Simulator for IP Telephony Access Network

Toni Janevski[1], Bojan Angelov[2], Aleksandar Tudzarov[3]

Abstract - **IP telephony is an evolving service in telecommunications. Therefore, having a tool for convenient dimensioning is essential. In this paper we present a simulator, based on existing model of the Markov Modulated Poisson Process (MMPP), that calculates the most important parameters such as packet delay and loss for a set of voice input sources and specified link properties.**

Keywords - **IP telephony, Link dimensioning, Markov Modulated Poisson Process, Quality of Service.**

I. Introduction

IP telephony has been one of the dominant issues in the telecommunications industry for several years. Many telephone operators and Internet providers have future plans concerning this kind of service. Voice applications, such as telephony, have been used on the best effort service bases provided by the Internet for quite some time. There is an ongoing work in the research community and industry for providing QoS (quality of Service) architectures and dimensioning principles for implementing IP telephony as a standard service with satisfying QoS offered to the users [1].

This kind of service is especially interesting for new investors in the business of telecommunications because with considerately smaller investments than ones used in the past they can become competitive on the consumer market.

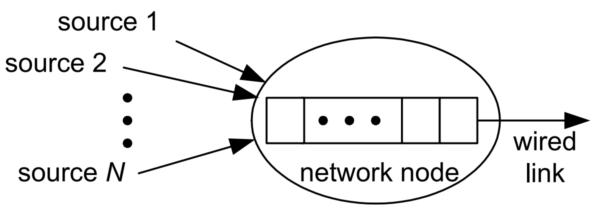

Fig. 1. The problem: dimensioning a link for voice sources over IP

Fig. 1 illustrates the problem scenario we are addressing. A number of packet voice sources are multiplexed onto the link. The link has a limited amount of buffering space, which sometimes will result in the loss of packets, thus degrading the quality of service. So, for a given link and given number of voice calls our goal is to estimate the quality of the system and set its parameters for best results.

On the other side, buffering of the voice packets introduces additional delay. Telephony is real-time communication, which is sensitive to delays. Therefore, there should not be allowed large delays and henceforth we do not need large buffers for voice packets. According to the ITU-T [2], recommended total delay for one-way telephone communication is less then 150 ms, and more then 400 ms is not tolerable. Because users are used to certain quality of telephone communication, introducing IP telephony should not degrade it. If maximum number of hops is 15, then maximum acceptable delay per hop is 400 ms /15 = 26.7 ms. Not all delay is due to buffering, there is delay due to the transmission, signal processing etc. We may say, without losing the generality, that half of the total allowed delay budget can be allocated to queuing delay [1]. Then, we get 13.3 ms budget delay per hop. In this paper we are interested in the queuing delay on the wireless access link. If capacity of the wireless link is 2 Mbps and IP packet size is 100 bytes, then maximum buffer size should be 33 packets of 100 bytes each or 13.3 ms. Of course, we assume that IP telephony traffic is separated from the other traffic, such as non-real time traffic, by using appropriate mechanisms, i.e. Differentiated services. IP telephony traffic should be served with a priority.

Considering the discussion above, our aim was to focus on dimensioning IP network links for carrying packetized voice calls. The first step was to find a suitable mathematical model that will be a basis for our IP telephony simulator. The model that we used is presented in Section II. After that follows the verification of the model with the results from the simulations. The results from the analysis are given in Section III. The paper is concluded with Section IV.

II. Analytical modeling for IP telephony

In this section we define a mathematical model for IP telephony analysis. Usually, in teletraffic theory voice calls are well modeled by using Poisson process. For IP telephony we are using a similar approach as for traditional public switched telephone networks. Therefore, we begin with some basic definitions of Markov and Poisson processes which are being used to describe the arrival process of a single IP telephony source and proceed with the superposition of independent identically distributed sources. The sources are multiplexed on a bottleneck link through a queue of limited size. This means that the previously described arrival process is fed into a simple D/1/K queue. It is deterministic, has a single FIFO server and variable buffer size.

A. Markov Processes

An important class of stochastic processes are Markov processes. This class of processes have some special properties that make them manageable to treat mathematically. A Markov process is governed by the *Markov*

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property which states that the future behaviour of the process given its path only depends on its present state.

Let $\{X(t) \geq 0\}$ be a time continuous stochastic process which assumes non-negative integer values. The process is called a discrete Markov process if for every n $\geq$ 0, time points $0 \leq t_0 < t_1 \cdots < t_n < t_{n+1}$ and states $i_0, i_1, \cdots, i_{n+1}$ it holds that:

$$P(X(t_{n+1}) = i_{n+1} | X(t_n) = i_n, X(t_{n-1}) = i_{n-1}, ..., X(t_0) = i_0)$$
$$= P(X(t_{n+1}) = i_{n+1} | X(t_n = i_n)) \quad (1)$$

By a definition, only the present state gives any information of the future behaviour of the process. Knowledge of the history of the process does not add any new information.

In this paper we are intersted in processes with time homogeneous properties. In other words the intensity of leaving a state is constant in time. So, the transition probabilities depends only on the current state of the process.

The most used class of the Markov processes in analysis of queuing systems is *birth-death* process. The only possible state transitions in this kind of processes are from i to i-1 or from i to i+1. The transition intensity from state i to i+1 is $\lambda_i \geq 0$ for $i \geq 0$ and the transition intensity from state i to state i-1 is $\mu_i \geq 0$ for $i \geq 1$. Birth-death processes suitably model certain types of queuing systems. In that case the numbers λ_i and μ_i are interpreted as the arrival rate of the queue and service rate of server, respectively.

B. The Poisson process

The Poisson process is one of the most used processes in teletraffic modeling. Here, we provide one of its possible definitions [1].

We introduce a variable $X(t)$ = *number of occurrences in the interval* (0, t]. Further, let assume that occurrences in non-overlapping intervals are independent of each other. Also, we assume that probability distribution of the variable X is stationary, i.e. does not change in time. Then, we refer to the process as a Poisson process if the number of occurrences (e.g. call/packet arrivals) follows the distribution:

$$P[X(t) = x] = e^{-\lambda t} \frac{(\lambda t)^x}{x!} \quad (2)$$

where λ is the rate of occurence. In a telecommunication system, number of users is usulaly many times greater than number of active users, thus we may apply the Poisson process. Another important feature of the Poisson process is very low correlation, and description of the process with a single parameter i.e. the mean rate λ. Because voice traffic is predictable and has low correlation (in contrast to data traffic, such as web traffic, which is bursty in nature in different time scales), we may apply the Poisson process to analysis of the voice traffic, even when we are considering the voice-packet arrivals and departures.

C. Single source properties

Most standard voice encodings have fixed bit rate and fixed packet delays. Thus, they are producing a stream of fixed size packets. However, this packet stream is produced during talk spurts only, because the voice coder sends no packets during silence periods. We can model the behavior of a single source by a simple ON-OFF model (Fig. 2).

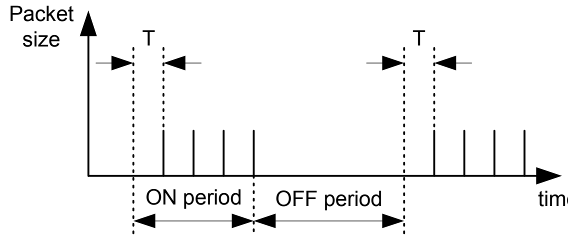

Fig. 2. Characteristics of a single source

During talk spurts (ON-periods), the model produces a stream of fixed size packets with fixed inter-arrival times T. We assume that the ON and OFF periods are exponentially distributed with mean α and β respectively. A voice source may be viewed as two state birth-death process with birth rate α and death rate β. Because of the exponentially distributed talk spurts and consequently OFF periods, the emission of packets can be regarded as a Poisson process with intensity T.

D. The superposition of independent voice sources

The superposition of voice sources can be viewed as a birth-death process where the states represent the number of sources that are currently in the ON-state. Here the state i represents that i sources are active in a talk spurt. This kind of superposition can be modeled with the Markov Modulated Poisson Process (MMPP). The model graph of the MMPP is shown in Fig. 3.

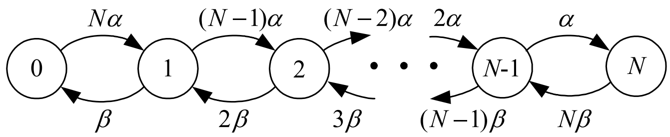

Fig. 3. Superposition of N voice sources with exponentially distributed interarrival times

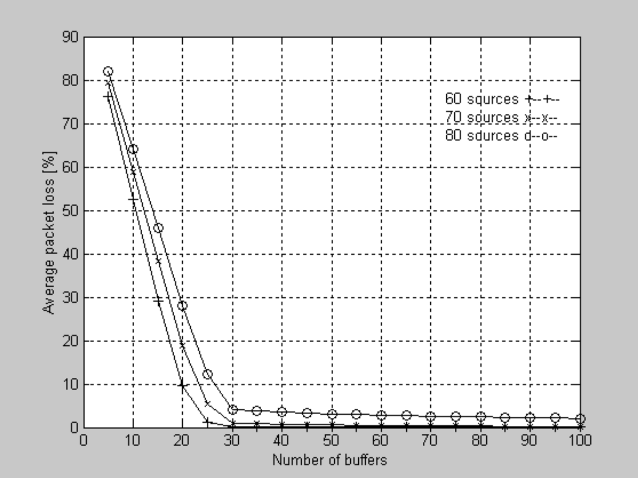

Fig. 4. Average packet loss vs. number of buffers for different number of voice sources

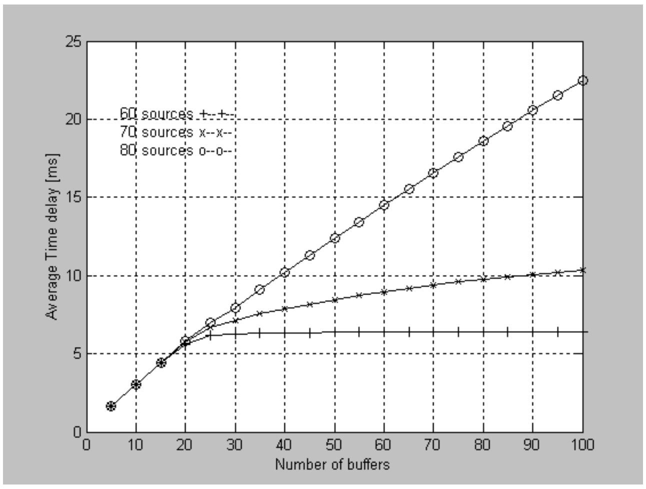

Fig. 5. Average packet delay vs. number of buffers for different number of voice sources

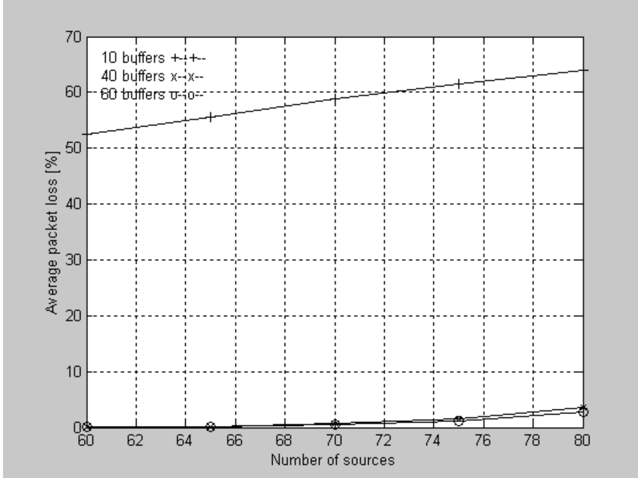

Fig. 6. Average packet loss vs. number of voice sources for different buffer lengths

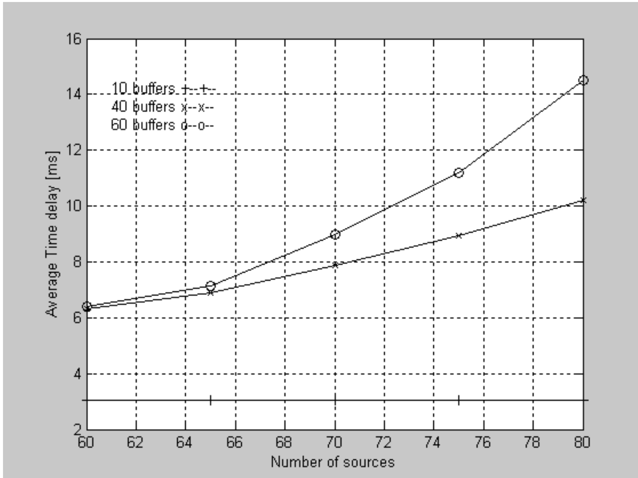

Fig. 7. Average packet delay vs. number of voice sources for different buffer lengths

The superposition of Poisson processes is also a Poisson process. We can therefore simply add the intensities of the sources that are currently in a talk spurt and receive a new Poisson process for the superposition.

III. Simulation Analysis

We implemented our model in Matlab programming environment. In this section we present the results considering the IP telephony analysis.

As input parameters we use number of IP voice sources, bandwidth of the link and buffer length at the network nodes. We observe QoS parameters, packet losses and packet delay.

There are various voice coders. All of them are using RTP/UDP/IP protocol stack. For the input parameter values, considering the simulator setup, we used the same values as found in [1]:

- 32 kb/s ADPCM voice encoding with 16 ms packet inter-arrival time, which results in 64 bytes of voice payload per packet, a protocol overhead of 40 bytes (12 bytes for RTP, 8 bytes for UDP and 20 bytes for IP). Link headers are not included. So, the total packet size is 104 bytes. We use fixed size for all voice packets.
- The duration of talk-spurts and silence periods is exponentially distributed on the positive integers with a mean of 0.351s for ON-periods and 0.650s for the OFF-periods.
- The bottleneck is T1 link with bandwidth of 1.536 Mb/s.

We may use different input parameters depending upon the capacity of the links that we are analyzing or voice coders.

In the simulations we use between 60 and 80 sources to load the link because with the given capacity of the link of 1.536 Mbps, 98 % link load is achieved when the number of voice sources is 84.

The buffer size we vary up to 100 buffers (one buffer correspondents to buffer space for one IP packet with length of 104 bytes), which mean that the maximum queuing delay in the buffer can be 54 ms.

The results from the graphs shown on Figs. 4, 5, 6 and 7 can be interpreted according to the specific desire of the operator. We were following the ITU recommendation G. 114 [2], which prescribes standard parameters such as average time delay and average packet loss. The values of these parameters can be read from the graphs and appropriate conclusion can be given. The first two graphs (Figs. 4 and 5) show the average packet loss and time delay where number of sources is used as a parameter regarding the number of places in the buffer. The other two graphs (Figs. 6 and 7) show the average packet loss and time delay where number of buffers is used as a parameter regarding the number of sources.

The duration of the simulation is 1000 s. We chose this value for the duration because numerous simulations showed that it is enough to obtain a steady state of the system. This means that for a set of parameter values mentioned earlier, 1000 seconds is enough for achieving satisfying results with desirable accuracy and fast calculations.

IV. CONCLUSIONS

Our goal was to find an accurate mathematical model and create a simulator, which can be used as a powerful tool for conveniently dimensioning network links for IP telephony. We implemented a model based on a Markov modulated Poisson process in Matlab. This model represents an IP telephony simulator capable of calculating the packet loss behavior and the time delay of the packets when a number of homogeneous voice sources are multiplexed onto a bottleneck link.

Considering the results shown in the previous Section, this model can be applied in several different situations. One of the typical scenarios is when an operator has reserved some bandwidth and needs to give to his users a quality of the IP telephony that will satisfy the requirements from the G. 114 [2]. This means that the operator can use the simulator to obtain the maximum number of users that can use this particular service by given constraints on the packet loss and delay.

One may extend this approach to a wireless environment. Also, it can be used as an input to an admission control algorithm. These are topics of a future work in this area.

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