Chapter 1
Analysis and Modeling of QoS Parameters in VoIP Traffic

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ABSTRACT

Several parameters influencing the voice quality on IP networks; in particular the one way delay (OWD), jitter and packet loss have an important impact on the quality of service (QoS) of voice over Internet protocol (VoIP). These parameters are intricately related to each other and can be used to configure other parameters to optimum values in order to afford good levels of QoS. Therefore, it is necessary to characterize the IP traffic nature, and implement adequate models of these QoS parameters in order to reliably evaluate the voice quality and design VoIP applications with reconfigurable parameters.

In this chapter, the jitter and packet loss behavior of VoIP traffic is analyzed by means of network measurements and simulations results. As result of these analyses, a detailed characterization and accurate modeling of these QoS parameters are provided.

Our studies have revealed that VoIP jitter can be modeled by self-similar processes, and through a decomposition based on Haar wavelet it is shown a possible reason of the presence of long range dependence (LRD) in VoIP jitter. On the other hand, we used a description of VoIP packet loss based on microscopic and macroscopic packet loss behaviors, where these behaviors can be modeled by 2-state and 4-state Markov chains, respectively. Besides, the distributions of the number of consecutive received and lost packets (namely gap and burst, respectively) are modeled from the transition probabilities of 2-state and 4-state Markov chains. Based on the above mentioned description, we presented a methodology for simulating packet loss and proposed a new model that allows to relate the Hurst parameter (H) with the packet loss rate (PLR). These models can be used by other researchers as input to problems related

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to the design of VoIP applications, performance evaluation of IP networks, and the implementation of QoS mechanisms on convergent networks.

INTRODUCTION

In the last years, VoIP has became the most attractive and important application running over Internet, poised to replace the Public Switched Telephone Network (PSTN) in the future. There are several advantages in the case of voice transmission using VoIP technology: the reduced communication cost, the use of joined IP infrastructure, the use of multimedia applications, etc. However, the current Internet provides best-effort services and cannot guarantee the *QoS* of real-time multimedia applications, such as VoIP.

To achieve the satisfactory voice quality, the VoIP networks must be designed by using correct traffic models. In order to implement adequate traffic models it is necessary to study the traffic characteristics by means of network measurements of the main *QoS* parameters, such as One Way Delay (*OWD*), jitter and Packet Loss Rate (*PLR*).

Different models are used to describe different types of traffic. One of the models which have been widely applied in classical teletraffic modeling is the Poisson model. However, the IP networks traffic has different characteristics and the Poisson approximation will be acceptable only under particular conditions. Empirical studies showed that IP traffic exhibits self-similar and LRD (Lelan et al, 1994; Park & Willinger, 2000; Sheluhin et al, 2007), i.e., the autocorrelation function approaches zero very slowly in comparison with the exponential decay characterizing SRD traffic. Long range dependent traffic produces a wide range in traffic volume away from the average data rate. This great variation in the traffic volume leads to buffer overflow and network congestion that result in packet loss and jitter, which directly impact on the quality of VoIP applications.

Amongst the different quality parameters, packet loss is the main impairment which makes the VoIP perceptually most different from the PSTN. On Internet, packet losses occur due to temporary overloaded situations. Packet losses are bursty in nature and exhibit a finite temporal dependency (Yajnik et al, 1999) due to the multiplexing policy on the shared resources such as bandwidth and buffer through the transmission paths in the network. So, if packet n is lost then normally there is a higher probability that packet $n + 1$ will also be lost. Consequently, there is a strong correlation between consecutive packet losses, resulting in a bursty packet loss behavior. The most generalized model to capture this temporal dependency is a finite Markov chain (ITU-T Recommendation G.1050, 2005). The objective of packet loss modeling is to characterize its probabilistic behavior, because an accurate model of the packet loss is required to design effective schemes for packet loss recovery.

Motivated by such concerns, we studied the jitter and packet loss behavior of VoIP traffic by means of networks performance measurements and simulations results. As result of these studies, it is shown that VoIP jitter can be modeled by self-similar processes with LRD or SRD (Toral et al, 2009). Through a decomposition based on Haar wavelet it is observed the components behavior of VoIP jitter as a function of packet loss and, by means of this decomposition it is shown a possible reason of the LRD presence in VoIP jitter.

Furthermore, we used a description of VoIP packet loss based on microscopic and macroscopic packet loss behaviors. The microscopic packet loss behavior can be represented by a *2*-state Markov chain in order to model the dependencies between packet losses. Correspondingly, the macroscopic packet loss behavior can be represented by *4*-state Markov chain. Here, substates represent phases of a given microscopic loss behavior. Ideally, an n-state model is required in order to capture the