Simulation and Modeling of Packet Loss on α-Stable VoIP Traffic

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Abstract: - In this paper, through an extensive analysis it is shown that VoIP traffic jitter exhibits heavy-tail characteristics, where α-stable distribution particularly gives the best goodness of fit; this fact has serious implications on the design of de-jitter buffer size. On the other hand, we investigate the packet loss effects on the VoIP jitter, and present a methodology for simulating packet loss on VoIP jitter traces with α-stable characteristics. In order to represent the packet loss process, the two state Markov model or Gilbert model is used. We proposed a new model for α-stable VoIP traffic, this model are based on voice traffic measurements, and allows to relate the α parameter and packet loss rate. We find that the relationship between α parameter and packet loss rate obeys a power-law function with three fitted parameters.

Key-Words: - VoIP, QoS, Packet Loss Rate, Jitter, Heavy-Tail Distributions, α Parameter, De-Jitter Buffer, Two-State Markov Model

1 Introduction

Voice over IP (VoIP) is now available on many IP networks carriers in the world with lower cost compared to Public Switched Telephone Network (PSTN). However, current IP networks only offers best-effort services and were designed to support non-real-time applications. VoIP demands strict quality of services (QoS) levels and real-time voice packet delivery. The QoS level of VoIP applications depends on many parameters; in particular, one-way-delay (OWD), jitter and packet loss have an important impact.

These parameters are complicatedly related to each other and affect voice quality. It is difficult to design and configure every parameter to optimum value and meet voice quality objectives, while maintaining efficient usage of network resources. Therefore it is necessary to implement adequate traffic models to evaluate the voice quality.

Packet losses are commonplace over the IP networks, and can severely affect the quality of VoIP applications. Basically, three reasons may account for voice packet losses: transmission errors, packet discarded at the network routers and at the de-jitter buffer. Packet loss is bursty in nature and exhibits a finite temporal dependency [1-2], i.e, the probability that the current packet is lost is dependent of whether the past packets have been received or lost. Specifically, if a lost packet is represented by the symbol one and a received packet by the symbol zero, then the packet loss process can be modeled as a finite memory binary random process, i.e., a binary Markov process [3]. The objective of packet loss modeling is to characterize its probabilistic behavior, because is relevant for the design and analysis of VoIP applications.

Since real-time applications cannot tolerate delay variations, in order to compensate jitter introduced by IP networks, a de-jitter buffer are used at the receiver side. An important design parameter, is the de-jitter buffer size, since it influences the packet loss probability and OWD. The de-jitter buffer size is the maximum amount of time a packet spends in the de-jitter buffer before being played out.

In this work we find that VoIP traffic jitter exhibits heavy-tail characteristics; this fact has serious implications on the design of de-jitter buffer size. If it is too small, as the probability of extremely large values occurrence is non-negligible, then many packets would miss the play out deadline, and thereby increasing the packet loss probability. On the other hand, if it is too large, then the OWD would increase. Therefore, is important to consider the heavy-tailed behavior when designing the de-jitter buffer size.

The main contributions of this paper are threefold:

- VoIP traffic jitter can be good modeled by α-stable distributions.
- A methodology for simulating packet loss on VoIP jitter traces.
- A new model for α-stable VoIP traffic.

The paper is organized as follows. In section 2, we provide some background on the QoS parameters of VoIP applications and the relationship between jitter and packet loss. VoIP traffic measurements are briefly presented in section 3. Section 4 presents the theory of α-stable distribution and a heavy-tail approximation of VoIP jitter. A new model for α-stable VoIP traffic is proposed in section 5. In section 6 simulation results are discussed. Section 7 concludes the paper.

2 QoS Parameters and their Relationships

Several parameters influencing voice quality on IP networks may be expressed in terms of delays and packet loss rate (PLR). OWD and jitter are the most critical parameters influencing voice quality, though excessive PLR can dramatically decrease the voice quality perceived by users of VoIP applications.

2.1 Jitter

When packets are transmitted from source to destination over IP networks, they may experience different delays. The packet Inter-Arrival Time (IAT) on the receiver side is not constant even if the packet Inter-Departure Time (IDT) on the sender side is constant. As a result, packets arrive at the destination with varying delays (between packets) referred to as jitter. The jitter is measured according to RFC 3550 [4], this is illustrated in Fig.1.

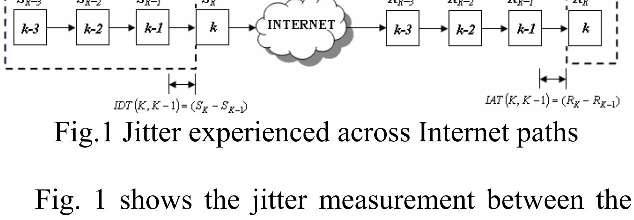

Fig.1 Jitter experienced across Internet paths

Fig. 1 shows the jitter measurement between the sending packets and the receiving packets. If S_k is the RTP timestamp for the packet *k* of size *L*, and R_k is the arrival time in RTP timestamp units for packet *k* of size *L*. Then for two packets *k* and *k-1*, $J^k(L)$ may be expressed as:

$$J^k(L) = (R_K - S_K) - (R_{K-1} - S_{K-1}) \quad (1)$$

$$IAT(K, K-1) = J^k(L) + IDT(K, K-1) \quad (2)$$

where $J^k(L)$ is the difference between the OWD of two consecutive packets, *k* and *k-1*; $IDT(K, K-1) = (S_K - S_{K-1})$ is the inter-departure time (in our experiments, IDT= {10ms, 20ms, 40ms, and 60ms}) and $IAT(K, K-1) = (R_K - R_{K-1})$ is the inter-arrival time or arrival jitter for the packets *k* and *k-1*. In the current context, it is referred to as jitter.

2.2 Packet Loss Rate

There are two main transport protocols used on IP networks, UDP and TCP. While UDP protocol does not allow any recovery of transmission errors, TCP include some error recovery processes. However, the voice transmission over TCP connections is not very realistic. This is due to the requirement for real-time (or near real-time) operations in most voice related applications. As a result, the choice is limited to the use of UDP which involves packet loss problems.

On the other hand a number of studies have shown that VoIP packet loss is bursty in nature and exhibits temporal dependency [1-2]. So, if packet *n* is lost then normally there is a higher probability that packet *n + 1* will also be lost. The most generalized model to capture temporal dependency, is a finite Markov chain [3]. Because of its simplicity and effectiveness, a two state Markov model or Gilbert model is often used to simulate packet loss patterns. Fig. 2 shows the state diagram of this 2-state Markov model.

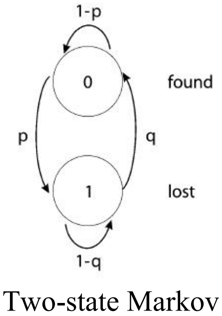

Fig. 2 Two-state Markov model

In this model, one of the states (state 1) represents a packet loss and the other state (state 0) represents the case where packets are correctly transmitted or found. The transition probabilities in this model, as shown in Fig. 2, are represented by p and q. In other words, p is the probability of going from state 0 to state 1, and q is the probability of going from state 1 to state 0.

The probability that n consecutive packets are lost is given by $p(1-q)^{n-1}$. If $(1-q) > p$, then the probability of losing a packet is greater after having already lost a packet than after having successfully received a packet. This is generally the case in data transmission on the Internet where packet losses occur as bursts.

Different values of p and q representing different packet loss and network conditions that can occur on the Internet.

In equation (3), b corresponds to the average burst length.

$$PLR = \frac{p}{p+q} \qquad b = \frac{1}{q} \tag{3}$$

2.3 Packet Loss Effects on the VoIP Jitter

The successive voice packets are transmitted at a constant rate, where the voice data rate is equal to the packetization interval or voice data length (i.e. 10ms, 20ms, 40ms and 60ms). However, when voice packets are transported over IP networks, they may experience delay variations and packet loss. On the other hand, in the measurements it is observed that packet loss has serious implications on the VoIP jitter.

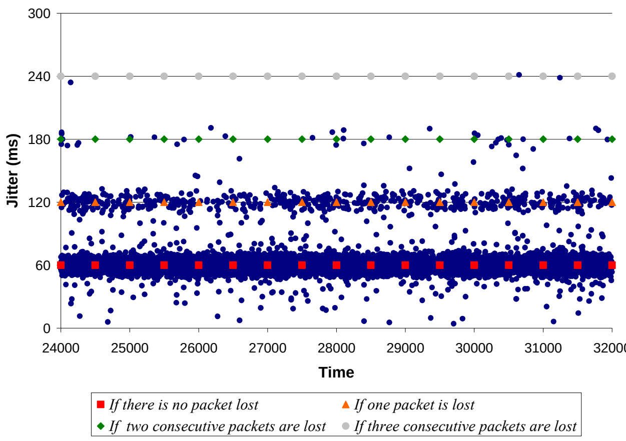

Fig. 3 Packet loss effects on VoIP jitter

The equation (2) describes the VoIP jitter for the packets k and k-1. From this equation can be found a relationship between jitter and packet loss. If the packet k-1 is lost, $IAT(K, K-2) = J^k(L) + (2)IDT$, therefore, if n consecutive packets are lost, then:

$$IAT(K, K-n-1) = J^k(L) + (n+1)(IDT) \tag{4}$$

were $J^k(L)$ is the difference between the OWD of two consecutive packets that arrive in the receiver side. This behavior is illustrated in Fig. 3, where a voice data length equal to 60ms is used.

Therefore, the equation (4) describes the packet loss effects on the VoIP jitter.

3 Measurements

The measurements corresponding to the data traces used in this work are shown in Table 1.

Table 1 Description of used VoIP jitter traces

Data Set	***Measurement Periods***	***Total Number of Traces***	***CODEC-Voice Data Length(ms)***
Set 1	Sep/07/2007, 10:00am-04:00pm	24 Jitter Traces	G.711-10ms G.711-20ms G.711-40ms G.711-60ms
Set 2	Sep/10/2007, 10:00am-04:00pm	24 Jitter Traces	G.729-10ms G.729-20ms G.729-40ms G.729-60ms
Set 3	Sep/11/2007, 10:00am-04:00pm	24 Jitter Traces	G.711-10ms G.711-20ms G.729-10ms G.729-20ms
Set 4	Sep/12/2007, 10:00am-04:00pm	24 Jitter Traces	G.711-40ms G.711-60ms G.729-40ms G.729-60ms

In this table, can be seen that VoIP jitter traces were collected in the following way:

- The measurement periods were 60 minutes (call duration time).
- For each measurement period (an hour), four data traces were obtained and four different CODEC configurations were used.

For better references of the used data sets in this paper, see [5].

4 Heavy-Tail Analysis

4.1 Mathematical Background

Distribution with 'Heavy-Tail' (DHT): A random variable (r.v.) X has a 'heavy-tail' distribution if:

$$P[X > x] \sim \frac{1}{x^{\alpha}}; \quad x \to \infty; \quad 0 < \alpha < 2 \tag{5}$$

where α is called the 'tail' index. Note that, heavy-tail distribution decays slower than exponential function. It is known that a heavy-tail r. v. has infinite variance, also when $0 < \alpha \leq 1$ its mean is infinite [6].

α-stable Distribution: A r. v. X is said to have an α-stable distribution if there are parameters $0 < \alpha \leq 2$, $a \geq 2$, $-1 \leq \beta \leq 1$, and $b \in \Re$, such that its characteristic function has the following form [7]:

$$\Phi(\omega) = E\left[e^{j\omega X}\right] = \exp\left\{jb\omega - |a\omega|^{\alpha}\left[1 - j\beta \operatorname{sgn}(\omega)\theta(\omega,\alpha)\right]\right\} \quad (6)$$

where:

$$\theta(\omega,\alpha) = \begin{cases} \tan\left(\frac{\alpha\pi}{2}\right); & \alpha \neq 1 \\ -\frac{2}{\pi}\ln|\omega|; & \alpha = 1 \end{cases} \quad (7)$$

In equation (6), α is the stability index; β, the skewness parameter; a, the scaling parameter and b, the shift parameter.

4.2 Heavy-Tail Approximation of VoIP Jitter

In order to evaluate if a given set of empirical data traces follows a particular distribution, the percentile-percentile plot (P-P Plot) is used. The empirical distribution of VoIP jitter traces are compared with α-stable, Laplace and t-Student distributions, see Fig. 4. For the α parameter estimation, the Nolan's Matlab toolbox [8] was used.

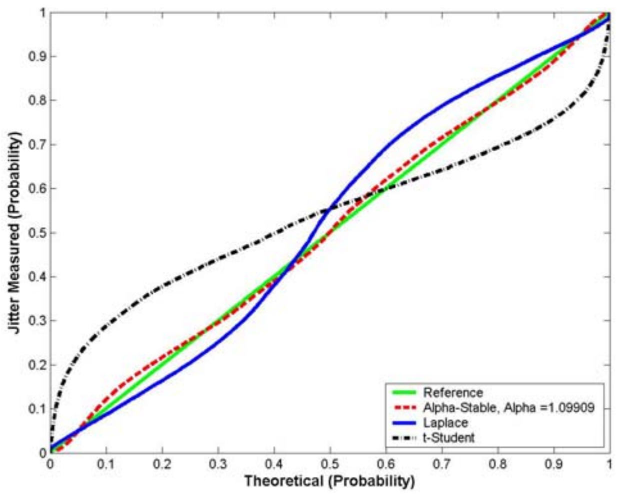

Fig. 4 P-P Plot for a VoIP jitter trace: α-stable, Laplace and t-Student distributions

The Fig. 4 shows that the α-stable distribution gives the best goodness of fit for the empirical distribution of VoIP jitter traces.

The differences between the empirical distribution and the theoretical distribution are measured in terms of MSE. The α-stable model achieves $MSE = 2.02 \cdot 10^{-4}$, more than 4 times better than Laplace model ($MSE = 9.72 \cdot 10^{-4}$) and more than twenty two times better than t-Student model ($MSE = 4.46 \cdot 10^{-3}$). This analysis shows that α-stable model is the most suitable to approximate the VoIP jitter; which means that extremely large values of VoIP jitter occur with non-negligible probability. It is described analytically by equation (8):

$$P[Jitter > x] \sim \frac{1}{x^{\alpha}}; \quad x \to \infty; \quad 0 < \alpha < 2 \quad (8)$$

In order to transmit voice requiring real-time delivery over a packet network, an important design parameter, is the de-jitter buffer size, since it influences the packet loss probability and OWD. The de-jitter buffer size is the maximum amount of time a packet spends in the de-jitter buffer before being played out. On the other hand, heavy-tailed behavior on VoIP jitter has serious implications on the design of de-jitter buffer size:

- If it is too small, as the probability of extremely large values occurrence is non-negligible, then many packets would miss the play out deadline, and thereby increasing the packet loss probability.
- If it is too large, then the OWD would increase.

Therefore, there is a trade-off between packet loss and OWD when it is designed the de-jitter buffer size and is important to consider the heavy-tailed behavior of VoIP jitter, as is expressed in equation (8).

5 Simulation and Modeling of Packet Loss

5.1 Methodology for Simulating Packet Loss

Let $X = \{X_t : t = 1,...,N\}$ be a VoIP jitter trace with α-stable distribution, α parameter $0 < \alpha_0 < 2$ and PLR PLR_0.

In order to represent the packet loss process or packet loss pattern, the two-state Markov model (Gilbert model) is used. The packet loss pattern is represented as a binary sequence $P = \{P_t : t = 1,...,wN\}$, where $P_t = 1$ means a packet loss, $P_t = 0$ means a received packet correctly and $w = 0.1,0.2,...,1$. In this model, different values of *p* and *q* define different packet loss patterns. We applied J different packet

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loss patterns over a time window W_l^u of X to simulate packet loss. The relationship between jitter and packet loss from equation (4) is used to apply the packet loss patterns to X by means of the algorithm shown in Table 2.

As it is well recognized that on Internet packet losses occur in bursts, in order to represent different packet loss bursts levels, various time windows W_l^u of size wN are used.

$$W_l^u = \left\{ \begin{array}{ll} X_l, X_{l+1}, ..., X_u : l = 1,2,..., N - \lfloor wN \rfloor + 1 & \\ u = l + \lfloor wN \rfloor - 1 \quad l < u & \end{array} \right\} \tag{9}$$

where X_l and X_u are the *l-th* and *u-th* element of time series X and represent the window beginning and ending, respectively.

Table 2 Algorithm for simulating packet loss:
A) Generating packet loss pattern
B) Applying packet loss pattern

A)	***B)***

FOR $n=1$ *to* $l-1$
$P[n]=0$
END FOR
 	
FOR $n=l$ *to* u
IF (*packet was lost*)
$P[n]=1$
ELSE
$P[n]=0$
END IF
END FOR
 	
FOR $n=u+1$ *to* N
$P[n]=0$
END FOR
	FOR $n=2$ *to* N
	IF ($P[n]=1$)
	$X[n]=X[n]+X[n-1]$
	END IF
	END FOR
	
	$i=1$
	FOR $n=2$ *to* N
	IF ($P[n] \neq 1$)
	$\hat{X}[i]=X[n-1]$
	$i=i+1$
	END IF
	END FOR

By means of the above algorithm the new time series $\hat{X}^j$ are obtained, where $j = 0,1,2,...J-1$. For each $\hat{X}^j$ the PLR and the α parameter were calculated, and the function $f_w(PLR_j, \alpha_j)$ was generated.

5.2 Proposed Model

From our simulations, we found that the relationship between α parameter and PLR can be modeled by a power-law function, characterized by three fitted parameters, $0 < \hat{\alpha}_0 < 2$, $a < 0$ and $b > 0$, as the following:

$$\alpha_M = \hat{\alpha}_0 + a(PLR)^b \tag{10}$$

where α_M is the α parameter of the found model, $\hat{\alpha}_0$ is the α parameter when $PLR = 0$.

6 Simulation Results

In this section, applying the methodology proposed in section 5, simulation results are presented. The simulations are accomplished over VoIP jitter traces corresponding to Table 1.

Fig. 5 illustrates the relationships between PLR and α parameter. The functions family $f_w(PLR_j, \alpha_j)$, is result to apply "J" packet loss patterns to time series X_t over a time window *"w"*. The time series X_t represents a VoIP jitter trace of the data sets described in Table 2. In this figure, each point of the function $f_w(PLR_j, \alpha_j)$ represents a new time series $\hat{X}^j$. $f_w(PLR_j, \alpha_M)$ is the function of the found model.

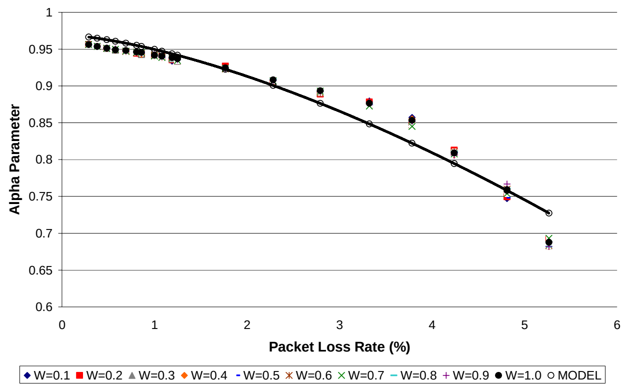

Fig. 5 Relationship between PLR and α parameter: $f_w(PLR_j, \alpha_j)$ vs. $f_w(PLR_j, \alpha_M)$

The difference between the function corresponding to simulation results $f_w(PLR_j, \alpha_j)$ and the function corresponding to the found model $f_w(PLR_j, \alpha_M)$, was quantified in terms of mean square error:

$$MSE = \frac{1}{PLR_{Max} - PLR_{Min}} \int_{PLR_{Min}}^{PLR_{Max}} \left[f_w(PLR_j, \alpha_M) - f_w(PLR_j, \alpha_j) \right]^2$$

Table 3 shows the fitted parameters and MSE between $f_w(PLR_j, \alpha_M)$ and $f_w(PLR_j, \alpha_j)$ corresponding for each time window.

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Table 3 Fitted parameters for Fig. 5

$f_w(PLR_j, \alpha_j)$	$\hat{\alpha}_0$	a	b	MSE
w=0.1	0.9693	-0.0198	1.5068	0.000274
w=0.2	0.9693	-0.0198	1.5033	0.000246
w=0.3	0.9693	-0.0198	1.5069	0.000272
w=0.4	0.9693	-0.0198	1.5058	0.000280
w=0.5	0.9693	-0.0198	1.5097	0.000240
w=0.6	0.9693	-0.0198	1.5066	0.000247
w=0.7	0.9693	-0.0198	1.5052	0.000198
w=0.8	0.9693	-0.0197	1.5036	0.000280
w=0.9	0.9693	-0.0196	1.5082	0.000273
w=1	0.9693	-0.0197	1.4996	0.000253

In Fig. 5 and Table 3 it is shown that the relationships between α parameter and packet loss can be good modeling by means of the power-law function proposed in section 5.

7 Conclusions

Several factors influencing voice quality on IP networks. These parameters are complicatedly related to each other and it is difficult to design and configure every parameter to optimum value and meet voice quality objectives, while maintaining efficient usage of network resources. Therefore it is necessary to implement adequate traffic models to evaluate the voice quality.

In this paper we found that VoIP jitter can be properly modeled by means of α-stable distributions; this fact has serious implications on the design of de-jitter buffer. Therefore, is important to consider the heavy-tailed behavior of VoIP jitter when designing the de-jitter buffer size.

On the other hand, we have presented a methodology for simulating packet loss on VoIP jitter traces. In this methodology the packet loss effects on VoIP jitter and the two state Markov model are used. Based on the above methodology, we have proposed a new model for α-stable VoIP traffic.

The new model are based on voice traffic measurement and allowed to relate two important parameters, the α parameter and PLR. We found that α parameter is related to PLR by a power-law with three fitted parameters. Simulation results show the effectiveness of our model in terms of MSE.

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