Project	**IEEE 802.16 Broadband Wireless Access Working Group** <http://ieee802.org/16>
Title	**Downlink VoIP Packet Delay Jitter Model**
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Re:	IEEE 802.16m-07/048– 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 IEEE 802.16m Evaluation Methodology Document (IEEE 802.16m-08/037r2)
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].

In Session #52, the DL VoIP jitter model was removed according to the resolution of comment 119L [2] in response to the call for comments on the Draft IEEE 802.16m Evaluation Methodology [3], as the Beta value of the VoIP jitter model was TBD. This contribution proposes to re-insert the DL VoIP Jitter model for system evaluations with an appropriate Beta value. This contribution contains text from a previously submitted DL VoIP Jitter model contribution [4] with enhanced description.

2. Network Delay Jitter

It is possible to define several types of delay jitter [5][6]:

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 [5][6] 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 [7]-[9], and is similar to that proposed in [10].

The net effect implementation of the DL VoIP delay jitter model is that packet arrivals will not be at regular 20 ms intervals, but rather randomly distributed about the 20ms intervals. For the purposes of measuring air interface delays, the delay measurement begins upon packet arrival at the base station, and hence, implementing the VoIP Jitter model does not impose any additional delays on the air interface. This is illustrated in the Figures 1 and 2 for cases with and without jitter.

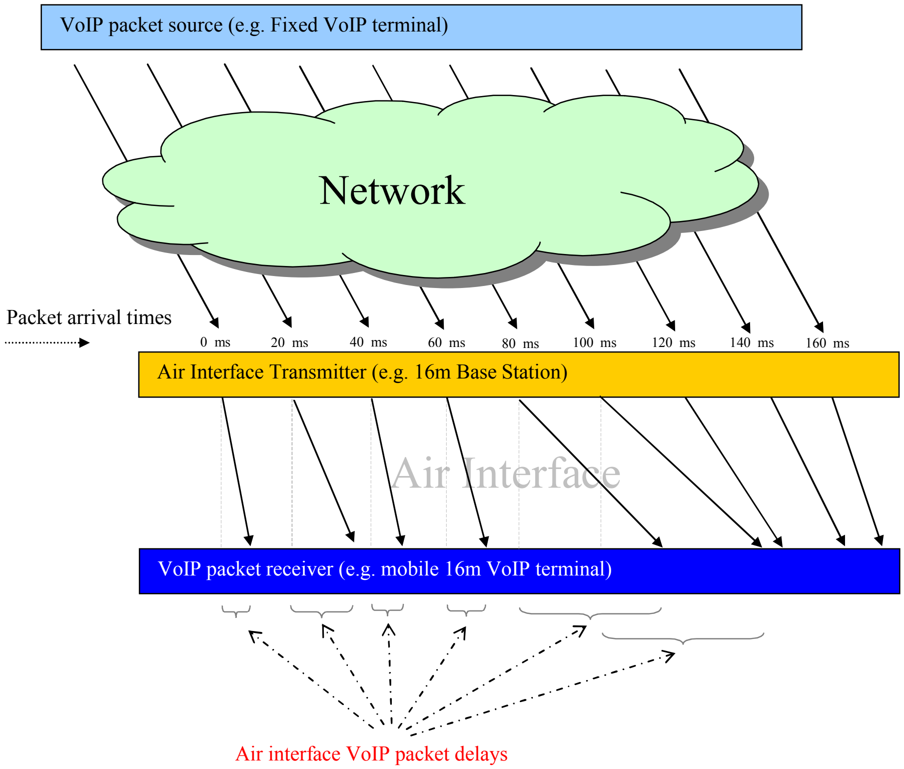

Figure 1. DL Model without Jitter

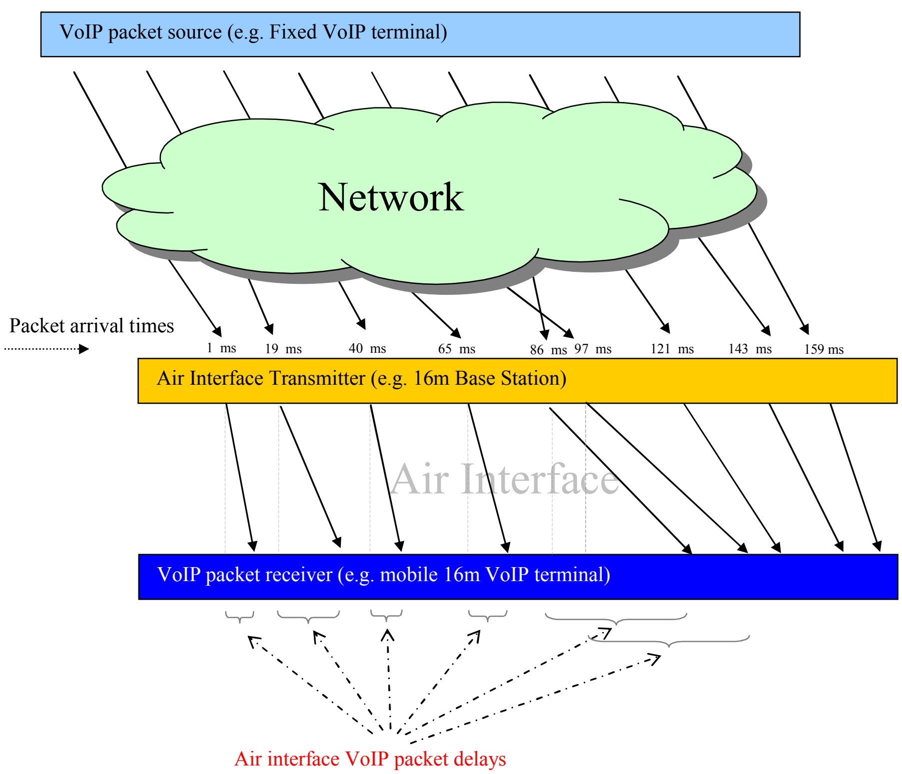

Figure 2. DL Model with Jitter

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:

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

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

and pdf given by:

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

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 [5][7][8][10], a value of β = 5.11 ms is proposed for modeling delay jitter in the VoIP traffic model. This Beta value corresponds to 98 % of the packets having delay jitter of +/- 20ms or less. In order to limit the complexity in simulating packet arrivals with delay jitter, we also suggest limiting the packet arrivals to a range of -80 ms ≤ x ≤ 80 ms.

4. Implementation

In order to implement the DL VoIP model in simulations, the packets are generated at packet arrival times $iT+\tau$, where T is the VoIP frame interval of 20 ms, and i ={0, 1, 2, 3… }. As the range of τ is -80 ms ≤ τ ≤ 80 ms, it is more useful to implement the model with packet arrivals at times $iT+\tau'$ seconds, where $\tau' = \tau + 80$ ms and is always positive, 0 ms ≤ τ' ≤ 160 ms.

In order to implement this model in simulations, the VoIP packet process model has two parts:

a) Update of the Markov model regularly every iT seconds to determine active/inactive state, and determination of the packet arrival time, $iT+\tau'$, for the i^{th} packet

b) Generation of the i^{th} VoIP packet (Active or inactive) at time $iT+\tau'$. Begin measuring air interface delay for the i^{th} VoIP packet from this time.

The process is illustrated for several VoIP packets in an active state in Figure 3 below. For the i^{th} packet, at time iT the Markov model is updated and i^{th} packet arrival time ($iT+\tau'_i$) is determined (blue). The i^{th} packet is generated later at the packet arrival time of $iT+\tau'_i$ (green).

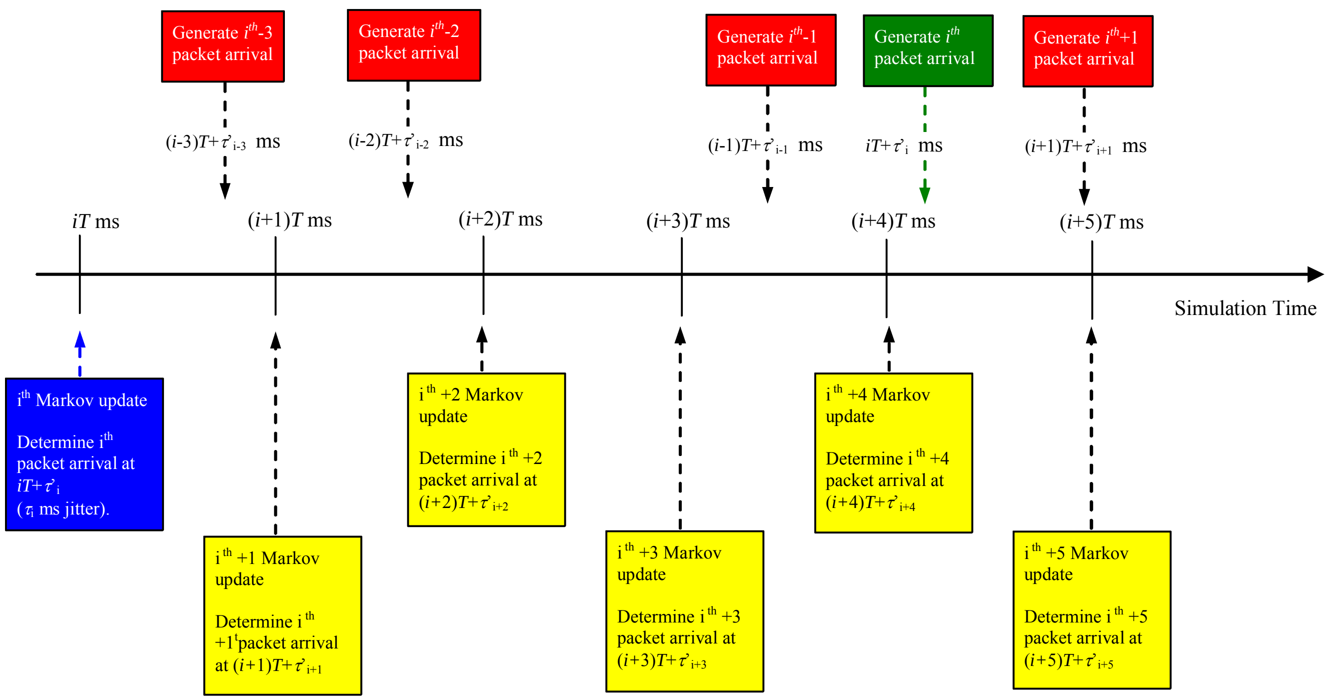

Figure 3. Illustrative timeline of Markov model updates and simulated VoIP packet arrival/generations, where $\tau' = \tau + 80 = \tau + 4T$ ms

5. References

[1] R. Srinivasan, J. Zhuang, L. Jalloul, R. Novak, J. Park, *Draft IEEE 802.16m Evaluation Methodology Document*, IEEE 802.16m-07/037r2, December 14, 2007.

[2] *Comments on Draft 802.16m Evaluation Methodology Document*, IEEE 802.16m-07/041r4, November 14, 2007.

[3] R. Srinivasan, J. Zhuang, L. Jalloul, R. Novak, J. Park, *Draft IEEE 802.16m Evaluation Methodology Document*, IEEE 802.16m-07/037r1, October 22, 2007.

[4] R. Novak, M.-H. Fong, K. Au, S. Vrzic, S. Yuan, *Downlink VoIP Packet Delay Jitter Model*, IEEE C802.16m-07/184, September 10, 2007.

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

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

[7] 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.

[8] 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.

[9] 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.

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

6. Proposed Text

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

1) Modifying text on page 106, lines 11-15:

"…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 iT+ τ, where τ is the network packet arrival delay jitter, and i is the encoder frame index. During the active state, packets of fixed sizes are generated at these time intervals, while the model is updated at regular frame intervals.

2) Modify text on page 108, lines 13-16:

"…fixed sizes will be generated at intervals of iT+τ seconds, where T is the VoIP frame interval of 20 ms, ~~and~~ τ is the DL network delay jitter, and i is the VoIP frame index. For the UL, τ is equal to 0. As the range of the delay jitter is limited to ~~120~~ 160 ms, the model may be implemented ~~as packet arrivals~~ by generating packets at times iT+τ' seconds, where $\tau' = \tau$ + ~~60~~ 80 ms and is always positive. The air interface delay is the time elapsed from the packet arrival time (iT+τ') to successful reception and decoding of the packet.

3) Adding row to Table 37 at line 19 of page 108:

Component	Distribution	Parameters	PDF
Active/Inactive state duration	Exponential	Mean = 1.25 second	$f_x = \lambda e^{-\lambda x}$, $x \geq 0$ $\lambda = 1/Mean$
Probability of transition	N/A	0.016	N/A
Packet arrival delay jitter (Downlink only)	Laplacian	β = 5.11ms	$f_x = \frac{1}{2\beta} e^{-\frac{\lvert\tau\rvert}{\beta}}$, -80 ms ≤ τ ≤ 80ms