NETWORK REQUIREMENTS FOR HIGH-SPEED REAL-TIME MULTIMEDIA DATA STREAMS

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Abstract

High-speed real-time multimedia applications such as Videoconferencing and HDTV have already become popular in many end-user communities. Developing better network protocols and systems that support such applications requires a sound understanding of the voice and video traffic characteristics and factors that ultimately affect end-user perception of audiovisual quality. In this paper we attempt to propose an analytical model for the various network factors whose interdependencies need to be modeled for characterizing end-user perception of audiovisual quality for high-speed real-time multimedia data streams. Towards validating our theory presented in this paper, we describe our initial results and future plans that consist of conducting a series of experiments and potential modifications to our proposed model.

As the portion of middle and high-speed RTP (Real-Time Protocols) applications in the total communication over the Internet grows, it is important to understand the behavior of such traffic over the TCP/IP networks. The impetuous growth of traffic volumes and the rapid advances in the technologies of new-generation networks like IPv6 have made it possible for use of such high-speed applications, but with new demands on the network to deliver superior Quality of Service (QoS) to these applications.

Our research focuses on the area of quality criteria for middle and high-speed network applications like

- Video Conferencing / Streaming
- Grid applications
- P2P applications

Audio- and Videoconferencing are two major multimedia communications that are defined by various international standards such as H.320, H.323 and SIP. These standards describe the system specifications for applications that support real-time transmission of continuous media information on packet-switched networks (Internet). Such real-time multimedia data stream applications have many properties in common with other applications. However, they have also a number of specific features:

- They may require real-time transmission of continuous media information.
- Volumes of data to be exchanged are substantial. Communications require significant amounts of real-time processing for encoding and decoding.
- Most of these applications have a distributed architecture, in particular when serving residential users.

Based on the above observations, several studies have been conducted [5,6] and researchers have formulated several types of criteria [8,9,10]. None of the previous work focuses on performance evaluation of high-speed multimedia data streams at bandwidth ranges of *384-2048 Kbps*.

There are two basic approaches to the estimation of audio- and videoconferencing quality. The first approach analyzes the various levels of network factors [3,4,7] that contribute towards assessing audiovisual quality. The other approach [5] analyzes the differences in the audio and video streams at the sender and the receiver ends. Specifically for video streams, the main quantitative parameters evaluated for each stream include frame rate and linearity in addition to the other parameters mentioned above.

Our considerations for modeling RTP data steam' quality are based upon the comparison of three inalienable components:

- Structure of initial and receiving signals
- Network as a transmitting media
- Human perception of real-time multimedia stream information

The previous investigations in this area have been restricted by comparative analysis of only two components from above list. In paper [3] the authors are interested in studying aspects of the Human factors, which deal with end-user perception of audiovisual quality and the Network factors, which contribute to any network's health. Performance bounds for delay, jitter and loss were determined for various network health scenarios [3]. The obtained performance bounds were then mapped to end-users perceptions of the overall audiovisual quality using the popular Mean Opinion Score (MOS) ranking technique.

Another novelty of our study is that it focuses on characterizing real-time multimedia traffic that demand large amounts of network bandwidth resources, more specifically in the range of *384-2048 Kbps*. The major vendors of multimedia equipment have such products in their lines and claim that higher speeds provide better quality of audiovisual information. We believe that our research has the potential to lead to a new standard that accurately characterizes the transmission of high-speed audiovisual information over packet switched networks. Our approach of analysis of various parameters at the initial multimedia signal level and at the network level allows us to derive a general relationship between them that ultimately affects end-user perception of audiovisual quality.

Our primary focus is on understanding how the various values of network diagnostics, obtained by measuring throughput B, delay D, jitter j and loss p in the network, can affect end-user perception of audiovisual quality.

In the pursuit to identify the most dominating network factors that affect end-user perception of audiovisual quality, the authors of paper [3] normalized the scales of these factors and plotted them against both the Subjective and Objective MOS assessments. They have observed that end-user perception of audiovisual quality is more sensitive to changes in jitter than to changes in delay and loss.

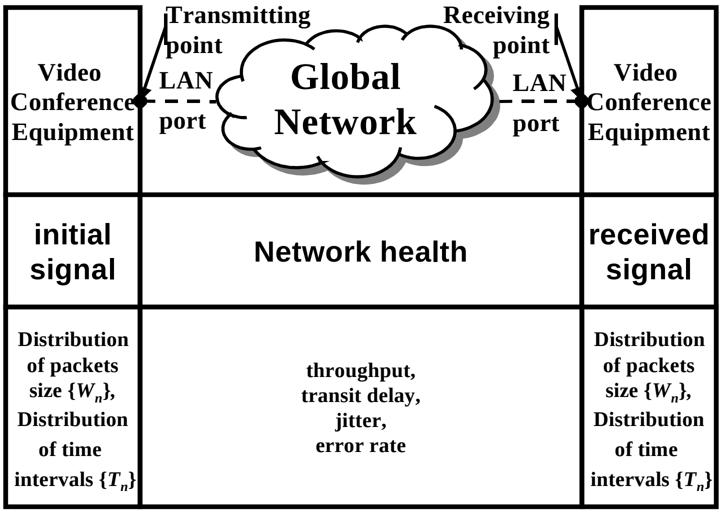


Fig.1. The influence of network factors

Such behavior of assessment function Q_{MOS} is easy explained by avoiding congestions at receiving point, see Fig.1. Comparing the magnitudes of inter-packets interval I_n and of network jitter j we can make a withdrawal about existence of coefficient k_1, described the limiting interconnection of this parameters.

$$k_1(Q_{MOS})=E(I_n)/E(j) \qquad (1)$$

The smaller value of interval I_n corresponds more qualitative video stream. We can improve the parameters of video stream between fixed end points when coefficient $k_1(Q_{MOS})$ begins to decrease below the limit defined by Eq. (1) then the assessment point Q_{MOS} is consequently reduced. In other words we calculate the upper limit of the $k_1(Q_{MOS})$ for a given video connection and end-to-end topology.

The basic condition for successful steaming is the least distortion of initial sequence at receiving point. There are three factors of distortion like the packet loss, re-ordering effect and grow skip of arriving intervals Y_n as it is show on Fig. 2. In order to eliminate distortion of initial sequence the process of buffering has been included in decoding process. This element allows keeping the re-ordering packets during the time Y_{max}. If the arriving interval Y_n exceeds this period Y_{max} then this packet is considered losing one.

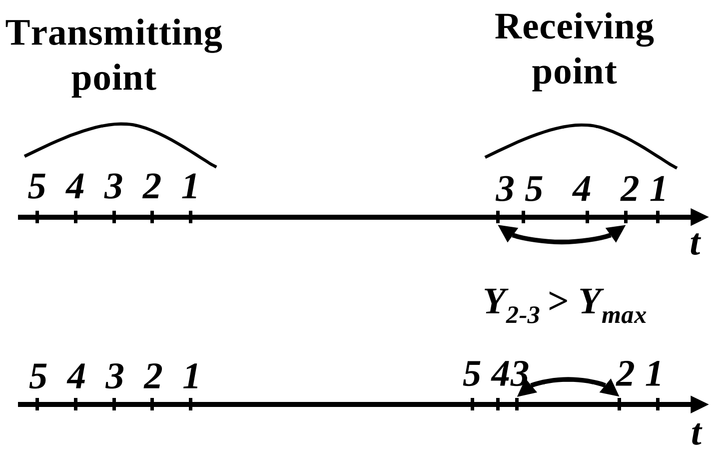


Fig.2. Packet reordering due to distortion of initial signal

The tail of such distribution (when $Y_n>Y_{max}$) coincides precisely with tail of Poison distribution. Therefore the probability of event $P(Y_n>Y_{max})$ when arriving interval exceeds the buffering time equals

$$P(Y_n>Y_{max}) = exp(-\lambda Y_{max}) \qquad (2)$$

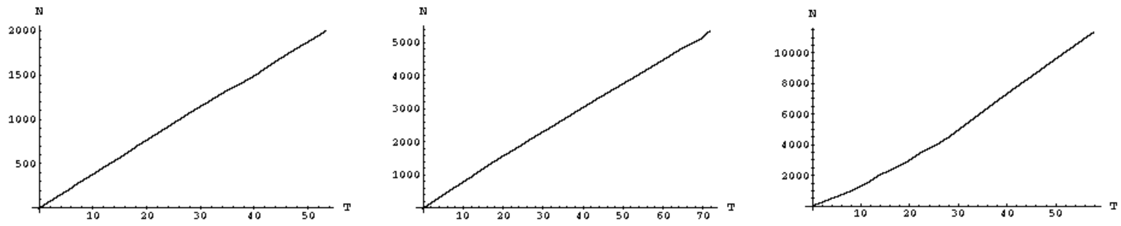
It is easy to calculate the buffering time Y_{max} for different speeds of video connection so that we have found λ magnitudes in Table 1. For example, the reasonable value for Y_{max} at *384 Kbps* equals *150 ms*.

Connection speed in *Kbps*	Packet size W_n in *bytes*	Inter-packet intervals I_n in *ms*
128	600-800	80-100
384	900-1100	25-30
768	1100-1300	15-20
1920	1200-1400	6-10

Table 1. Mean size of W_n and I_n

First series of experiments is aimed at investigating the structure of UDP streams. Our first set of tests will be conducted in a local network using Fast Ethernet of Gigabit Ethernet standards since the network factors do not significantly in6ence the connection quality. The results of this series of experiments have been shown in Table 1 and in Figures 3 – 5.

The Fig. 3 shows the cumulative number N of UDP packets that arrive at measurement point during a 60 seconds interval. The arrival rate $\lambda = 1/E(I_n)$ remains mostly constant throughout the 5 minutes interval, i.e. we can conclude this to be a Poisson process [1].



a) b) c)

Fig.3. RTP packet arrivals during a 60 seconds interval

The Fig. 4 shows a typical example of a frequency histogram of inter-packet intervals I_n. The local peaks on Fig.4a correspond approximately to 10, 20, 30 and 40 *ms* at connection speeds of 384 *Kbps*. The considerable portion of small inter-packets interval, i.e. $I_n<1$ *ms* in *60* percent events for 4c graphic, establishes the fact that one frame is transmitted by two ore more UDP packets. It is explained by restriction of packet size in 1460 bytes.

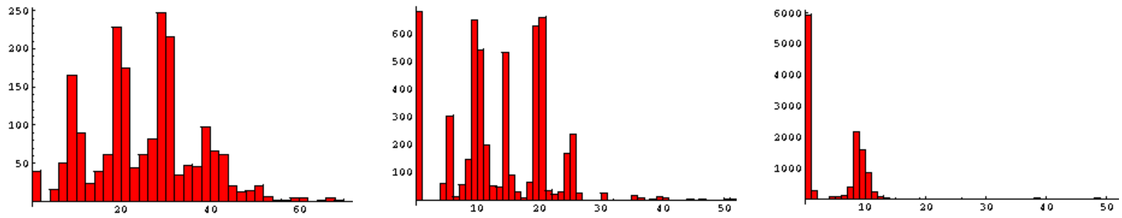


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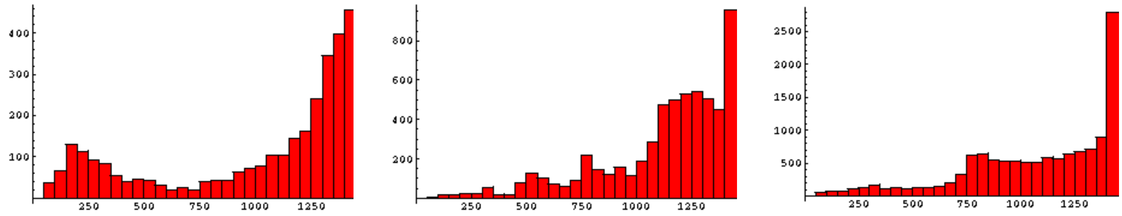
Fig.4. Typical distribution of inter-packet intervals I_n

A typical histogram of packet size distribution for H.263 traffic can be studied using Fig. 5. The x-axis corresponds to the size of W_n in bytes; y axe reflects the relative frequency of such packets in our data. It can be seen that about *25%* of the small packet sizes do not exceed *440 bytes* at the connection speed *384 Kbps*. The data sets from Fig. 5b,c include also the considerable part of small packet size.

The next experimental series will be dedicated to the investigation of the modified Eq. (1). We plan to deploy a number of measurement points that are distributed over the Internet to obtain realistic network scenarios. Each measurement site will procure hardware and software solutions for measurements as well as the required videoconferencing equipment. Our initial list of sites include OARnet, Ohio, USA; Irish Research and Education Network (Heanet), Dublin and Russian Research and Education Network (RBNet), Moscow and Samara. We plan to use the RIPE test box [12] and the OARnet H.323 Beacon [2] for obtaining measurements of the following network parameters: $D(RTT)$, j, p.



a) b) c)

Fig.5. Typical distribution of packet sizes W_n

The results of this study will lead to better understanding of the characteristics of high-speed real-time multimedia streams in the Internet [12] which could then result in better network design trade-offs and design of new network protocols that supplement Internet multimedia systems.

References

[1] Chadi Barakat, Patrick Thiran, Gianluca Iannaccone, Christophe Diot, Phillipe Owezarski, A flow-based model for Internet backbone traffic, IEEE Transactions on Signal Processing - Special Issue on Signal Processing in Networking, vol. 51, no. 8, pp. 2111-2124, August 2003

[2] P. Calyam, W. Mandrawa, M. Sridharan, A. Khan, P. Schopis, H.323 Beacon: An H.323 application related end-to-end performance troubleshooting tool, Proceedings of ACM SIGCOMM Network Troubleshooting Workshop (NetTs' 04), Portland, October 2004

[3] P. Calyam, M. Sridharan, W. Mandrawa, P. Schopis, Performance Measurement and Analysis of H.323 Traffic, PAM2004

[4] A. Clark, Modeling the effects of burst packet loss and recency on subjective voice quality. (2001)

[5] R. Finger, A. Davis, Measuring Video Quality in Videoconferencing Systems

[6] Francois 6ckiger, Understanding Networked Multimedia, Prentice Hall, Inc. 1995

[7] W. Jiang and H. Schulzrinne, Modeling of packet loss and delay and their effect on real-time multimedia service quality, Proc. of NOSSDAV, 2000

[8] ITU-T Recommendation G.107, The Emodel, a computational model for use in transmission planning, 1998

[9] ITU-T Recommendation P.911, Subjective audiovisual quality assessment methods for multimedia applications, 1998

[10] J. Mullin, L. Smallwood, A. Watson, G. Wilson, New techniques for assessing audio and video quality in real-time interactive communications. IHM-HCI Tutorial, 2001

[11] V. Prokhorov, V. Kosarev, A. Petukhov, A. Smirnov, I. Prokhorov, Multifunctional System of Internet Video Connection “VIPPHONE”, Proc. of Scientific Service in Internet 2004, p.262-264

[12] http://www.ripe.net/test-traffic/index.html