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A new tool for quality of multimedia estimation based on network behaviour☆

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Summary In this paper, we present a software tool capable of predicting the final quality of triple play services by using the most common assessment metrics. The quality of speech and video in network environment is a growing concern of all the internet service providers to carry the multimedia traffic without the excessive delays and losses, which degrade the quality of multimedia as it is perceived by the end users. Prediction mathematical model is based on results obtained from many performed testing scenarios simulating real behavior in the network. Based on the proposed model, speech or video quality is calculated with regard to policies applied for packet processing by routers and to the level of total network utilization. The application cannot only predict QoS parameters but also generate the source code of particular QoS policy setting according to the user interaction and apply the policy to the routers in the network. Contribution of the work consists of a new software tool enables network administrators and designers to improve and optimize network traffic efficiently.

Introduction

Since last two decades, the concept of next generation network (NGN) has become a dominant for network infrastructure developing. This concept is based on IP protocol and allows the transfer of formerly separate services (voice, video and data) by one common network infrastructure. However, this transition had to deal with some difficulties because packet networks based on IP protocol had not been designed to transfer delay-sensitive traffic. Packet network has to use supplementary mechanisms securing the quality of service during the transmission over the network, able to provide a high-quality interactive communication similar to standard fixed lines (PSTN).

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Congestion management features allow networkers to control congestion by determining the order in which packets are sent out an interface based on priorities assigned to those packets. Congestion management entails the creation of queues, assignment of packets to those queues based on the classification of the packet, and scheduling of the packets in a queue for transmission. During periods with light traffic, that is, when no congestion exists, packets are sent out the interface as soon as they arrive. During periods of transmit congestion at the outgoing interface, packets arrive faster than the interface can send them. If the QoS (Quality of Service) policies are used, packets accumulating at an interface are queued until the interface is free to send them; they are then scheduled for transmission according to their assigned priority and the queueing mechanism configured for the interface. The router determines the order of packet transmission by controlling which packets are placed in which queue and how queues are serviced with respect to each other.

The next important factor is network monitoring. Continuous monitoring of network infrastructure is the possible way to increase the quality of multimedia transmissions for most users, because it allows the network administrators to be informed as soon as the problem rises and consequently to change the network routing and queuing policies.

Therefore, the purpose of the application described in this paper is to provide an effective monitoring tool and allow fast reaction on the situation by changing settings of QoS policies. The application aims to be an alternative to expensive monitoring tools, as well as a helpful tool for designing the network infrastructure and implementation of QoS policies on selected routers.

State of the art

Keeping the user's satisfaction with the highest level should be one of the major goals of all internet service providers. The way how to evaluate the final QoS can be divided into two groups. The first group of tests is based on the human factor, and these tests are very subjective. On the other side, objective test metrics use mathematical models and offer time and cost savings. Both groups include standardized tests and metrics for quality of service assessment. Objective tests are preferred by network administrators because depends on network parameters like overall delay or packet loss. Subjective tests express human perception and research in last years try to find the relation between subjective and objective methods.

Works (Han and Gabriel-Miro, 2015; Khan and Sun, 2008; Uhrina et al., 2013) are focused on the correlation between subjective and objective metrics. According to their results, objective evaluation metrics for voice or video like PESQ and SSIM reached very good correlation with human perception. This is the reason we used for our estimation model these objective metrics.

International Telecommunication Union (ITU) created two well-known objective metrics for voice evaluation, namely PESQ and E-model. The results provided in (Khan and Sun, 2008) showed that the PESQ algorithm accommodates the effects of packet loss on speech quality better than the E-model, and is, therefore, better suited for the task making its estimation a sensible way for improving the precision of the estimation. In (Meky and Saadawi, 1997; Mrvova and Pocta, 2013; Bhattacharya et al., 2003) the authors use neural networks and genetic algorithms to estimate the quality of speech and video, but the model-specific approach for the packet loss determination is used.

For video objective evaluation, the best correlation with human perception reaches Structural Similarity Index (SSIM) (Uhrina et al., 2013). In papers (Frnda et al., 2015a; De Rango et al., 2008), are presented results that describe robustness of video codecs and their settings on packet loss impairment and analyze efficiency for video streaming service.

The main motivation behind this work is to present computational system able to estimate QoS according to the network behavior and bring a comprehensive view of all network parameters and codec types and their influence on final quality of multimedia services.

Methodology

Nowadays, practically, computer networks are not built only on a homogeneous infrastructure, but they use heterogeneous devices. We created testing topology that contains several well-used network vendors. Implementation of the computational model was carried out in programming language C#.

Measured parameters

The multimedia quality estimation is meant to be used with the basic characteristics of the IP networks. Factors affecting the multimedia transmission are Delay, Jitter and Packet Loss (Han and Gabriel-Miro, 2015).

Objective measurement can be either an intrusive or a non-intrusive method. The intrusive method typically compares two input signals, a reference signal, and a degraded signal which are received at the receiving side of the communication chain. The most successful intrusive method of ITU-T is defined in Recommendation P.862 (PESQ). Compared to nonintrusive methods, most of the intrusive methods have more accuracy but are inconvenient to use for real time monitoring. The non-intrusive methods focus on measuring real-time voice traffic. This type of measurement is designed to predict speech quality without a reference signal.

The most popular non-intrusive method of ITU-T is defined in Recommendation G.107 E-model, which was designed as a transmission planning tool. The E-model is a mathematical model that merges the impairment factors on the conversational path, before calculating conversational quality.

Recommendation G.107 contains default values which enable to simplify the calculation so that it corresponds with packet networks as follows:

$$R = 93.35 - I_D - I_{E-EFF} \tag{1}$$

The parameter I_D represents the factor of impairment caused by the too long transfer delay and the factor I_{E-EFF} represents impairment caused by low bit-rate codecs and actual packet loss ratio of the network.