

Modeling the Packet Loss in VoIP

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1. Introduction

Internet is the most effective means of communication in today's world. Information, on the Internet, is transmitted in the form of a packet which is passed over several networks to finally reach the destination. In this process, the intermediate hosts forward the packets until the packet reaches the destination. However, if the queue is full, then these hosts drop the packet.

Voice over Internet Protocol is a technology of transmitting voice on the Internet. Human voice is digitized, encoded, and transmitted on the Internet in the form of voice packets. These packets are received at the other end, decoded and converted back to analog signal. Loss and delay of voice packets result in the degradation of voice quality, since voice is a real-time application. Consecutive losses worsen the situation.

2. Experiment, Results, Discussion, and Significance

Packet loss is an indication of congestion. Packet loss models in the literature [1] argue that the chances of a packet getting dropped completely depend on the status of the previous packet. For instance, if the packet 'i' is lost, then the chances of packet (i+1) getting lost are very high. Similarly, if the packet 'i' is successfully forwarded, then the packet (i+1) will be successfully transmitted. The probability of a packet loss given the status of the previous packet can be easily determined using Markov chains [1], [2], [3]. In these models, parameters associated with time were not taken into consideration.

In this paper, inter-arrival time was considered to be an important parameter affecting the packet loss given the status of the previous packet. This model was based on the semi-Markov process which considered the current state, and the corresponding state holding time to decide the future state of a system.

Markov chain is a stochastic process in which the future state of a system depends on the current state of the system. The relationship can be expressed as follows:

$$P\{X_n=j \mid X_{n-1}=i, X_{n-2}=i-1 \dots X_0=i_0\} = P\{X_n=j \mid X_{n-1}=i\} \quad (1)$$

where X_n indicates the state of the system, at n th instant of time, and j represents one of the states of the Markov chain.

Markov chains are associated with a set of equations that help predict the future states of the system based on the initial state probabilities, and the state transition probabilities. The status of the system at a given instant of time is given by the n th instant probabilities of the Markov chain which are obtained by the following equations:

$$\mathbf{p}(n) = \mathbf{p}(0) * [\mathbf{P}]^n \quad (2)$$

$$\mathbf{p}(n) = \mathbf{p}(n-1) * [\mathbf{P}] \quad (3)$$

where $\mathbf{p}(n)$ represents the state probability vector for the n th instant, $\mathbf{p}(0)$ represents the initial state probability vector, $\mathbf{p}(n-1)$ represents the state probability vector for (n-1) instant, and $[\mathbf{P}]$ represents the state transition probability matrix.

The extended Gilbert model discussed in [1] uses the above set of equations for predicting the future state of the system, given the current state.

The semi-Markov process shown in Fig.1 has (m+1) states. The no loss state is represented by 0, and the remaining 'm' states represent the corresponding loss states. A randomly selected state 'i' represents 'i' consecutive losses. Depending on the current state and the state holding time, the host decides the next state. The queue can make a transition from state 'i' to either (i+1) or 0. No other transitions are possible. However, if 'i' is equal to 0, then this rule doesn't apply. That is, the host can make an immediate transition back to state 0 if it was previously in that state or it can jump to the state 1.

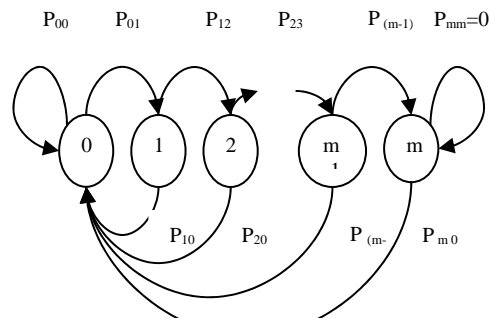


Fig. 1. A semi-Markov process with (m+1) states

The n^{th} instant probabilities of the semi-Markov process can be obtained from the following equation [3]:

$$p_i^{\text{sMm}}(n) = p_i(n) E[H_i] / \left(\sum_{j=0}^{\alpha} p_j(n) E[H_j] \right) \quad (4)$$

where, $p_i^{\text{sMm}}(n)$ represents the i^{th} element of the limiting state probability vector that indicates the probability of the semi-Markov process being in the i^{th} state at n^{th} instant of time.

Simulations were run on Matlab. A flow of identical packets was allowed to pass through a first-in-first-out (FIFO) queue. The packet arrival was assumed to follow Poisson distribution. By varying the mean inter-arrival time of the packets, the n^{th} instant probabilities for the semi-Markov process were determined using Eqs. (2), (3), and (4). These probabilities indicate the probability of the system being in a particular state at the equilibrium conditions. The value of 'n' was assumed to be 1000.

Following tabular column shows the limiting state probabilities for both the extended Markov chain and the semi-Markov process with a mean inter-arrival time of 15ms.

loss run	0	1	2	3
$p_i^{\text{eGm}}(n)$	0.5904	0.1208	0.0779	0.0531
$p_i^{\text{sMm}}(n)$	0.7464	0.1256	0.0548	0.0274

loss run	4	5	6	7
$p_i^{\text{eGm}}(n)$	0.0376	0.0287	0.0224	0.0170
$p_i^{\text{sMm}}(n)$	0.0155	0.0097	0.0063	0.0043

lossrun	8	9	10	11	12
$p_i^{\text{eGm}}(n)$	0.0139	0.0122	0.0095	0.0095	0.0070
$p_i^{\text{sMm}}(n)$	0.0031	0.0025	0.0017	0.0017	0.0010

Table.1 Limiting State Probabilities for semi Markov process

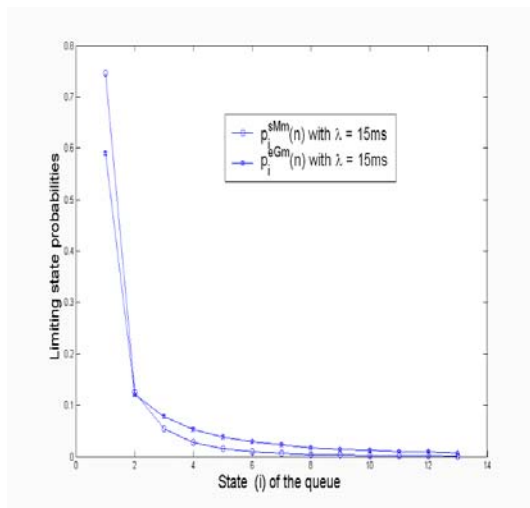


Figure 2: Limiting state probability Vs state of the node

It was observed that the packet loss can be controlled by increasing the inter-arrival time of the voice packets.

3. Conclusions

In this paper, the voice packet loss was modeled using the semi-Markov process. This process considers both the current state of the system, and the state holding time for predicting the limiting state probabilities [3], [4]. The effect of mean inter-arrival time on the limiting state probabilities was studied. It was observed that the limiting state probabilities decrease with an increase in the mean inter-arrival time. Therefore, the packet loss can be controlled by increasing the inter-arrival time of the voice packets.

4. References

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