Project	**IEEE 802.16 Broadband Wireless Access Working Group <http://ieee802.org/16>**
Title	**Mean VoIP Traffic Bit Rate with ON-SID Modeling for Simplified AMR and G729 Operations**
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Source(s)	Yih-Guang Jan, Yang-Han Lee, Ming-Hsueh Chuang, Hsien-Wei Tseng, Jheng-Yao Lin, and Chih-Wei Su Voice: +886-2-2625-2303 E-mail: yihjan@yahoo.com yhlee@ee.tku.edu.tw Institute for Information Industry 7F., No. 218, Sec. 2, Dunhua S. Rd., Taipei City, Taiwan. Department of Electrical Engineering, Tamkang University 151 Ying-chuan Road, Tamsui, Taipei County, Taiwan 25137, R. O. C. [co-authors added here]
Re:	IEEE 802.16m-07/080r3– Call for Comments on Draft 802.16m Evaluation Methodology Document
Abstract	This document contains proposed text for the draft evaluation methodology for IEEE 802.16m technical proposals.
Purpose	For discussion and approval by TGm
Notice	
Release	
Patent Policy	

Mean VoIP Traffic Bit Rate with ON-SID Modeling for Simplified AMR and G729 Operations

References

[1] A. Estepa, R. Estepa and J. Vozmediano,” A new Approach for VoIP Traffic Characterization”, IEEE Communications Letters, Vol. 8, No. 10, October 2004, pp. 644-646.
[2] P. Pragtong, T. Erke, and K. M. Ahmed,” Modeling of VoIP Conversational Speech in Noisy Environment,” TENCON 2005 2005 IEEE Region 10, Nov. 2005, pp. 1- 5.
[3] 3GPP TSG-RAN1#48 R1-070674, LTE Physical layer frame work for performance verification, Orange, China Mobile, KPN, NTT DoCcMo, Sprint, T-Mobile, Vodafone, Telecom Italia, February 2007.
[4] R. Estepa, A. Estepa and J. Vozmediano,” Accurate Prediction of VoIP Traffic Mean Bit Rate”, Electronics Letters, 18th August 2005, Vol. 41, No. 17.
[5] RFC3551: H. Schulzrinne,” RTP Profile for Audio and Video Conferences with Minimal Control”, Internet Engineering Task Force, 2003.
[6] Draft IEEE 802.16 Evaluation Methodology Document, C802.16m/080r2, June 18, 2007, IEEE 802.16 Broadband Wireless Access Working Group.

1. Introduction

Voice over IP (VoIP) is the transmitting of packetized voiced signal through the lnternet, and then these IP packets are reassembly at the receiver side. In order to improve the conversation quality in the silence conversation period a shorter frame named Silence Insertion Descriptor (SID) has been inserted in the silence period in the modern audio codec. This SID carries the information of the talker’s noisy background, allowing its reproduction in the receiver side. Although the size of the SID frame is small comparing with the active voice frames at the physical level, it is at the transport level that it needs to include the RTP/UDP/IP header that might generate a significant increment in the total traffic generated during the silence period. A general formula for evaluating the mean VoIP bit rate in one conversation will be derived by including the SID frames and these header overheads. The resulting VoIP bit rates will be explored for the simplified AMR and G.729 operations by implementing the derived formulas.

2. Mean VoIP Bit Rate with the ON-SID Model

The 2-state Markov Model for the VoIP traffic model can be depicted as in Figure 1.

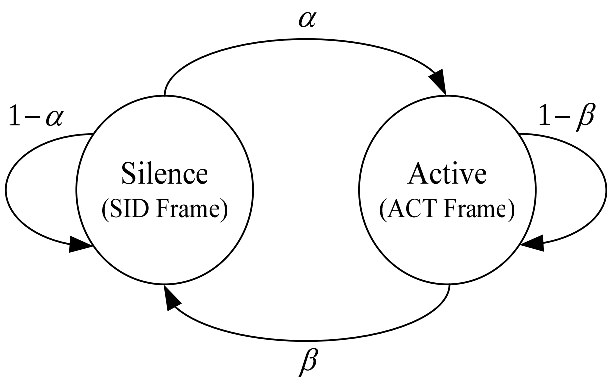

Figure 1 2-state Markov Model

The mean bit-rate of one conversion is the sum of the contributions of the traffic generated during voice activity (R_{ON}) and voice in silence periods (R_{SID}). Let ρ denote the voice activity rate then the bit rate can be expressed as:

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$$R = \rho * R_{ON} + (1 - \rho) * R_{SID} \qquad (1)$$

During voice activity periods a new voice packet loaded with N_{fpp} frames is generated every $N_{fppN} *T$ seconds, where T is a frame period.

Then the mean bit-rate for these periods is

$$R_{ON} = \frac{N_{fpp} * L_{ACT} + H}{N_{fpp} * T} \qquad (2)$$

where

L_{ACT} and H represent the size in bits of a voice frame and the RTP/UDP/IP header, respectively.

The mean bit-rate in the silence period can be expressed from the contributions of the header and the SID frames as:

$$R_{SID} = R_H + R_{SIDf} \qquad (3)$$

The contribution of the SID frames can be obtained from:

$$R_{SIDf} = \frac{L_{SID}}{T\,E[X]} \qquad (4)$$

where L_{SID} is the size of a SID and $T.E[X]$ is the expected inter-arrival time of the SID frame.

In considering the contribution of the packet header generated during the silence period, it follows from RFC3551 [5] that one packet header is sent for every non-consecutive SID frame and for consecutive SID frames one packet header is sent for every N_{fpp} frames. If *P1* denotes the probability of consecutive SID frames generated, then the header contribution can be evaluated as:

$$\begin{aligned} R_H &= P_1 \frac{H}{N_{fpp} TE[X]} + (1-P_1) \frac{H * N_{fpp}}{N_{fpp} TE[X]} \\ &= \frac{H}{TE[X]} \left(1 - \frac{P_1(N_{fpp}-1)}{Nfpp}\right) \end{aligned} \quad (5)$$

Then R_{SID} is the sum of R_{SIDf} and R_H. If it is ON-OFF model then R_{SID} would be zero. By summing R_{ON} and R_{SID} we have the overall mean bit-rate in one conversation as:

$$\begin{aligned} R &= \rho \left(\frac{L_{ACT}}{T} + \frac{H}{N_{fpp} T} \right) + \frac{(1-\rho)}{E[X\}T} * \\ &\left(L_{SID} + H \left(1 + \frac{P_1(1-N_{fpp})}{N_{fpp}}\right) \right) \end{aligned} \quad (6)$$

The parameters in the evaluation of R depend on the codec's characteristics and the number of frames per packet N_{fpp}.

3. Mean Bit-Rate Comparisons for Simplified AMR and G729 Operations

The detailed parameters characteristics of the VoIP traffic model for simplified AMR and G.729 operations are listed in Table 1 [6]. Based on these parameters, the resulting mean VoIP bit-rate for the simplified AMR and G.729 operations using IPv4 is shown in Figure 2 when it contains 1 to 10 frames in one conversation while by using IPv6 it has result as shown in Figure 3.

Table 1 Detailed Descriptions of the VoIP Traffic Model for Simplified AMR and G.729 [6]

Description	AMR without Header Compression IPv4/IPv6	AMR with Header Compression IPv4/IPv6	G.729 without Header Compression IPv4/IPv6	G.729 with Header Compression IPv4/IPv6
Voice Payload (20 ms aggregation interval)	7 bytes/inactive /IPv4/IP6 33 bytes/active /IPv4/IP6	7 bytes/inactive /IPv4/IP6 33 bytes/active /IPv4/IP6	0 bytes/inactive /IPv4/IP6 20 bytes/active /IPv4/IPv6	0 bytes/inactive /IPv4/IPv6 20 bytes/active /IPv4/IPv6
Protocol Headers (including UDO checksum)	40 bytes/IPv4 60 bytes/IPv6	3 bytes/IPv4 5 bytes/IPv6	40 bytes/IPv4 60 bytes/IPv6	3 bytes/IPv4 5 bytes/IPv6
802.16e Generic MAC Header	6 bytes/IPv4 6 bytes/IPv6	6 bytes/IPv4 6 bytes/IPv6	6 bytes/IPv4 6 bytes/IPv6	6 bytes/IPv4 6 bytes/IPv6
802.16e CRC for HARQ	2 bytes	2 bytes	2 bytes	2 bytes
Total VoIP Packet Size	55 bytes/Inactive /IPv4 75 bytes/Inactive /IPv6 81 bytes/active /IPv4 101 bytes/active /IPv6	18bytes/Inactive /IPv4 20 bytes/Inactive /IPv6 81 bytes/active /IPv4 101 bytes/active /IPv6	0 bytes/Inactive /IPv4 0 bytes/Inactive /IPv6 68 bytes/active /IPv4 88 bytes/active /IPv6	0 bytes/Inactive /IPv4 0 bytes/Inactive /IPv6 31 bytes/active /IPv4 33 bytes/active /IPv6

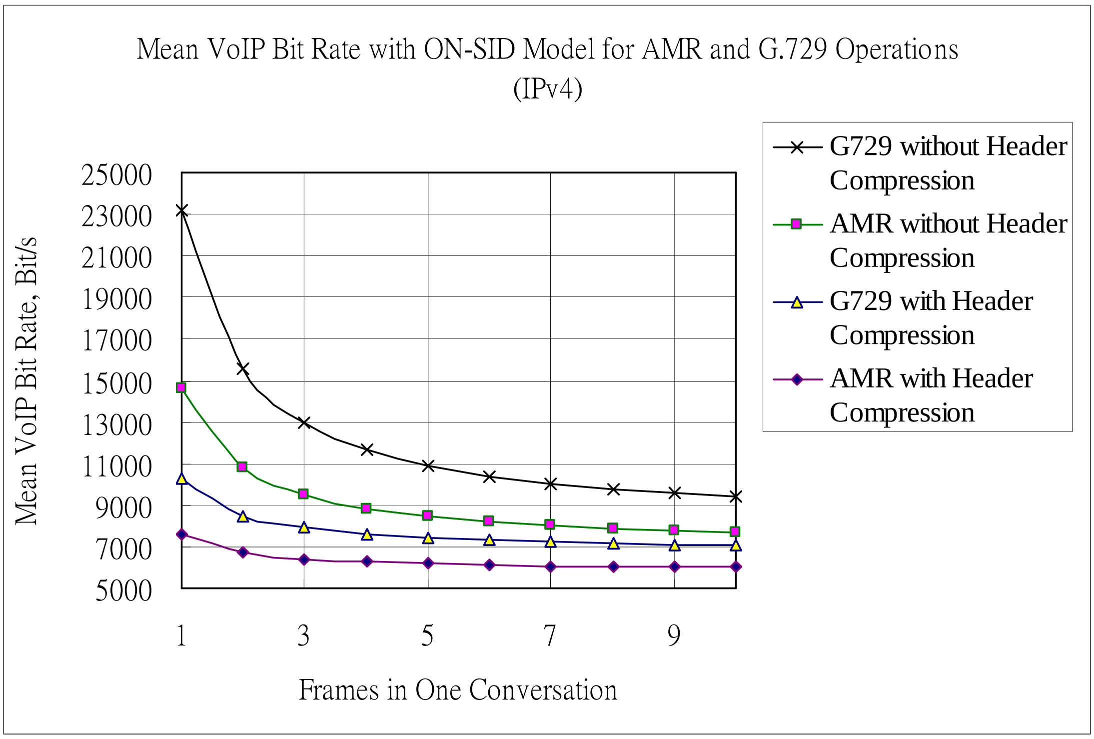

Figure 2 Mean Bit Rate with ON-SID Model for Simplified AMR and G729 Operation Using IPv4

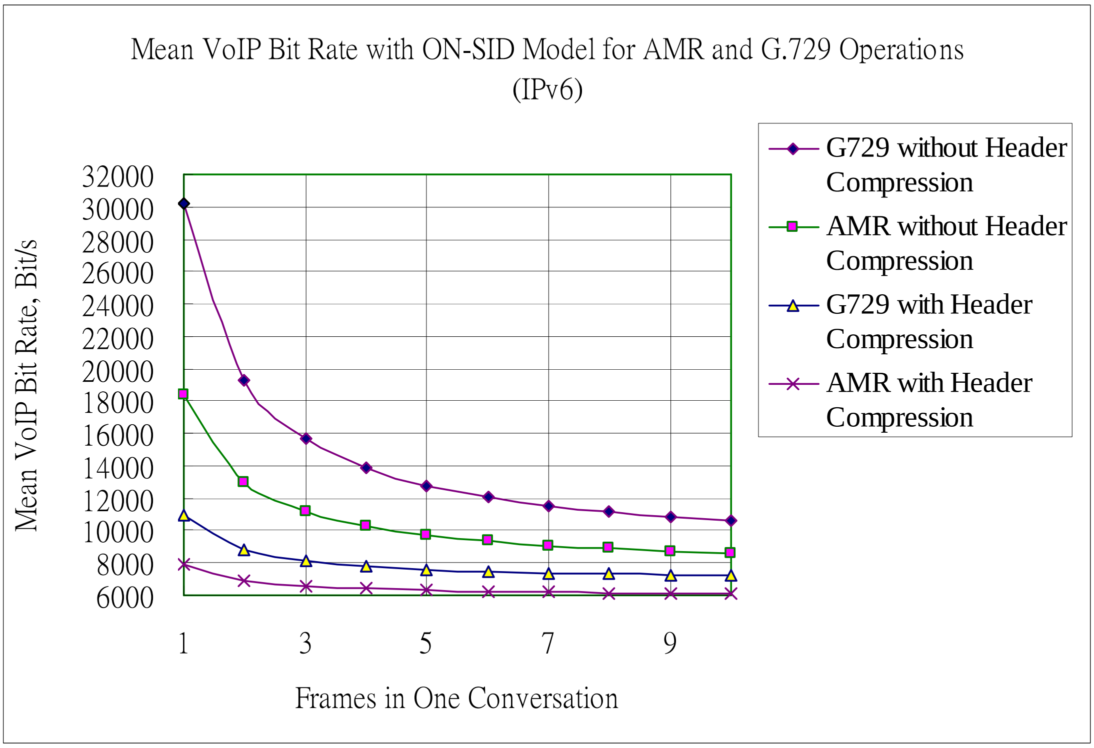

Figure 3 Mean Bit Rate with ON-SID Model for Simplified AMR and G729 Operation Using IPv6