Analyzing the Network Real-Time Multimedia Traffic Profile Based on Content

Antoaneta A. Popova, Ivo R. Draganov, Vladimir K. Poulkov, Alexander A. Krupev[1]

Abstract – **In this paper an analysis is presented concerning the network real-time multimedia traffic profile based on content. The throughput, round trip time and packets' size are analyzed when slow, medium and fast changing videos with low, medium and high resolution are being streamed over an Internet channel. The results obtained reveal number of useful directions for enhancing video streaming systems' operation.**

Keywords – **Multimedia Traffic, Real-Time Video Streaming, Throughput, Round Trip Time, Packet Size.**

I. Introduction

Real-time video and audio streaming became extremely popular in the last 15 years. At first low bit rates were incorporated for the separate flows – based on low resolutions and frame rates with higher quantizing factors. With the increase of the network bandwidths more and more multimedia flows become available for simultaneous reception along with the wide variety of other services. This leads to a situation where the network traffic profile even at the very user entry point becomes very complicated and in number of cases prevents some of the services to be used in accordance with some preliminary expectations.

In 2004 Pinson and Wolf [1] suggested a standard way for measuring video quality of video which could be streamed among other ways for transmission. Based on such measurements a lot of useful techniques for enhancing the quality of the receiving video were proposed such as dynamically resource allocation based on traffic prediction [2], synchronization control schemes [3] for video conferencing and preliminary defined QoS constraints approach [4].

So far most of the research for video and audio quality degradation when streamed over a network is based assuming a fixed size video usually at low resolutions and slow changes in the scenes [5-7]. Only recently some really extensive research on incorporating simultaneous video and audio streams with vast dynamic ranges along with other services was made [8, 9]. In this paper we try to propose an investigation on the network traffic profile based on different multimedia content. The paper is organized as follows. In part 2 a description of the test environment is given, in part 3 – the experimental results are shown and part 4 is a conclusion.

[1]Antoaneta A. Popova, Ivo R. Draganov, Alexander A. Krupev, and Vladimir K. Poulkov are with the Faculty of Telecommunications, 8 Kliment Ohridski Blvd., 1000 Sofia, Bulgaria, E-mail: {antoaneta.popova, idraganov, vkp}@tu-sofia.bg, asfalot@abv.bg

II. Test Environment Description

A. Network Set Up Configuration

The network set up for the performance evaluation of the video streaming throughput and other statistical network data is given in Fig. 1.

The local network connection from the gate to the host is 100 Mbps Ethernet over which the Internet traffic is passing at speed no less than 64 Mbps without any other clients on the same sub-network after the local router.

The local host is equipped with 100/1000 Mbps network interface card over ASROCK P43 Motherboard (CPU Quad Core® 2.8 GHz, 4 GB RAM) with 750 GB HDD running at 7400 rpm and VGA ATI Radeon 4800 Series. The operating system is MS® Windows ® XP® SP2 64-bit. As a local multimedia player VLC v.1.1.7 is used, as a bandwidth limiter NetPeek v.1.1 and as a network monitor Wireshark v.1.4.4.

All the traffic is measured (captured) at two dialing speeds which are proven to be standard in practice – 768 and 2500 kbps respectively.

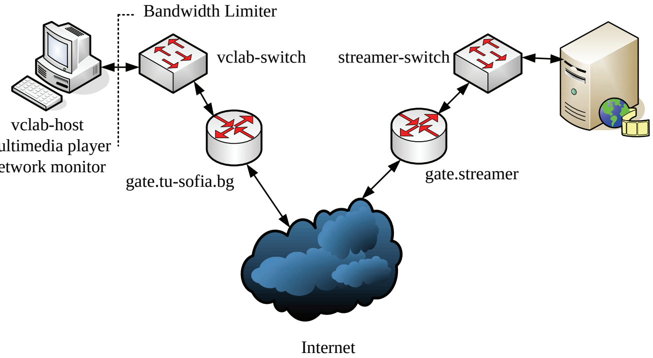

Fig. 1. Network Set Up Configuration

B. Test Videos

There are 3 types of videos used in the experimentation – a slow motion video with a news reader being filmed in a studio with no sharp moves in the scenes, a medium motion video showing football players moving with no too fast moves and a fast motion video – representing car racing in a city with lots of sharp moves, zooming in and out in large amount of details in the picture, large scale pan and tilt moves of the camera. Each video is in low, standard and high resolution.

The low resolution videos (400 x 226 pixels) are H.263 coded, while for the standard (854 x 480 pixels) and high resolution (1280 x 720 pixels) videos MPEG-4 AVC (Advanced Video Codec) coder is used all with variable bit rate option. The frame rate for all the videos is fixed to 29.970

frames per second. The color space is YC_bC_r 4:2:0 with 8 bits per component representation and the scan is progressive.

The audio interleaved along with the video for each clip is AAC (Advanced Audio Codec) coded consisting of 2 channels (stereo) sampled at 44.1 kHz in variable bit rate mode.

C. Test Procedure

The test procedure follows the steps given below:

1. Set up a speed limit of 768 kbps.
2. Receive slow motion video at low resolution.
3. Capture all the traffic at the local host.
4. Calculate the throughput, the round trip times and the packet lengths distribution.
5. Change the resolution to standard (medium).
6. Repeat steps from 2 to 4.
7. Change the resolution to high.
8. Repeat steps from 2 to 4.
9. Change the video to medium motion.
10. Repeat steps from 2 to 8.
11. Change the video to fast motion.

Repeat steps from 2 to 8.

III. Experimental Results

It is obvious and expected that with the decrease of the speed the temporal change of the throughput (Fig. 2 – Fig. 5) is more and more wide meeting a certain limits that lead to transitory interrupting in the video being received. For the dialing speed of 2500 kbps (Fig. 3) this doesn't happen although for the fast motion video with high resolution certain saturation in the top limit is visible (Fig. 5).

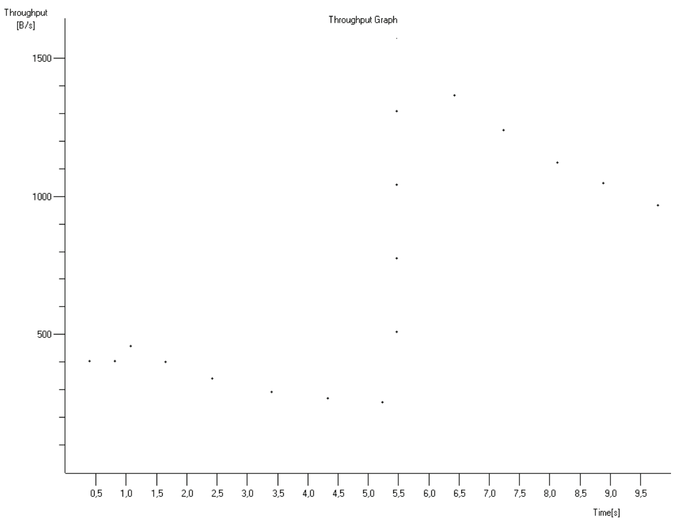

Fig. 2. Throughput for slow motion video with low resolution at 768 kbps limit

Interestingly there are number of cases in which for standard resolution and slow and medium motion the throughput is virtually constant but changing its average value from video to video.

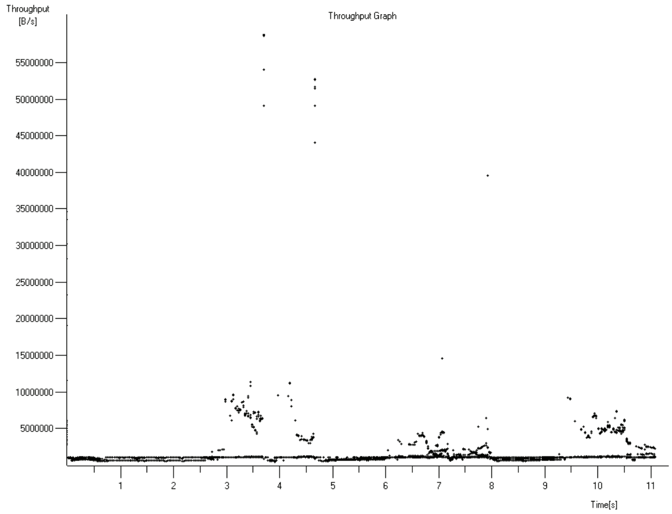

Fig. 3. Throughput for slow motion video with high resolution at 2500 kbps limit

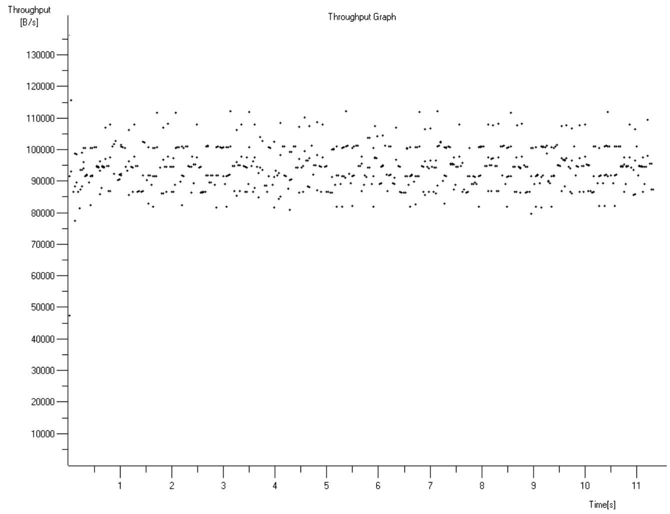

Fig. 4. Throughput for fast motion video with low resolution at 768 kbps limit

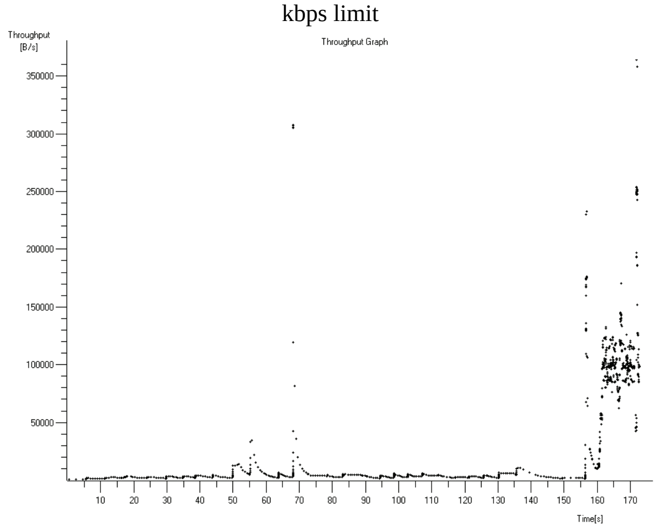

Fig. 5. Throughput for fast motion video with high resolution at 768 kbps limit

This property can be utilized in a wide range of practical cases where simultaneous transmit is needed for more than one multimedia streams and proper statistical scheme of time division could be applied saturating the channel to an extent such that interrupting can be minimized.

As for the round trip times (Fig. 6 – Fig. 9) the same behavior in the dynamic range change is observed for the slow and fast motion videos. This change ia a little bit unexpected for the medium motion video no matter of the resolution – a narrow wide range reveals the flat law for the distribution times for the packets. This could be of great importance when again multiple multimedia streams are going to be transmitted and statistically it is much easier to calculate the total amount of the bandwidth reserved for each of them in some timely manner. Actually the uniform distribution in such cases leads to simple linear equations for the time division intervals being estimated for each video to be streamed down.

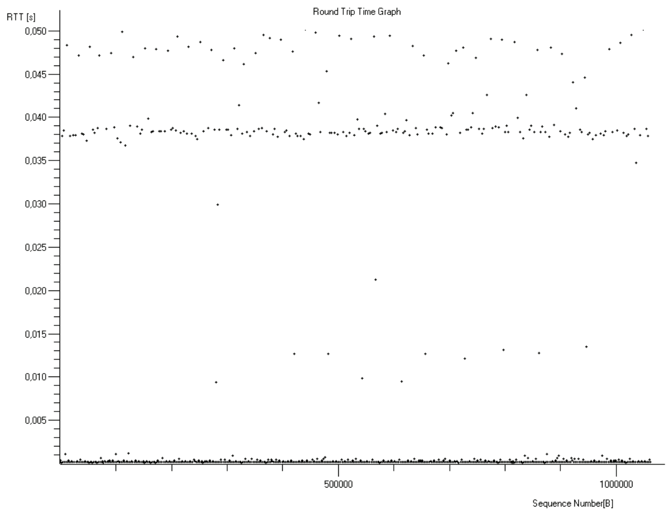

Fig. 6. Round trip time for slow motion video with low resolution at 768 kbps limit

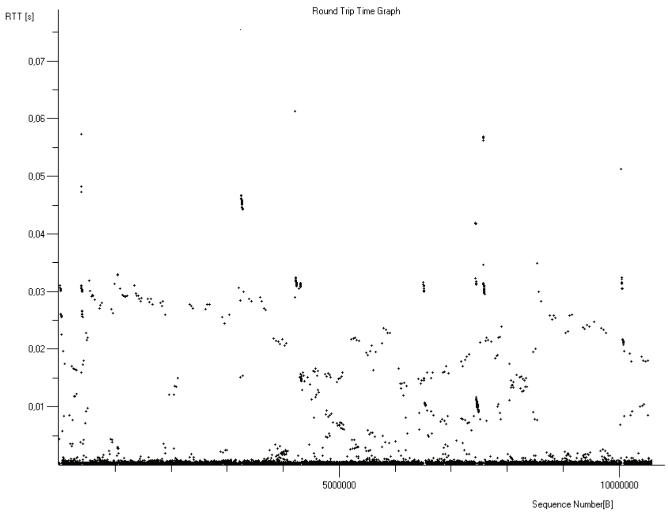

Fig. 7. Round trip time for slow motion video with high resolution at 2500 kbps limit

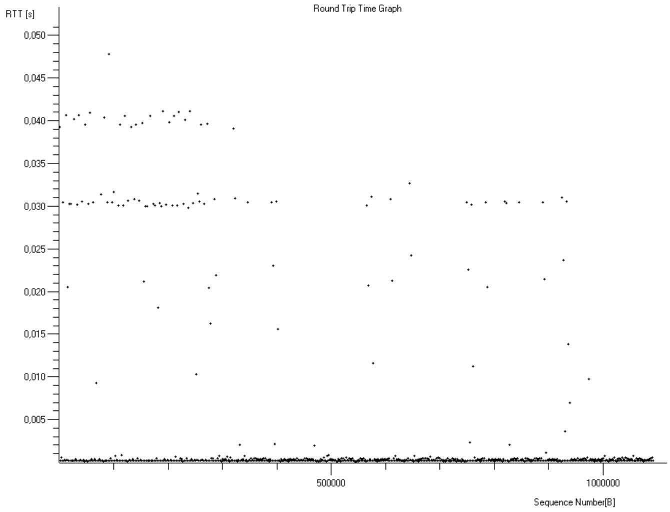

Fig. 8. Round trip time for medium motion video with standard resolution at 768 kbps limit

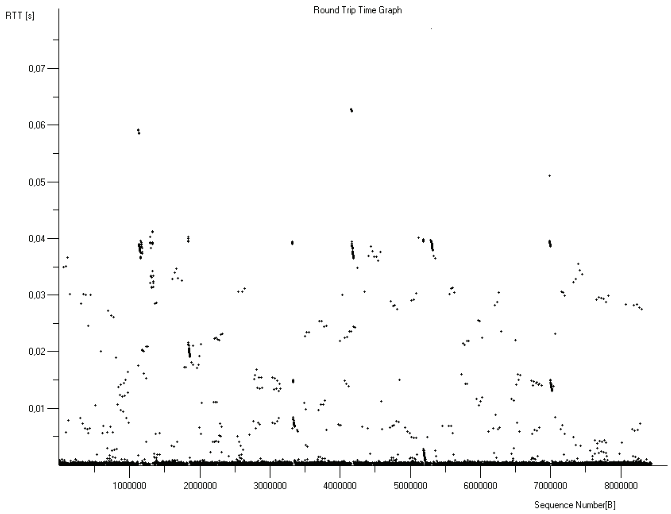

Fig. 9. Round trip time for fast motion video with high resolution at 2500 kbps limit

The packet lengths (Fig. 9 – Fig. 11) vary from about equally divided, in the ranges from 80 to 160 Bytes and from 160 to 320 Bytes for the low resolution videos (Type 1). This tendency occurs no matter of the speed of the scene changes which is then slightly changed, in the direction to change this ratio from 1:1 to 1:2 for both the ranges for the medium (standard) resolution videos.Then it goes up to even 1:3 for the high resolution videos (Type 6). A number of packets – between 7 and 12 %, for the high resolution video especially at faster scene changes, fall into the range from 40 to 80 Bytes which probably is a consequence of the saturations being observed for this particular cases in the bandwidth and due to the protocol specifications some considerable amount of data is being split into smaller packets for transmission. Further study of such behavior could be undertaken which might prove useful in cases where a small amount of users are being supplied with fast videos through narrow links perhaps in the last mile to their locations.

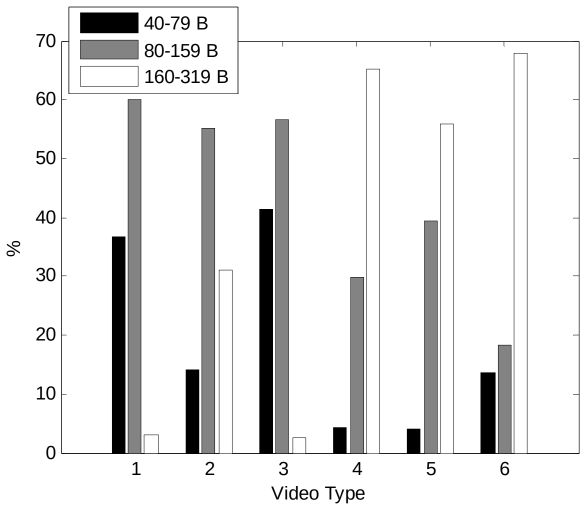

Fig. 9. Packet lengths distribution for slow motion video

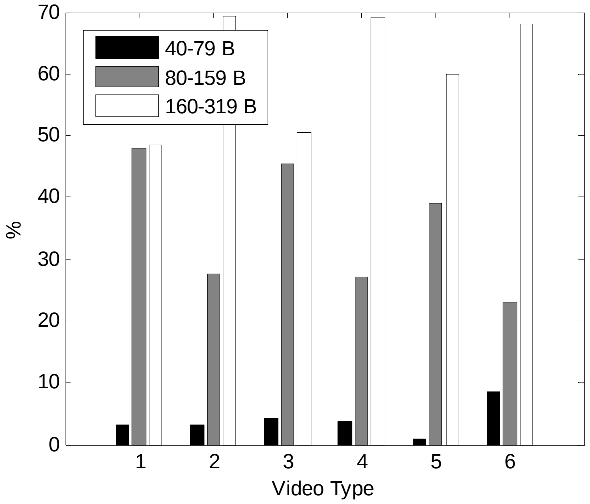

Fig. 10. Packet lengths distribution for medium motion video

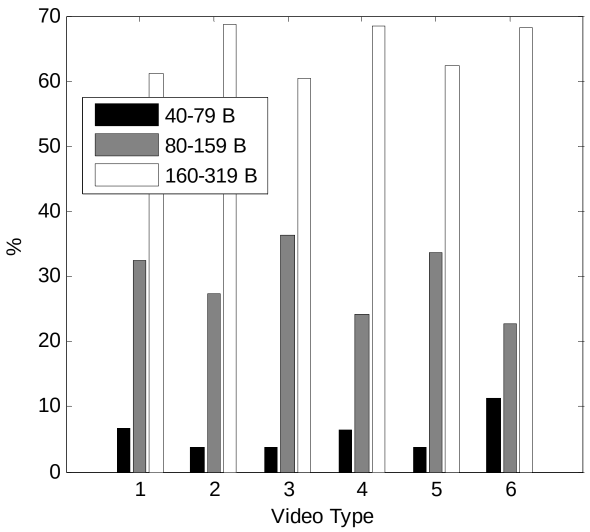

Fig. 11. Packet lengths distribution for fast motion video

IV. CONCLUSION

In this paper an analysis concerning the network real-time multimedia traffic profile based on content changing from slow to fast motion and at different resolutions is presented. The throughput, round trip time and packets' size are analyzed which reveal useful directions for network bandwidth planning when multiple clients should be supplied with high quality multimedia resources. Specifically the packet size distributions show what kind of other types of services could be incorporated along with the multimedia streaming in such a way that not exceeding the bandwidth limits no interference will occur based on this size in the process of packet flow control.

ACKNOWLEDGEMENT

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