Simulation and Modeling of Packet Loss on VoIP Traffic: A Power-Law Model

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Abstract: - In this paper, through an extensive analysis it is shown that VoIP traffic jitter exhibits self-similar and heavy-tail characteristics. From this analysis, we observed that α-stable distribution particularly gives the best goodness of fit; this fact has 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. In order to represent the packet loss process, the two state Markov model or Gilbert model is used. We proposed two new models for VoIP traffic; these models are based on voice traffic measurements, and allow relating the Hurst parameter and α parameter with the packet loss rate. We found that the relationship between these parameters and packet loss rate obeys a power-law function with three fitted parameters.

Key-Words: - VoIP, QoS, Packet Loss Rate, Jitter, Heavy-Tail Distributions, Self-Similar, α Parameter, Hurst 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 [1]; 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 [2-3], 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 [4]. The objective of packet loss modeling is to characterize its probabilistic behavior, because is relevant for the design and analysis of VoIP applications.

On the other hand, we find that VoIP Jitter exhibit self-similar characteristics. The degree of self-similarity is expressed by H parameter, called Hurst parameter. The fact that network traffic exhibits self-similarity characteristics means that it is bursty [5] at a wide range of time scales and this behavior has negative impact on network performance. Therefore, it is important to consider models that capture this behavior for the design and performance analysis of computer networks.

Many real-time applications are very sensitive to 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 function of the maximum amount of time a packet spends in the buffer before being played out.

In this work, we find that VoIP traffic jitter exhibits heavy-tail characteristics. This fact has 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 self-similar processes and α-stable distributions.
- A methodology for simulating packet loss on VoIP jitter traces.
- Two new models for VoIP traffic.

The paper is organized as follows. In section 2, we provide the related background of the QoS parameters for VoIP applications and the relationship between jitter and packet loss. VoIP traffic measurements are presented in section 3. Section 4 presents the theory of self-similar processes, α-stable distribution and an analysis of VoIP jitter traces that exhibits these behaviors. Two new models for 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 [6], this is illustrated in Fig.1.

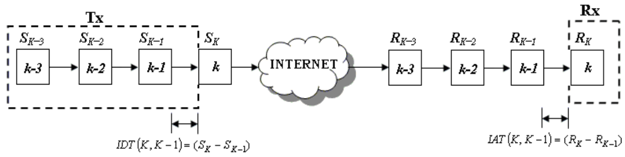


Fig.1 Jitter experienced across Internet paths

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 [2-3]. 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 [4]. 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.

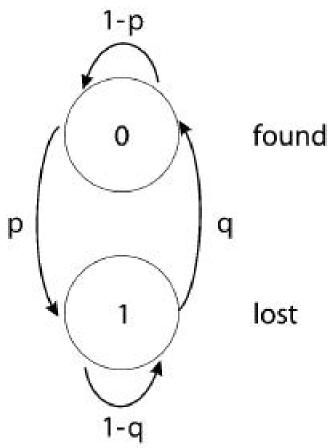


Fig. 2 Two-state Markov model

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 a lost packet than after having successfully a received one. 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.

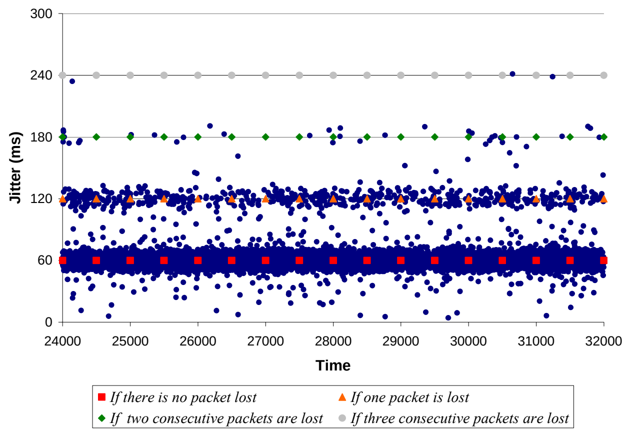


Fig. 3 Packet loss effects on VoIP jitter

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

Voice over IP demands strict quality of service levels. However, the current Internet only offers best-effort services due to its shared nature and cannot guarantee the required QoS. VoIP is susceptible to suffer impairments, which result in voice quality degradation. Therefore, it is necessary, to monitor voice quality constantly, and to cope with possible voice quality degradation. To do this, active measurement and passive measurement can be considered.

In the active measurements, VoIP traffic was generated by establishing test calls with the Alliance FXS VoIP application [7]. Alliance FXS is a system developed at CTS CINVESTAV GDL composed by a PCI board and application software that allows connecting regular telephone sets to the IP network. Each board allows having four extensions. Multiple Alliance FXS boards can be installed in a single PC, as shown in Figure 4.

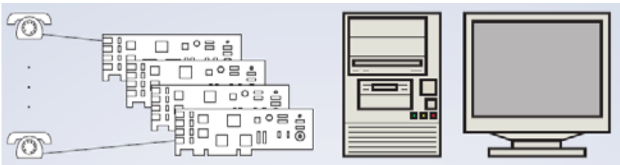


Fig. 4 Alliance FXS

The main characteristics of Alliance FXS are the following:

- PCI board (26.2cm x 12.0cm)
- 4 RJ-11 ports for analog telephone sets
- Each port supports lines up to 4 kilometers long
- H.323 architecture [8]
- G.711-A Law [9] and G.729 [10] hardware compression
- ITU-T G.165 and G.168 echo canceller on the four ports
- DTMF detection compliant with ITU-T Q.24, Bellcore GR-30 CORE and BAPT 223 ZV5
- Adjustable gain in reception from -10 dB to +2 dB
- Adjustable gain in transmission from +10 dB to -5 dB
- Ring signal compatible with ANSI/EIA/TIA-464-A-1989
- ITU-T V.23 and Bell 202 Caller Id generation

In the passive measurements, we realized the capture of VoIP traffic using Wireshark [11] to obtain a set of data traces. The subject of these measurements is to gather traffic patterns of RTP packets such as: Jitter, sequence number of the packets and post-processing analysis.

Table 1 Relationship between the voice data length and samples size

Data Traces Length (samples)	*Voice Data Length (ms)*	*Voice Data Length (Bytes)*	
		G.711	*G.729*
360,000	10	80	10
180,000	20	160	20
90,000	40	320	40
60,000	60	480	60

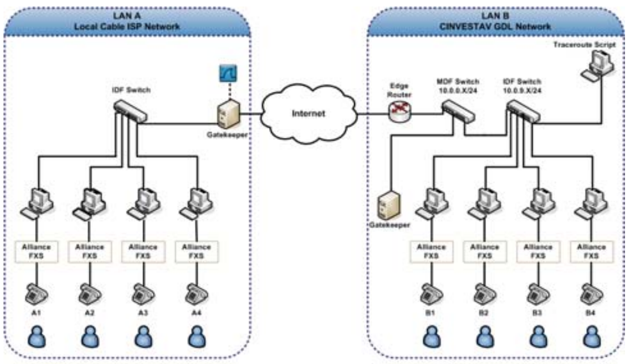
For the collected data sets (Table 2), we used the parameters showed in Table 1. In this table it is observed the relationship between the voice data lengths and the samples size for one hour of measurement. The voice codec G.711 and G.729 with voice data length of 10, 20, 40 and 60 ms are used.

The test scenario where the VoIP traffic was measured consists of two different LANs (different ISPs), interconnected by the Internet backbone [12]:

- LAN "A" - Local Cable ISP network (Link Speed-3MB)
- LAN "B" - CINVESTAV GDL network (Link Speed-2MB)

Figure 5 shows a typical H.323 architecture [8], composed of two zones interconnected via Internet. Each zone consists of a single H.323 gatekeeper (GK) [13] which acts as the administrator of the zone [14], and a number of H.323 terminal endpoints (TEs), interconnected via a LAN. The zone "A" is composed of the endpoints A1, A2, A3, and A4, this zone is administrated for the gatekeeper "A". In this zone, the network protocol analyzer Wireshark was installed for collecting the data traces. On the other hand the zone "B" is composed of the endpoints B1, B2, B3 and B4, the gatekeeper "B" and a workstation where is monitoring the routes that follows the test calls.

In order to make calls between any endpoints, each endpoint has installed an Alliance FXS PCI card and a conventional cord phone.



Set	*A1/B1*	*A2/B2*	*A3/B3*	*A4/B4*
Set 1	G.711-10ms	G.711-20ms	G.711-40ms	G.711-60ms
Set 2	G.729-10ms	G.729-20ms	G.729-40ms	G.729-60ms
Set 3	G.711-10ms	G.711-20ms	G.729-10ms	G.729-20ms
Set 4	G.711-40ms	G.711-60ms	G.729-40ms	G.729-60ms

Fig. 5 Alliance FXS

The measurements corresponding to the data traces used in this work are shown in Table 2.

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

- Four simultaneous test calls were established between A1/B1, A2/B2, A3/B3 and A4/B4 endpoints, see Figure 5.
- The measurement periods were 60 minutes for each test call (call duration time)
- Four different configuration sets for the endpoints were used, see Figure 5.
- For each measurement period (an hour), four jitter and sequence number data traces were obtained

Table 2 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

4 Self-Similar and Heavy-Tail Analysis

4.1 Mathematical Background

Self-Similar Processes: Traffic processes are said to be self-similar if certain property of the processes is preserved with respect to scaling in space and/or time [15-18].

Considering a discrete time stochastic process or time series $X_t = (X_t; t \in \mathrm{N})$ with mean μ_X, variance σ_X^2, autocorrelation function $r(k)$ and autocovariance function (ACV) $\gamma(k)$, $k \geq 0$; where X_t can be interpreted as the traffic volume at time instance t, or Jitter.

To formulate the phenomenon of scale invariance, the aggregated process is defined as

$$X_k^{(m)} = \left(X_k^{(m)}; k \in \mathrm{N}\right) \tag{5}$$

where $X_k^{(m)}$ is obtained by averaging the original series X_t over non-overlapping blocks of size m, and each term $X_k^{(m)}$ is given by

$$X_k^{(m)} = \frac{1}{m} \sum_{i=(k-1)m+1}^{km} X_i; \quad k = 1,2,3,... \tag{6}$$

Here, X_t is self-similar $(H - ss)$ with self-similarity parameter H, i.e. Hurst Parameter $(0 < H < 1)$ if:

$$X_k^{(m)} \overset{d}{=} m^{H-1} X_t \tag{7}$$

where $\overset{d}{=}$ denotes equality in distribution.

Let $\gamma^m(k)$ denote the autocovariance function of $X_k^{(m)}$. The process X_t is called exactly second order self-similar with Hurst parameter H if

$$\gamma(k) = \frac{\sigma_X^2}{2}\left((k+1)^{2H} - 2k^{2H} + (k-1)^{2H}\right) \tag{8}$$

for all $k \geq 1$. X_t is called asymptotically second-order self-similar if

$$\lim_{m \to \infty} \gamma^m(k) = \frac{\sigma_X^2}{2}\left((k+1)^{2H} - 2k^{2H} + (k-1)^{2H}\right) \tag{9}$$

Equations (8) and (9), express the fact than X_t and $m^{1-H} X_k^{(m)}$ are required to have exactly or asymptotically the same second-order structure. From equation (7) it follows that

$$\operatorname{var}\left(X_k^{(m)}\right) = \sigma_X^2 \cdot m^{2H-2} \tag{10}$$

Let $r(k) = \gamma(k)/\sigma_X^2$ denote the autocorrelation function. For $0 < H < 1$

$$r(k) \sim H(2H-1)k^{2H-2} \quad k \to \infty . \tag{11}$$

In particular, if $\frac{1}{2} < H < 1$, $r(k)$ asymptotically behaves as $ck^{-\eta}$ for $0 < \eta < 1$, where $c > 0$ is a constant, $\eta = 2 - 2H$ and $\sum_{k=-\infty}^{\infty} r(k) = \infty$. That is, the autocorrelation function decays slowly, which is the essential property that causes it to diverge. When $r(k)$ obeys a power-law, the corresponding stationary process X_t is called long range dependent (LRD). X_t is short range dependent (SRD) if the sum $\sum_{k=-\infty}^{\infty} r(k) < \infty$ does not diverge.

Following are some simple facts regarding the value of H and its impact on $\gamma(k)$.

- $\gamma(k) = \begin{cases} 1, & k = 0 \\ 0, & k \neq 0 \end{cases}$ for $H = 0.5$. This is the well-known property of white Gaussian noise.
-
- $\gamma(1) < 0$ for $0 < H < 0.5$.
-
- $\gamma(1) > 0$ for $0.5 < H < 1$.

Properties 2 and 3 are often termed antipersistent and persistent correlations, respectively.

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

$$P[X > x] \sim \frac{1}{x^{\alpha}}; \quad x \to \infty; \quad 0 < \alpha < 2 \qquad (12)$$

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 [15].

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 [19]:

$$\Phi(\omega) = E\left[e^{j\omega X}\right] = \exp\left\{jb\omega - |a\omega|^{\alpha}\left[1 - j\beta\,\mathrm{sgn}(\omega)\theta(\omega,\alpha)\right]\right\} (13)$$

where:

$$\theta(\omega,\alpha) = \begin{cases} \tan\left(\frac{\alpha\pi}{2}\right); & \alpha \neq 1 \\ -\frac{2}{\pi}\ln|\omega|; & \alpha = 1 \end{cases} \qquad (14)$$

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

4.2 Self-Similar Analysis

For the H parameter estimation at seven aggregation levels, ($m = \{2,4,8,16,32,64,128\}$) six methods, implemented in SelQos [20-22], were used. They are: R/S statistic (R/S), Absolute Moment (AM), Variance Method (VAR), Modified Allan Variance (MAVAR), Periodogram (PER), y Local Whittle (WHI).

Figure 6 shows the Hurst Parameter of the Jitter traces as a function of the aggregation level. We observe the phenomenon of scale invariance at different time scales. These results indicate that the analyzed VoIP Jitter traces have self-similar characteristics.

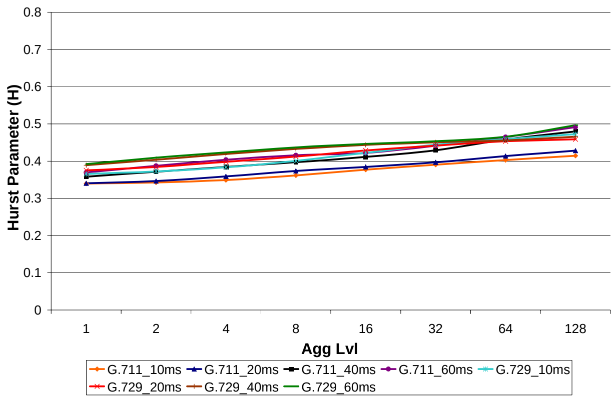


Fig. 6 Hurst parameter for different aggregation levels

The autocovariance function of representative VoIP Jitter traces is illustrates in Figure 7.

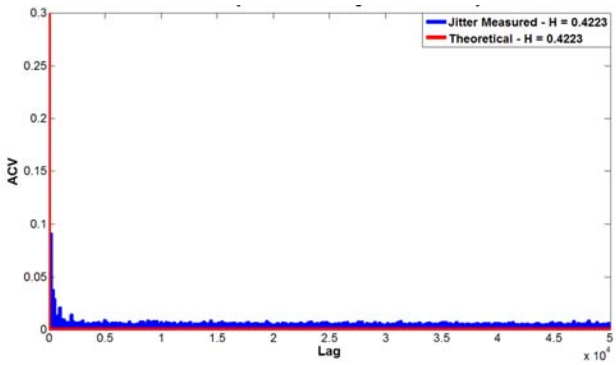


Fig. 7 Autocovariance function for VoIP Jitter traces

¡Error! No se encuentra el origen de la referencia. shows a comparison between the autocovariance function of a measured data trace with $H = 0.4223$ and the theoretical autocovariance function defined by equation (8). It can be observed that the autocovariance function of the measured data trace behaves similarly to the ideal model. This indicates that the VoIP Jitter traces exhibit memoryless property or SRD.

4.3 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. 8. For the α parameter estimation, the Nolan's Matlab toolbox [23] was used.

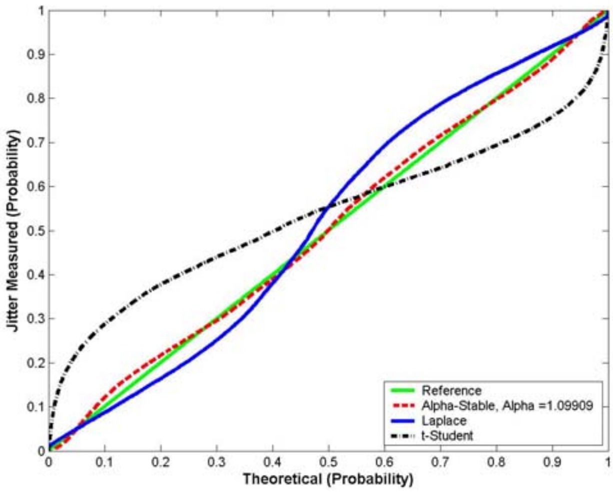


Fig. 8 P-P Plot for a VoIP jitter trace: α-stable, Laplace and t-Student distributions

The Fig. 8 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 (15):

$$P[Jitter > x] \sim \frac{1}{x^{\alpha}}; \quad x \to \infty; \quad 0 < \alpha < 2 \qquad (15)$$

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 function of the maximum amount of time a packet spends in the buffer before being played out. On the other hand, heavy-tailed behavior on VoIP jitter has 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 (15).

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 length *N,* self-similar (H parameter $0 < H_0 < 0.5$), α-stable distribution (α parameter $0 < \alpha_0 < 2$) and packet loss rate 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 success and $w = 0.1, 0.2, ..., 1$, is the burst level. In this model, different values of *p* and *q* define different packet loss patterns. We applied J different packet loss patterns over a time window W_l^u (see Eq. 16), of the trace X to simulate packet loss. The relationship between jitter and packet loss from equation (4) is used to apply the packet loss patterns to the trace X by means of the algorithm shown in Table 3.

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.

$$W_l^u = \left\{ \begin{array}{l} X_l, X_{l+1}, ..., X_u : l = 1,2,...,N - \lfloor wN \rfloor + 1 \\ u = l + \lfloor wN \rfloor - 1 \quad l < u \end{array} \right\} \qquad (16)$$

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 3 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, H parameter and the α parameter were calculated, and the functions $f_w(PLR_j, H_j)$ and $f_w(PLR_j, \alpha_j)$ was generated.

5.2 Proposed Models

From our simulations, we found that the relationship between H parameter and α parameter with the PLR can be modeled by a power-law function, characterized by three fitted parameters, as following:

- *Relationship between H and PLR:*

$$H_M = \hat{H}_0 + a_1 (PLR)^{b_1} \tag{17}$$

$0 < \hat{H}_0 < 0.5$, $a_1 > 0$ and $b_1 > 0$

- *Relationship between α and PLR:*

$$\alpha_M = \hat{\alpha}_0 + a_2 (PLR)^{b_2} \tag{18}$$

$0 < \hat{\alpha}_0 < 2$, $a_2 < 0$ and $b_2 > 0$

Where H_M and α_M are the H parameter and α parameter of the found models, respectively, $\hat{H}_0$ and $\hat{\alpha}_0$ are the H parameter and α parameter when $PLR = 0$, respectively.

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 2.

6.1 Relationship between H and PLR

Figure 9 illustrates the relationships between packet loss rate and Hurst parameter. The functions family $f_w(PLR_j, H_j)$, 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 $f_w(PLR_j, H_j)$ represents a new time series $\hat{X}^j$. The function $f_w(PLR_j, H_M)$ is the function of the found model.

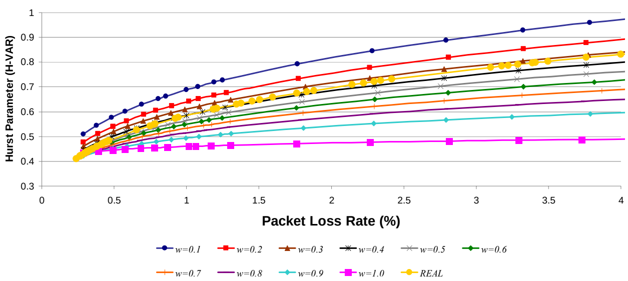


Fig. 9 Relationship between PLR and H parameter: $f_w(PLR_j, H_j)$ vs. $fw(PLR_j, H_M)$

The difference between the function corresponding to measurements results $f_w(PLR_j, H_j)$ and the function corresponding to the found model $f_w(PLR_j, H_M)$, was quantified in terms of mean square error:

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

$$i = 1,2,\ldots I$$

Table 4 shows the fitted parameters and MSE between $f_w(PLR_j, H_M)$ and $f_w(PLR_j, H_j)$ corresponding for each time window.

Table 4 Fitted parameters for Figure 9

$f_w(PLR_j, H_j)$	$\hat{H}_0$	a	b	MSE
w=0.1	0.0428	0.6457	0.2623	0.0144
w=0.2	0.0428	0.5963	0.2548	0.0032
w=0.3	0.0428	0.5673	0.2453	0.0004
w=0.4	0.0428	0.5431	0.2390	0.0003
w=0.5	0.0428	0.5213	0.2328	0.0019
w=0.6	0.0428	0.5069	0.2160	0.0047
w=0.7	0.0428	0.4901	0.2001	0.0091
w=0.8	0.0428	0.4724	0.1789	0.0155
w=0.9	0.1738	0.3194	0.2010	0.0261
w=1	0.0428	0.4168	0.0498	0.0540

6.2 Relationship between α and PLR

The same analysis is repeated for all VoIP Jitter traces corresponding to Table 2; we generated a functions family $f_w(PLR_j, \alpha_j)$. The results are summarized in Figure 10 and Table 5. Fig. 10 illustrates the relationships between PLR and α parameter. 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.

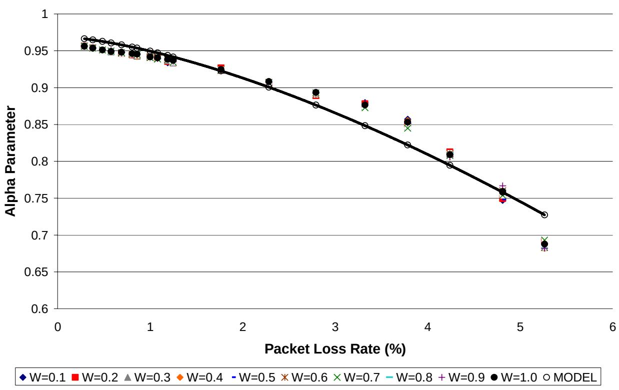


Fig. 10 Relationship between PLR and α parameter: $f_w(PLR_j, \alpha_j)$ vs. $f_w(PLR_j, \alpha_M)$

Table 5 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.

Table 5 Fitted parameters for Figure 10

$f_w(PLR_j, \alpha_j)$	$\hat{\alpha}_0$	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. 10 and Table 5 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 intricately 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 traces present heavy-tail and self-similar characteristics and can be properly modeled by means of self-similar processes and α-stable distributions. This facts has implications:

- The self-similar behavior on IP traffic has negative impact on network performance.
- 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, it is important to consider models that capture these behaviors for the design and performance analysis of computer networks and 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 VoIP traffic.

The new models are based on voice traffic measurement and allowed to relate three important parameters, the H parameter, the α parameter and PLR. We found that H parameter and α parameter is related with the 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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