Project	**IEEE 802.16 Broadband Wireless Access Working Group** <http://ieee802.org/16>
Title	**Downlink VoIP Packet Delay Jitter Model**
Date Submitted	**2007-09-10**
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Re:	IEEE 802.16m-07/031– Call for Comments on Draft 802.16m Evaluation Methodology Document
Abstract	This contribution proposes a network delay jitter model for the DL VoIP traffic modeling.
Purpose	To incorporate the proposed text changes into the Draft 802.16m Evaluation Methodology Document (IEEE C802.16m-07/080r3)
Notice	
Release	
Patent Policy	

Downlink VoIP Packet Jitter Model

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1. Introduction

Variable network delays cause VoIP packets to arrive at the base station transmitter with some delay jitter. In the presence of this jitter, VoIP packets are not necessarily available every 20 ms for transmission on the downlink. Robust VoIP design requires the ability to meet VoIP latency requirements in the presence of delay jitter, and hence, this contribution proposes a delay jitter model for the DL VoIP traffic modeling to be included in the Evaluation Methodology [1].

2. Network Delay Jitter

It is possible to define several types of delay jitter [2][3]:

a) Constant delay jitter. This is the roughly constant portion of the delay jitter, or in other words a fixed delay added to each packet.

b) Transient jitter. This is characterized by large delays that may occur for only a one packet. Examples of such causes of jitter are system packet scheduling, CPU congestion, routing table updates and timing drift.

c) Short term delay variation. This occurs due to changing routes and may be accompanied by an increase in packet to packet delay variation. This type of packet delay occurs for multiple packets. This type of delay jitter is also commonly associated with network congestion.

Sources of VoIP packet arrival jitter is well documented, and the reader if referred to [2][3] for further information.

The presence of delay jitter is a significant consideration in wireless system design. System design must be sufficiently robust to meet the delay requirements under considers VoIP delay jitter. This contribution proposes a simulation model based on studies from [4]-[6], and similar that proposed in [7].

3. Simulation Model

A suitable approximate model for the VoIP delay jitter arrival is to model the jitter delays with a Laplacian distribution, which is given by:

$$F(x) = \frac{1}{2} e^{-\frac{|x-\alpha|}{\beta}}, \qquad \text{for } x \le \alpha \qquad (1)$$

$$F(x) = 1 - \frac{1}{2} e^{-\frac{|x-\alpha|}{\beta}}, \qquad \text{for } x > \alpha \qquad (2)$$

and pdf given by:

$$f_x = \frac{1}{2\beta} e^{-\frac{|x-\alpha|}{\beta}} \qquad (3)$$

where α is the mean of the jitter process and $\beta = \sqrt{\sigma^2 / 2}$, where σ^2 is the variance. These delays are applied to the regular frame intervals for VoIP generation (which is commonly 20ms).

For the purposes of air interface evaluation, the contribution of constant delays are not important, and hence, α = 0 for purposes of this model in the evaluation methodology. Considering values for parameters and delays suggested in [2][4][5][7], a value of β = 5 ms is proposed for modeling delay jitter in the VoIP traffic model. In order to limit the complexity in simulating packet arrivals with delay jitter, we also suggest limiting the packet arrivals to range of -60 ms $\le x \le$ 60 ms.

4. References

[1] R. Srinivasan, J. Zhuang, L. Jalloul, R. Novak, J. Park, *Draft IEEE 802.16m Evaluation Methodology Document*, IEEE C802.16m-07/080r3, August 28, 2007.

[2] http://www.voiptroubleshooter.com/indepth/jittersources.html

[3] http://www.voipnow.org/2005/09/causes_of_jitte.html

[4] L. Zheng, L. Zhang, D. Xu, "Characteristics of network delay and delay jitter and its effect on voice over IP (VoIP)", in *Proceedings of the IEEE International Conference on Communications (ICC)*, June 2001, pp. 122-126.

[5] E. J. Daniel, C. M. White, K. A. Teague, " An inter-arrival delay jitter model using multi-structure network delay characteristics for packet networks," in *Proceedings of the 37th Asilomar Conference of Signals, Systems, and Computers*, vol. 2, November 2003, pp. 1738-1742.

[6] M. J. Karam, F. A. Tobagi, "Analysis of the delay and jitter of voice traffic over the internet", in *Proceedings of INFOCOM '2001*, April 2001, pp. 824-833.

[7] Z. Pi, E. Kwon, D. Kim, D. Kim, *FL VoIP packet delay jitter model*, 3GPP2 C30-20060719-006, July 2006.

5. Proposed Text

The following modifications are required to the current Evaluation Methodology [1]:

1) Modifying text on page 103, lines 6-7:

"…model is assumed to be updated at the speech encoder frame rate R=1/T, where T is the encoder frame duration (typically, 20 ms). Packets are generated at time intervals T+ τ, where τ is the network packet arrival delay jitter. During the active state, packets of fixed sizes are generated at these intervals, while the model is updated at regular frame intervals. ~~a regular interval~~."

2) Modifying text on page 104, line 40:

"Without header compression, an AMR payload of 33 bytes is generated in the active state every 20 $+ \tau$ ms and an AMR payload of 7 bytes is generated in the inactive state every 160 $+ \tau$ ms. "

3) Modify text on page 105, line 19:

"…fixed sizes will be generated at ~~a fixed~~ intervals of $T+\tau$ seconds, where T is the VoIP frame interval of 20 ms, and τ is the DL network delay jitter. For the reverse link, τ is equal to 0. As the range of the delay jitter is limited to 120 ms, the model may be implemented as packet arrivals at times $T+\tau'$ seconds, where $\tau' = \tau + 60$ ms and is always positive. ~~Hence, both the datagram size and datagram arrival intervals are fixed within a packet call~~. **Error! Reference source not found.** shows parameters associated with the VoIP traffic model."

4) Adding row to Table 31 at line 23 of page 105:

Component	Distribution	Parameters	PDF
Active state duration	Exponential	Mean = 1 second	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 1/Mean$
Inactive state duration	Exponential	Mean = 1.5 second.	$f_x = \lambda e^{-\lambda x}, x \ge 0$ $\lambda = 1/Mean$
Probability of transition from active to inactive state	N/A	0.02	N/A
Probability of transition from inactive to active state	N/A	0.0133	N/A
Packet arrival delay jitter (Downlink only)	Laplacian	β = 5 ms	$f_x = \frac{1}{2\beta} e^{-\frac{\lvert\tau\rvert}{\beta}}$, -60 ms $\le \tau \le$ 60ms