multiple Regression Analysis (MRA) was applied to the reduced dimensions to deduce a relationship with Thurstone’s law of comparative judgment. The authors report that they could accurately estimate the ”user-level QoS” via
their MRA.

Schneps-Schneppe et al. [28] studied the design issues of Service Level Agreement (SLA) agreements in 3G networks.Fiems et al. [29] developed an analytical model of the output buffer of an on-demand video server using a queueing model with dependent transmission times. Using probability generating functions, the performance was measured with regardsto packet loss ratio and average frame transmission time. Idle and busy periods of the buffer were the main focus of their work. Fiems et al. showed that a reduction of transmission times can lead to lower values of the frame loss ratio.

Liang and Liangdeveloped an analytical model of jitter buffer exhaustion based on a Markov Variable Bit Rate (VBR) channel model. The model was studied under different buffer sizes at the receiver, initial start-up delays and video freeze recovery schemes. Simulations were conducted over wireless links with the extended Gilbert loss model. The authors claim that their analytical model quantifies the tradeoffs between the start-up delay, buffer size, and video freeze frequency.

The referred articles approach the QoD from a theoretical point of view. In their frameworks simplified models of data transmission over the Internet are used. On the other hand, we are studying the notion of QoD from a more practical point of view. The goal is to identify problematic network conditions in the context of QoD in contrast to theoretical bottlenecks. Real-world experiments are used not only to capture the behavior of wireless technologies but also to identify QoS metrics that are useful in the definition of QoD.

Paper Outline

The paper is concerned over video streaming and the notion of the jitter buffer. The robust statistical methods are introduced. The next concept in the study is about the QoS metrics that are used during the analysis. Next concept in the study is to explain the methods used to analyze the VBR Image Processing. VBR achieves better quality and compression rates than fixed bit rate. In the next area of study, experimental method is used to analyze the QoS measurement for new applications which has video surveillance, storage and subsequent retrieval. The short comings in the system are encode complexity and low resiliency to channel errors. At the last concept of the paper, the complete proposed work and its conclusions are listed out.

Background of the study

In the background of the study, the overview of video streaming and Real-Time Transport Protocol (RTP) is studied. Video on Demand (VoD) are very popular over the internet users. Video streaming used in applications such as remote surveillance, distance learning etc.

The Internet Protocol in current is used for video streaming. This IP manages internet traffic. The 3GPP and RTP is the main component for the transport media streams. The Real Time Transport Control Protocol (RTCP) is used with RTP is the active media players and streaming server.

Darwin Streaming Server (DSS), Quick Time Streaming Server (QTSS) are some of video streaming servers. User Diagram Protocol (UDP) with Transfer Control Protocol (TCP) features retransmission and congestion control such as over buffering.

The video encode provides a video at constant or variable bit rate. The Constant Bit Rate (CBR) video is fixed over time whereas Variable Bit Rate (VBR) video exhibits a fluctuating bit rate.Clumping (CL) is an effect that results in the alteration of the inter-arrival time of packets traversing a network. CL effects are mostly visible in packet streams.

The QoS (Quality of Service) is used to measure how well the server performs. In video transmission QoS is used as the quality quantification of a communication channel in terms of measurable parameter in a computer network. The QoS metrics include Latency, Delay Variation, Reordering, Errors, Bandwidth and Packet Loss.

VBR Image Processing

Through VBR technique better quality and compression rates are achieved by Random Access and On-The-Fly decompression which allowed by DXT method which uses lossy fixed bit rate of GPU Texturing, less texture bandwidth DXT was developed. For GPU, a new variable bit rate lossy texture compression algorithm is stated.

The higher compression rate is used in VBR. VBR texture compression algorithm is introduced for fast GPU decompression. The related works are stated below.

a) Compression/Entropy Coding:

The data has to be compressed with the use of various techniques. The compression has some problems as i) modeling the probabilities for input symbols and using those probabilities to construct a compressed data stream.

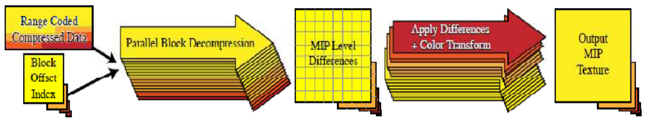


Figure 1: Overview of GPU decompression algorithm

b) Image Compression:

To compress a single image, most image compression and decompression is designed to run a CPU. But in VBR, a better compression quality and/or rate can be attained. The compressed data stream is created by encoding the quantized values losslessly.

c) GPU Texture Compression:

Fixed-Bit ratelossy methods have focused by GPU texture compression. This allows sending directly from a compressed rate. The method decompresses Variable Bit Rate Textures using standard GPU computing.

d) GPU Compression/Decompression:

The work has been stated to date on direct compression or decompression on the GPU. The lossless compression with average compression rate for RGBA textures and it is delivered by a VBR image compression algorithm uses fixed bit rate lossy compression algorithms and lossy compressions rate averaging better than CBR.

VBR Compression Algorithm

Store any MIP levels smaller than 2x2 as raw colors
Convert to appropriate color space
For each MIP level from coarsest to finest
Compute difference with previous level
Truncate bits
Compute probabilities for all levels < block size
Compress levels below the block size together
Flush compression stream
For each MIP level
For each block
Write starting position to index texture
Compute static probabilities
Encode range of b its and probability table
For each 2x2 neighborhood

For each bit
Compute classes and encode bits
Flush compression stream

Figure 2: Pseudo-code for full VBR compression algorithm

VBR Decompression Algorithm
Decode MIP levels smaller than 2x2 as raw colors
Read stats for levels below block size and decompress
For each block in index in parallel
Decode bit range and probability table
For each 2x2 neighborhood
For each bit
Compute classes and decode bits
Write 2x2 neighborhood to difference texture
For each MIP level from coarsest to finest
For each pixel in parallel
Apply MIP differences and in verse color transform

Figure 3: Pseudo-code for VBR decompression algorithm

Wireless Multimedia Sensor Networks

Wireless Multimedia Sensor Networks (WMSN) are self-organizing systems of embedded devices deployed to retrieve, distributive process in real-time, store, correlate, and fuse multimedia streams originated from heterogeneous sources. WMSNs are enablers for new applications including video surveillance, storage and subsequent retrieval of potentially relevant activities, and person locator services. In recent years, there has been intense research and considerable progress in solving numerous wireless sensor networking challenges. However, the key problem of enabling real-time quality-aware video streaming in large-scale multi hop wireless networks of embedded devices is still open and largely unexplored. There are two key shortcomings in systems based on sending predicatively encoded video through a layered wireless communication protocol stack, i.e., *encoder complexity* and *low resiliency to channel errors*.

• **Encoder Complexity:**Predictive encoding requires complex processing algorithms, which lead to high energy consumption. New video encoding paradigms are therefore needed to reverse the traditional balance of complex encoder and simple decoder, which is unsuited for embedded video sensors. Recently developed *distributed video coding* algorithms exploit the source statistics at the decoder, thus shifting the complexity to the decoder. While promising, most practical Wyner-Ziv codecs require end-to-end feedback from the decoder, which introduces additional overhead and delay. Furthermore, gains demonstrated by practical distributed video codes are limited to 2- 5 dBs PSNR. Distributed video encoders that do not require end-to-end feedback have been recently proposed, but at the expense of a further reduction in performance. In addition, all of these techniques require that the encoder has access to the *entire video frame* (or even multiple frames) before encoding the video.

Proposed System

With the proposed controller, nodes adapt the *rate of change* of their transmitted video quality based on an estimate of the impact that a change in the transmission rate will have on the received video quality. While the proposed method is general, it works particularly well for security videos. In addition, all of these techniques require that the encoder has access to the *entire video frame* (or even multiple frames) before encoding the video.

Advantages

The proposed CSV encoder is designed to: i) encode video at low complexity for the encoder; ii) take advantage of the temporal correlation between frames.

Implementation

Implementation is the stage of the project when the theoretical design is turned out into a working system. Thus it can be considered to be the most critical stage in achieving a successful new system and in giving the user, confidence that the new system will work and be effective.

The implementation stage involves careful planning, investigation of the existing system and it's constraints on implementation, designing of methods to achieve changeover and evaluation of changeover methods.

Modules

1. **CS Video Encoder (CSV)**

The CSV video encoder uses compressed sensing to encode video by exploiting the spatial and temporal redundancy within the individual frames and between adjacent frames, respectively.

Sensing the channel**:** those that have the cost of sensing channel have higher energy consumption and so they are not suitable for WMSNs.

Using extra packets:Using retransmission time of dropped packets includes not only retransmission request but also transmission of dropped packet. These methods waste a great amount of energy for congestion detection in sensor nodes.

Low cost*:* Some methods do not necessitate extra cost for congestion detection. These methods are the most suitable for congestion detection in WMSNs.

2. **Rate Change Aggressiveness Based on Video Quality:**

With the proposed controller, nodes adapt the *rate of change* of their transmitted video quality based on an estimate of the impact that a change in the transmission rate will have on the received video quality. The rate controller Uses the information about the estimated received video quality *directly* in the rate control decision. If the sending node estimates that the received video quality is high, and round trip time measurements indicate that current network congestion condition would allow a rate increase, the node will increase the rate less aggressively than a node estimating lower video quality and the same round trip time. Conversely, if a node is sending low quality video, it will gracefully decrease its data rate, even if the RT T indicates a congested network. This is obtained by basing the rate control decision on the *marginal distortion factor*, i.e., a measure of the effect of a rate change on video distortion.

3. **Video Transmission Using Compressed Sensing:**

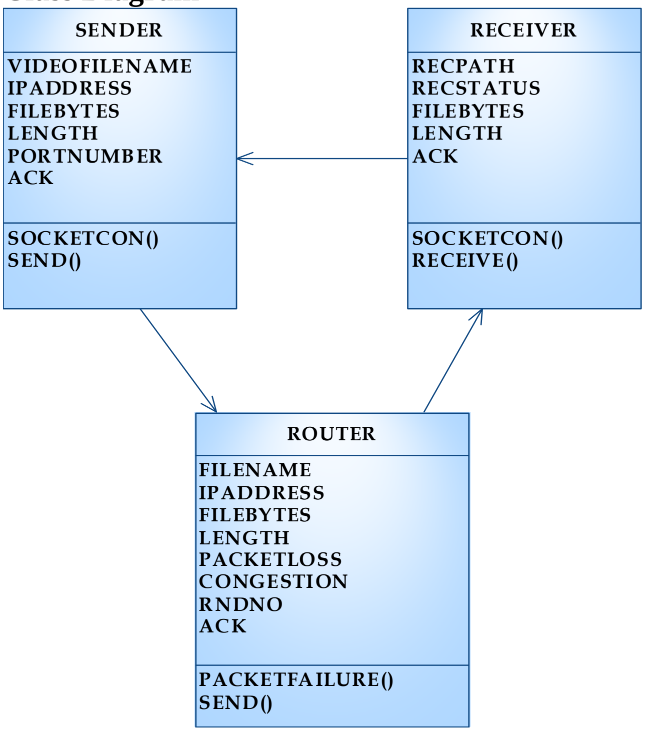
We develop a video encoder based on compressed sensing. We show that, by using the difference between the CS Samples of two frames, we can capture and compress the frames based on the temporal correlation at low complexity without using motion vectors.

4. **Adaptive Parity-Based Transmission:**

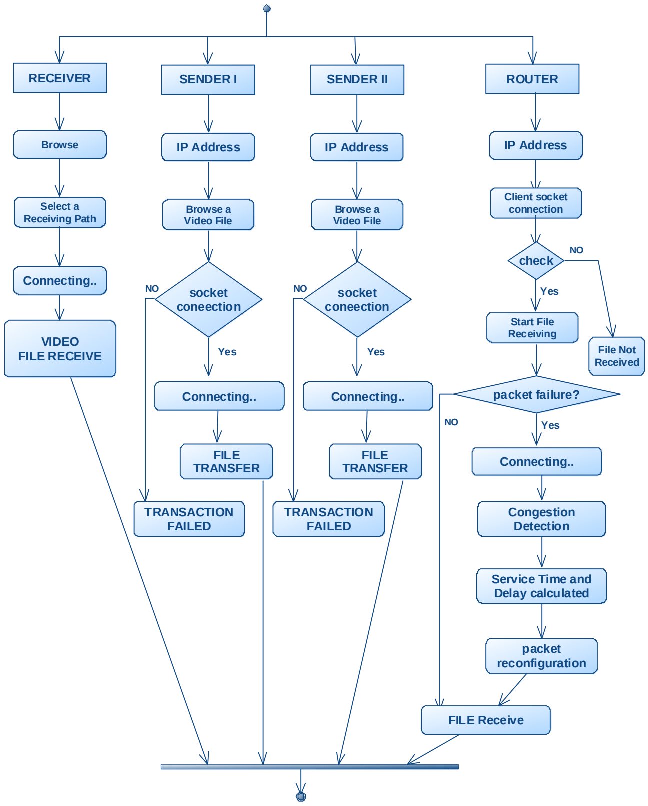
For a fixed number of bits per frame, the perceptual quality of video streams can be further improved by dropping error samples that would contribute to image reconstruction with incorrect information. Which shows the reconstructed image quality both with and without including samples containing errors? It assume that the receiver knows which samples have errors, they demonstrate that there is a very large possible gain in received image quality if those samples containing errors can be removed.We studied adaptive parity with compressed sensing for image transmission, where we showed that since the transmitted samples constitute an unstructured, random, incoherent combination of the original image pixels, in CS, unlike traditional wireless imaging systems, no individual sample is more important for image reconstruction than any other sample. Instead, the number of correctly received samples is the only main factor in determining the quality of the received image.

Data Flow Diagram

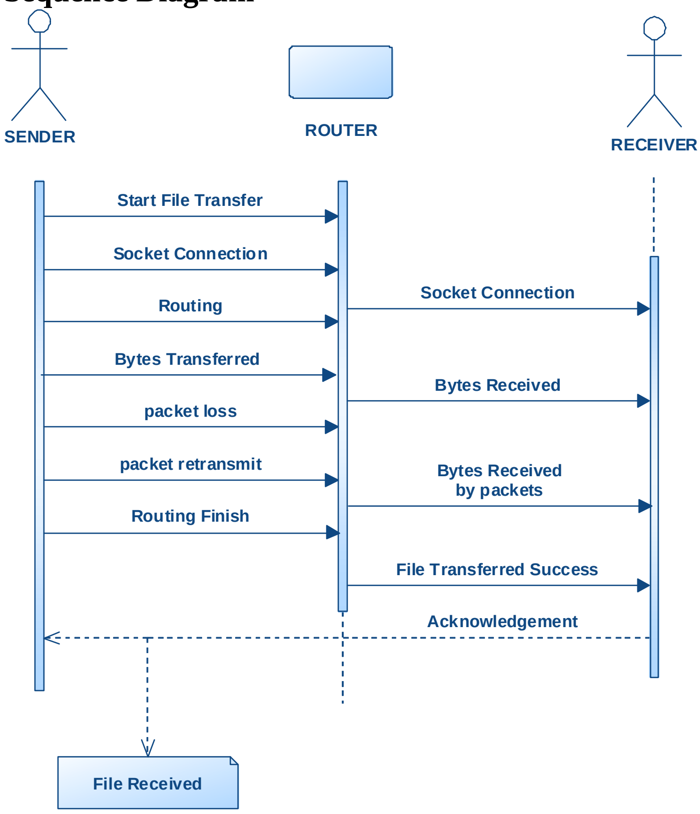
Class Diagram



Activity Diagram



Sequence Diagram



Mash Sensing

Mashed Sensing (MS) is a technique that can overcome many problems like encoder complexity and low resiliency to channel error regarding to video over WMSNs. The rate of Mashed Sensed Video (MSV) can be control by varying the Mashed sensing sampling rate. MS can offer an alternative to traditional video encoders by enabling imaging systems that sense and Mash data simultaneously at very low computational complexity for the encoder. In conventional digital image or video capturing the large amount of raw data acquired which need immediate Mashing in order to store or transmit. This process has two main disadvantages.

First, acquiring large amount of raw image or video data can be expensive particularly at wavelength where CMOS/CCD technology is limited.

Second Mashing raw data can be computationally complex in case of video. The process of sample, process, keep important information, and throw away the rest, is known as Mashed Sensing (MS).

Mash Sensing (MS) is able to processed many signals that could be difficult to capture or encode using conventional methods. From a relatively small number of random measurements, a high-dimensional signal can be recovered. Standard video capture systems require a complete set of samples to be obtained for each frame, at which point a Mashing algorithm may be applied. In some applications, such as imaging at non-visible (e.g. infrared) wavelengths, it may be difficult or expensive to obtain these raw samples. In other applications, such as multi-image capture in camera networks, implementing a Mashing algorithm may in other applications, such as multi-image capture in camera networks, implementing a Mashing algorithm may itself be a challenge. These burdens may be reduced by using Mash imaging hardware such as the single-pixel camera where random measurements are collected independently from each frame and no additional Mashing protocol is needed.

- **Rate Change Based on Video Quality**

The system is based on this MS architecture, where transmission rate is depend on the received video quality. This means that the rate controller uses the information about the estimated received video quality directly in the rate control decision. If the sending node found that the received video quality is very high, it will be decrease to increase the rate. If a node is sending poor-quality video, it will decrease its data rate, to improve quality of video.

- **Video Transmission Using Mash Sampling-**

The video encoder can be developed based on Mashed sensing. By using the difference between the CS samples of two frames, we can capture and Mash the frames based on the temporal correlation at low complexity.

Experimental Results

Figure -4: Service Time Experimental Result

SERVICE	ONE	TWO
Service Time 1	0.079	0.219
Service Time 2	0.735	0.516
Service Time 3	0.094	0.313
Service Time 4	0.766	0.125
Service Time 5	0.407	0.125
Service Time 6	0.266	0.016
Service Time 7	0.938	0.766
Service Time 8	0.782	0.688
Service Time 9	0.407	0.625
Service Time 10	0.047	0.047

Figure -5: Service Time Timing Result

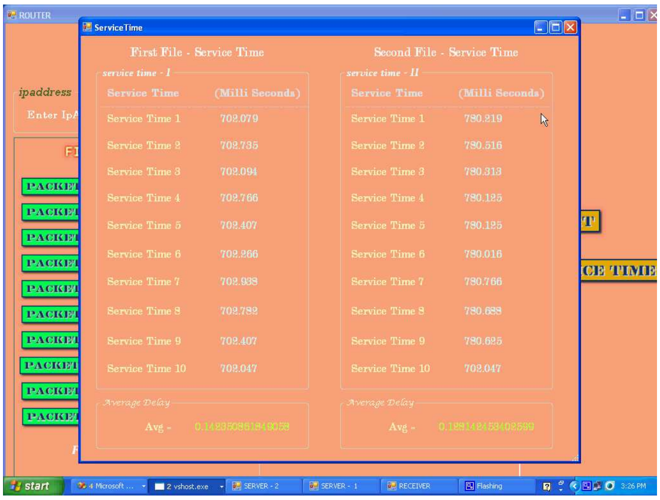


Figure -6: Output Screen Experimental Result

Sample Java Coding

Routers.java

```
packageserverstream;
import java.net.*;
import java.io.*;
importjava.util.*;
public class Router {
MainFrame mf;
DatagramSocket ds;
        DatagramPacket dp,dp1;
        byte data[]=new byte[1000];
        byte data1[]=new byte[1000];

        NodeDetailsnd=new NodeDetails();
        String route[][] = new String[1000][2];
staticintrin=0;
        String ip="";
        String msg="";
        String dest="";
intmyport=0;
      String destination="";
Router(MainFramem,int port)
      {try{
mf = m;
myport=port;
ds = new DatagramSocket(port);
for(int i=0;i<route.length;i++)
      {
route[i][0]="";
route[i][1]="";
```

```
        }
rin=0;
route();
}catch(Exception e){}
        }

public void route()
        {
try{

while(true)
                {data = new byte[1000];

dp = new DatagramPacket(data,0,data.length);
ds.receive(dp);

                    String sr=new String(dp.getData()).trim();
System.out.println(sr);
                    String rs[] = sr.split("#");
if(rs[0].equals("File"))
                    {
ServerSocketss = new ServerSocket(9999);
System.out.println("Socket Created");
```

Conclusion and Future Directions

From the study if we reduce the propagation delay for VBR by using some techniques like Head light Prefetching and with routing protocols and topology generation plus On-Demand Routing Protocols.

In this paper, the following VBR Image Processing algorithm which already exists is used. The concept Head light Prefetching Techniques is used with the integration of Mashing Video Sensing Techniques.

This study presents the design of a networked system for Mashing, rate control and error correction of video in Wireless Multimedia Sensor Network Environment on the theory of Mashed sensing. The objective of this work is to study a cross-layer system that jointly controls the video encoding rate, the transmission rate, and the channel coding rate to maximize the received video quality.

It is shown that Mashed sensing can overcome many of the current problems of video over WMSNs, primarily encoder complexity and low resiliency to channel errors. It is shown that the rate of Mashed sensed video can be predictably controlled by varying only the Mashed sensing sampling rate.

The study can be developed by further as a research work by the researchers who have interest in Image Processing Techniques as well as Networking Data Communication. It is a broad area for future work in Research.

Bibliography

[1] Ni, J., and Tsang, D. Large-scale cooperative caching and application-level multicast in multimedia content delivery networks. Communications Magazine, IEEE 43, 5 (May 2005), 98 – 105.

[2] Zepernick, H.-J., Fiedler, M., Lundberg, L., and Pettersson, M. Quality of Experience based cross-layer design of mobile video systems. In ITC Specialist Seminar on Quality of Experience (Karlskrona, Sweden, 2008), pp. 73–80.

[3] ˙Içkin, S., De Vogeleer, K., Fiedler, M., and Erman, D. The effects of packet delay variation on the perceptual quality of video. In 4th IEEE Workshop On User MObility and VEhicular Networks (On-MOVE 2010) (Denver, CO, USA, 2010), pp. 679–684.

[4] ˙Içkin, S., De Vogeleer, K., Fiedler, M., and Erman, D. On the choice of performance metrics for user-centric seamless communication. In Third Euro-NF IA.7.5 Workshop on Socio-Economic Issues of Networks of the Future (2010), EuroNF Network of Excellence, pp. 4–5.

[5] De Vogeleer, K., ˙Içkin, S., Fiedler, M., Erman, D., and Popescu, A. Estimation of quality of experience in 3G networks with the mahalanobis distance. In The Fourth International Conference on Communication Theory, Reliability, and Quality of Service (CTRQ 2011) (Budapest, Hungary, 2011), pp. 72–77.

[6] Chen, X., Wang, W., and Wei, G. User satisfaction evaluation of gprs network using pca. In Proceedings of the 6th international conference on Fuzzy systems and knowledge discovery - Volume 3 (Piscataway, NJ, USA, 2009), FSKD'09, IEEE Press, pp. 612–616.

[7] Hakami, S., Zaidi, Z., Landfeldt, B., and Moors, T. Detection and identification of anomalies in wireless mesh networks using principal component analysis (PCA). In Proceedings of the The International Symposium on Parallel Architectures, Algorithms, and Networks (Washington, DC, USA, 2008), IEEE Computer Society,pp. 266–271.

[8] Liang, G., and Liang, B. Balancing interruption frequency and buffering penalties in vbr video streaming. In 26th IEEE International Conference on Computer Communications (May 2007), pp. 1406 –1414.

[9] Akamai Technologies. http://www.akamai.com/, 2011.

[10] Real Networks Inc. http://www.realnetworks.com/, 2011.

[11] YouTube LLC. http://www.youtube.com/, 2011.

[12] GPAC. http://gpac.wp.institut-telecom.fr/, 2011.