Project	IEEE 802.16 Broadband Wireless Access Working Group <**http://ieee802.org/16**>	
	Harmonized proposal to replace traffic models in IEEE 802.16j-06/013	
Date Submitted	2006-09-27	
Source(s)	Wendy C Wong wendy.c.wong@intel.com Roshni Srinivasan Hannah Hyunjeong Lee Kerstin Johnsson Jerry Sydir Sassan Ahmadi Belal Hamzeh Shailender Timiri Intel Corporation 2200 Mission College Blvd., Santa Clara, CA 95054, USA. I-Kang Fu IKFu@itri.org.tw National Chiao Tung University / Industrial Technology Research Institute 1001 Ta Hsueh Road, Hsinchu , Taiwan 300, ROC. Peter Wang peter.wang@nokia.com Nokia US 6000 Connection Dr Irving, TX 75039, USA. David Chen david.t.chen@motorola.com Hua Xu Roger Peterson Motorola Inc 1441 W. Shure Drive, Arlington Heights, IL 60004 USA. Aik Chindapol aik.chindapol@siemens.com Teck Hu Siemens 755 College Road East, Princeton, NJ 08540	Mike Hart, Sunil Vadgama Mike.Hart@uk.fujitsu.com Fujitsu Laboratories of Europe Ltd. Hayes Park Central Hayes End, Middx., UB4 8FE, UK. Peng-Yong Kong, Haiguang Wang kongpy@i2r.a-star.edu.sg Institute for Infocomm Research 21 Heng Mui Keng Terrace Singapore 119613 Dharma Basgeet, Yong Sun Dharma.Basgeet@toshiba-trel.com Toshiba Research Europe Limited 32 Queen Square Bristol BS1 4 ND Jun Bae Ahn jbahn@st.co.kr SOLiD Technologies 10th Fl., IT Venture Tower East Wing, 78 Garak-Dong, Dongpa-Gu, Seoul, 138-803 Korea Hyunjeong Kang Hyunjeong.kang@samsung.com Jaeweon Cho Hyoungkyu Lim Samsung Electronics Co., Ltd 416 Maetan-dong, Yeongtong-gu, Suwon-si, Gyeonggi-do, 443-742, Korea Gamini Senarath gamini@nortel.com Wen Tong, Peiying Zhu, Hang Zhang, David Steer, Derek Yu, Mark Naden, and Dean Kitchener Nortel 3500 Carling Avenue Ottawa, On, K2H 8E9 Canada
Re:	Response to chair's call for comments on Multi-hop System Evaluation Methodology (IEEE 802.16j-06/013)	
Abstract	This is the harmonized contribution of the proposed C802.16j-06/093r3, C802.16j-06/094, and traffic models (Section 3 and Appendix C) proposed in IEEE 802.16j-06/013.	
Purpose	Improve the traffic models in IEEE 802.16j-06/013.	

Notice	This document has been prepared to assist IEEE 802.16. It is offered as a basis for discussion and is not binding on the contributing individual(s) or organization(s). The material in this document is subject to change in form and content after further study. The contributor(s) reserve(s) the right to add, amend or withdraw material contained herein.
Release	The contributor grants a free, irrevocable license to the IEEE to incorporate material contained in this contribution, and any modifications thereof, in the creation of an IEEE Standards publication; to copyright in the IEEE's name any IEEE Standards publication even though it may include portions of this contribution; and at the IEEE's sole discretion to permit others to reproduce in whole or in part the resulting IEEE Standards publication. The contributor also acknowledges and accepts that this contribution may be made public by IEEE 802.16.
Patent Policy and Procedures	The contributor is familiar with the IEEE 802.16 Patent Policy and Procedures <http://ieee802.org/16/ipr/patents/policy.html>, including the statement "IEEE standards may include the known use of patent(s), including patent applications, provided the IEEE receives assurance from the patent holder or applicant with respect to patents essential for compliance with both mandatory and optional portions of the standard." Early disclosure to the Working Group of patent information that might be relevant to the standard is essential to reduce the possibility for delays in the development process and increase the likelihood that the draft publication will be approved for publication. Please notify the Chair <mailto:chair@wirelessman.org> as early as possible, in written or electronic form, if patented technology (or technology under patent application) might be incorporated into a draft standard being developed within the IEEE 802.16 Working Group. The Chair will disclose this notification via the IEEE 802.16 web site <http://ieee802.org/16/ipr/patents/notices>.

1 Introduction

This is the harmonized contribution of the proposed C802.16j-06/093r3, C802.16j-06/094, and traffic models proposed in IEEE 802.16j-06/013. We propose to replace traffic models description in Section 3 and Appendix C of IEEE 802.16j-06/013 with the harmonized version presented in Section 2 of this document. This contribution supersedes the previous comments on traffic models proposed in IEEE 802.16j-06/013 which are included in C802.16j-06/110r1.

We used the following color codes to identify the changes. However, these color codes apply to only those paragraphs discussed during the comment resolution meeting on 9/26/2006

- Blue (ABC) : Approved insertion from the comment resolution meeting on 9/26/2006
- Black Strikethrough ~~(ABC)~~: Approved deletion from the comment resolution meeting on 9/26/2006
- Blue Strikethrough ~~(ABC)~~ : Approved insertion from the comment resolution meeting on 9/26/2006but we would like to delete.
- Red Strikethrough ~~(ABC)~~ : Existed in IEEE 802.16j-06/013 and approved sentence from the comment resolution meeting on 9/26/2006 but we would like to delete
- Green (ABC) : Editor's note

[Editor's note: Table and Figure numbers are subject to change according to the numbering used in IEEE 802.16j-06/013. Editor's note shall be removed when replacing Section 3 and Appendix C with the following Section 2 of this document.]

2 Traffic models

This section describes the traffic models in detail. ~~Section 3.1 addresses the DL and Section 3.2 the UL~~ A major objective of multihop simulations is to provide the operator a view of the maximum number of ~~how many~~ active users that can be supported for a given service under a specified multihop configuration at a given coverage level. The traffic generated by a service should be accurately modeled in order to find out the performance of a system. This may be a time consuming exercise. Traffic modeling can be simplified, as explained below, by not modeling the user arrival process and/or assuming full queue traffic which is considered as the baseline. These two assumptions are further discussed proceeding paragraphs. Modeling non-full-queue traffic is also discussed in the next subsections. ~~explained below~~

Modeling of user arrival process: Typically, all the users are not active at a given time and even the active users ~~they~~ might not register for the same service. In order to avoid different user registration and demand models, the objective of the proposed simulation is restricted ~~made limited~~ to evaluate the performance with the users who are maintaining a session with transmission activity. These can be used to determine the number of such registered users that can be supported. This document does not address the arrival process of such registered users, i.e. it does not address the statistics of subscribers that register and become active.

Full Queue model: In the full queue user traffic model, all the users in the system always have data to send or receive. In other words, there is always a constant amount of data that needs to be transferred at a given source, in contrast to bursts of data that follow an arrival process. This model allows the assessment of the spectral efficiency of the system independent of actual user traffic distribution type.

~~At the relay station, however, the traffic availability depends on the forwarded traffic from either base station, user or by another relay even in the full queue model and full queue model may or may not be applicable.~~

[Editor's note: Insert the following text]**

In the following sections, we will concentrate on traffic generation only for the non-full queue case. In addition, the interaction of the generated traffic with the higher layer protocol stack such as TCP is not included here. However, we will provide references to document which provide the detailed TCP transport layer implementation and its interaction with the various traffic models.

****
The traffic models proposed in this document ~~section~~ apply only to the MMR-BS and SS.

2.1 Traffic Models (Non-Full Queue) to be used for IEEE802.16j Services

The required traffic models and their corresponding sections where they are defined are listed in Table 1.

[Editor's note: Order of the rows is changed and new model, i.e., gaming is added to Table 1]

Table 1: Services to be considered

#	Application	Traffic Category	Definition ~~Priority~~

1	Full buffer		~~Provided above and in Appendix C.2.1.~~ Section 3 [Editor's note: Section 2 in this document]
2	HTTP (UL and DL)	Interactive	~~Provided in Appendix C.1.2 and C.2.3~~ Section 3.1.1 [Editor's note: Section 2.1.1 in this document]
3	FTP (UL and DL)	Best-effort / Non real-time	~~Provided in Appendix C.1.1 and C.2.2.~~ Section 3.1.2 [Editor's note: Section 2.1.2 in this document]
4	Near Real Time (NRT) Video Streaming (UL and DL)	Streaming	~~Provided in Appendix C.3.2~~ Section 3.1.3 [Editor's note: Section 2.1.3 in this document]
5	VoIP	Real-time	~~Provided in Appendix C.3.1~~ Section 3.1.4 [Editor's comment: Section 2.1.4 in this document]
6	Gaming (UL and DL)	Real-time	Section 3.1.5 [Editor's comment: Section 2.1.5 in this document]
7	Live Video	Interactive Real-time	TBD

[Editor's note: Insert the following text and Figure 1. The first paragraph summarizes the Appendix C.3.3]
****

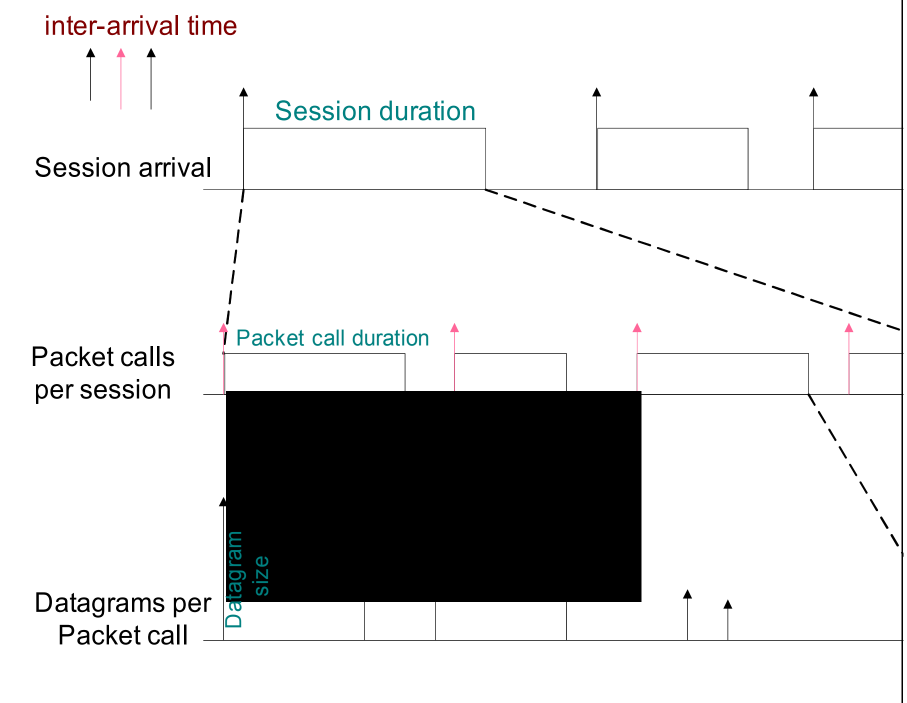
For a simulation with HTTP, FTP and NRT video streaming traffic models, if simulation is for DL (or UL) traffic only, UL (or DL) traffic modeling (e.g. HTTP/FTP requests) can be neglected for the simplicity as the bandwidth requirements for these messages are small compared to the data traffic.

The FTP and HTTP traffic models listed in Table 1 can be generated using the bursty traffic generation model described in Figure 1. For each traffic source, the following characteristics are modeled:

1. Session arrival in terms of session inter-arrival time and session duration.
2. Packet call arrival in terms of packet call inter-arrival time and packet call duration within a session. Within a packet call, there are periods of active traffic generation and periods of no activity.
3. Finally, datagram inter-arrival times and datagram size within a packet call.

We consider that a single session stays from the beginning of the simulation till the end of the simulation, i.e., the whole simulation time. Therefore, packet call and datagram inter-arrival times, packet call duration and datagram size distributions for these bursty traffic models will be described in the next sections.

Figure 1. Bursty traffic generation model



****

The following subsections (3.1.1 to 3.1.5) describe these traffic models in detail. [Editor's note: Sections 2.1.1 - 2.1.5 in this document]

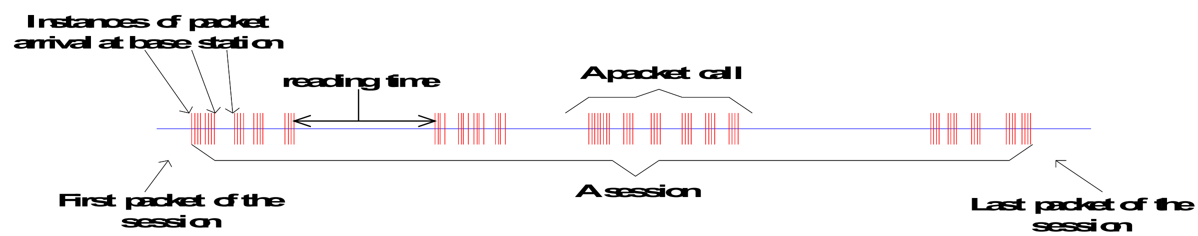
[Editor's note: Merge the entire sections of C.1.2 and C.2.3 and replace these sections with the section 2.1.1 in this document. All the TCP related details and Figure 14 in C.1.2 have been removed and Tables 8 and 10 are merged.]

2.1.1 HTTP model (UL and DL) [1][2][7]

2.1.1.1 HTTP traffic model characteristics

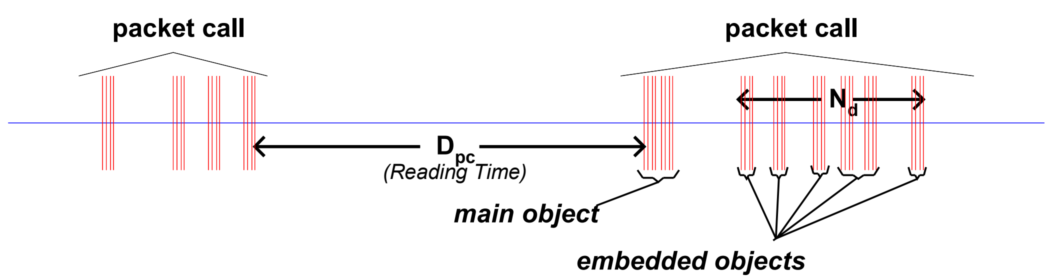
Figure 2 shows a typical web browsing session. Each session is divided into ON/OFF periods representing web-page downloads and intermediate reading times. Each web-page download is referred to as packet calls in Figure 2. During an ON period (packet call), users are requesting information. During an OFF period, user is reading/digesting the web-page.

Figure 2. Packet trace of a typical web browsing session



The activity within each packet call can be found in Figure 3. Note the similarity of the distribution for the packet calls within a session in Figure 2 and the datagram arrivals within a packet call in Figure 3. This can possibly be a result of self-similarity in web-browsing traffic.

Figure 3. Contents of a packet call



There are ON and OFF periods within a packet call. During an ON period, objects are being retrieved. Parsing time and protocol overhead are represented by the OFF periods within a packet call. During a packet call, the initial HTML page (referred to as the main object) is first downloaded. However, within the initial HTML page, there can be additional references to embedded object files such as graphics and buttons. After parsing the information on the embedded objects, the embedded objects will be loaded next as indicated in Figure 3.

2.1.1.2 HTTP traffic model parameters

The parameters for web browsing traffic are:

No of pages per session;
S_M: size of the main object in a packet call;
S_E: size of an embedded object in a packet call;
N_d: number of embedded objects in a packet call;
D_{pc}: reading time;
T_p: parsing time for main page

Table 2 HTTP Traffic Model Parameters

Component	Distribution	DL Parameters	UL Parameters	
Main object size (S_M)	Truncated Lognormal	Mean = 10710 bytes, Std. Dev = 25032 bytes, Minimum = 100 bytes; Maximum = 2Mbytes, 1.37, 8.35	Mean = 9055 byes, Std. dev. = 13265 bytes, Minimum = 100 bytes, Maximum = 100Kbytes 1.37, 8.35	$f_x \quad \frac{1}{\sqrt{2} \quad x} \exp \quad \frac{\ln x \qquad ^2}{2 \quad ^2}$, $x \quad 0$
Embedded object size (S_E)	Truncated Lognormal	Mean = 7758bytes, Std. dev. = 126168bytes, Minimum = 50bytes, Maximum = 2Mbytes 2.36, 6.17	Mean = 5958bytes, Std. dev. = 11376bytes, Minimum = 50bytes, Maximum=100kbytes 1.69, 7.53	$f_x \quad \frac{1}{\sqrt{2} \quad x} \exp \quad \frac{\ln x \qquad ^2}{2 \quad ^2}$, $x \quad 0$
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 5.64, Maximum = 53 1.1, k 2, m 55	Mean = 4.229, Maximum = 53 1.1, k 2, m 55	$f_x \quad \frac{k}{x \quad ^{1}}, k \quad x \quad m$ $f_x \quad \frac{k}{m}, x \quad m$ Subtract k from generated random value to obtain N_d.
Reading Time (D_{pc})	Exponential	Mean = 30seconds	Mean = 30seconds	$f_x \quad e^{\quad x}, x \quad 0$ 0.033
Parsing time (T_p)	Exponential	Mean = 0.13second	Mean = 0.13second	$f_x \quad e^{\quad x}, x \quad 0$ 7.69

Note: when generating a random sample from a truncated distribution, discard the random sample when it is outside the valid interval and regenerate another random sample.

2.1.1.3 HTTP and TCP interactions for DL HTTP traffic

Two versions of the HTTP protocol, HTTP/1.0 and HTTP/1.1, are widely used by servers and browsers. Users shall specify 30% HTTP/1.0 and 70% HTTP/1.1 for HTTP traffic.

For people who have to model the actual interaction between HTTP traffic and the underling TCP connection, refer to 4.1.3.2, 4.2.4.3 of [1] for details.

2.1.1.4 HTTP and TCP interactions for UL HTTP traffic

HTTP/1.1 is used for UL HTTP traffic. For details regarding the modeling of the interaction between HTTP traffic and the underling TCP connection, refer to 4.2.4.1, 4.2.4.2 of [1].

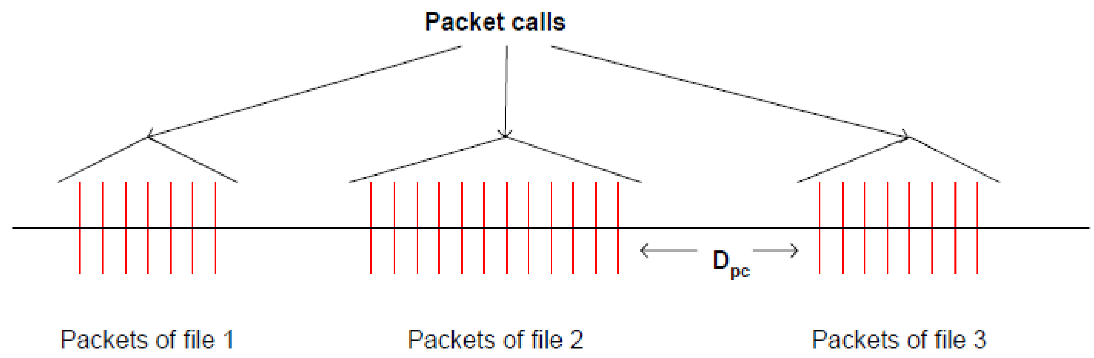
[Editor's note: Merge the entire sections of C.1.1 and C.2.2 and replace these sections with Section 2.1.2 in this document]

2.1.2 FTP model (UL and DL) [1][2]

2.1.2.1 DL FTP traffic model characteristics

For DL FTP, activities within a FTP session can be found in Figure 4. A typical FTP session consists of a sequence of file transfers separated by reading time. Each file transfer can be treated as a packet call. Reading time can be treated as the OFF period within a session. Within each packet call, only the file size is randomly generated.

Figure 4. Packet trace in a typical DL FTP session



2.1.2.2 DL FTP traffic model parameters

Hence, there are two main parameters for a DL FTP session:

1. S: size of file to be transferred;
2. D_{pc}: reading time. This is the time interval between end of download of the previous file and the user request for the next file.

The parameters distribution and values can be found in Table 3.

Table 3 DL FTP traffic model parameters

Component	Distribution	Parameters	PDF
File size (S)	Truncated Lognormal	Mean = 2Mbytes Std. Dev. = 0.722 Mbytes Maximum = 5 Mbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 0.35, \mu = 14.45$
Reading time (D_{pc})	Exponential	Mean = 180 sec (TBD).	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.006$

2.1.2.3 UL FTP traffic model characteristics

FTP traffic in the UL direction is generated mainly from file upload and email attachment upload. Each FTP upload user stays in the system until it finishes the transmission of its file. The FTP upload user leaves the system immediately after it finishes the transmission of its file.

Hence, for UL FTP traffic, each FTP session consists of 1 packet call. Within the packet call, only the file size is randomly generated.

2.1.2.4 UL FTP traffic model parameters

The only traffic model parameter is the upload file size and can be found in Table 4.
For UL FTP traffic, users shall arrive according to a Poisson process with arrival rate .

Table 4 UL FTP traffic model parameter

Arrival of new users	Poisson with parameter
Upload file size	Truncated lognormal; lognormal pdf: $f_x \quad \frac{1}{\sqrt{2}\quad x} \exp \frac{(\ln x \quad)^2}{2 \quad ^2}, x \quad 0$ 2.0899, 0.9385 Min = 0.5 kbytes, Max = 500 kbytes If the value generated according to the lognormal pdf is larger than Max or smaller than Min, discard it and regenerate a new value. The resulting truncated lognormal distribution has a mean = 19.5 kbytes and standard deviation = 46.7 kbytes

2.1.2.5 FTP and TCP interactions

To model the FTP and TCP interactions, please refer to 4.1.4.2 of [1] for details.

[Editor's note: Replace Section C.3.2 with the section 2.1.3 in this document.]

2.1.3 Near real time video streaming (NRT video streaming) (UL and DL) [1][2]

A video streaming session is defined as the entire video streaming call time. It is equal to the simulation time for this model. Hence, a video streaming session occurs during the whole simulation period. No session inter-arrival time is needed. It is originally modeled for DL direction. However, the same model is proposed to be used for UL direction.

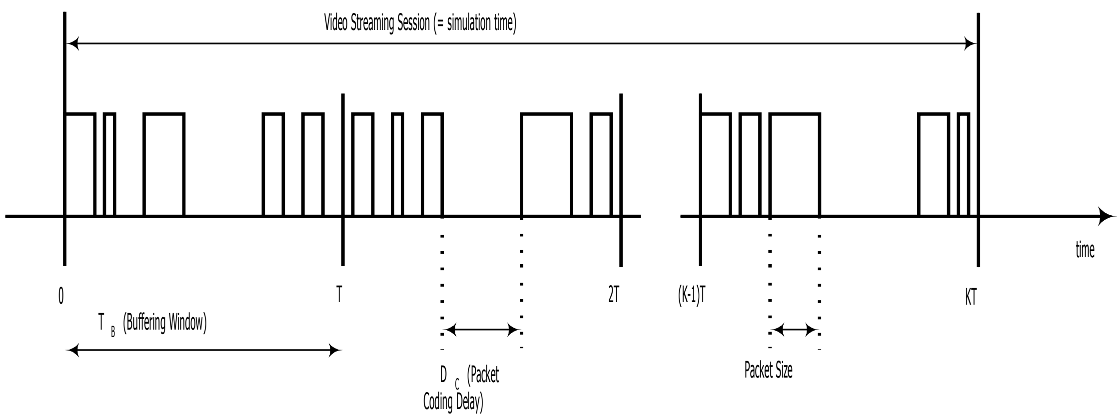
2.1.3.1 NRT video streaming traffic model characteristics

Figure 5 describes a steady state of video streaming traffic from the network as observed by the base station. Call setup latency and overhead is not considered in this model.

Each frame of video data arrives at a regular interval T. Each frame can be treated as a packet call and there will be zero OFF duration within a session. Within each frame (packet call), packets (or datagrams) arrive randomly and the packet sizes are random as well.

To counter the jittering effect caused by the random packet arrival rate within a frame at the MS, the MS uses a de-jitter buffer window to guarantee a continuous display of video streaming data. The de-jitter buffer window for video streaming service is 5 seconds. At the beginning of simulation, the MS de-jitter buffer shall be full with video data. During simulation, data is leaked out of this buffer at the source video data rate and filled as DL traffic reaches the MS from the BS. As a performance criterion, the simulation shall record the length of time, if any, during which the dejitter buffer runs dry.

Figure 5. Video streaming traffic model



2.1.3.2 NRT video streaming traffic model parameters

The packet sizes and packet inter-arrival rate can be found in Table 5 when using a source rate of 64 kbps.

Table 5 Near Real-Time Video Traffic Model Parameters

Information types	Inter-arrival time between the beginning of each frame	Number of packets (slices) in a frame	Packet (slice) size	Inter-arrival time between packets (slices) in a frame
Distribution	Deterministic (Based on 10fps)	Deterministic	Truncated Pareto (Mean= 50bytes, Max= 125bytes)	Truncated Pareto (Mean= 6ms, Max= 12.5ms)
Distribution parameters	100ms	8	K=20bytes = 1.2	K=2.5ms = 1.2

[Editor's note: Replace Section C.3.1 with the section 2.1.4 in this document. C.3.1 has been substantially detailed]

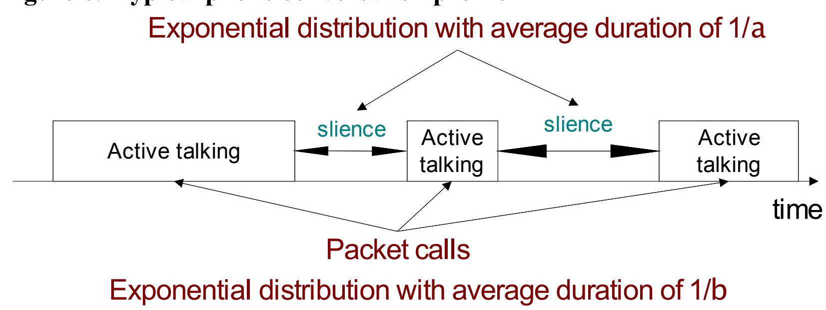
2.1.4 VoIP model [1][3][4][5]

VoIP refers to real-time delivery of packet voice across networks using the Internet protocols. A VoIP session is defined as the entire user call time and VoIP session occurs during the whole simulation period.

2.1.4.1 VoIP traffic model characteristics

A typical phone conversation is marked by periods of active talking interleaved by silence/listening period as shown in Figure 6.

Figure 6. Typical phone conversation profile



A two state Markov process (active-inactive) is used to model a VoIP source in Figure 7. The alternating periods of activity and silence are exponentially distributed with average durations of 1/ and 1/ respectively. Hence, the fraction of time the voice source is active is /(+). For a voice activity factor of 40%, 1/ = 1s and 1/ = 1.5s. Each active state period can be treated as a packet call and inactive period as the OFF period within a session.

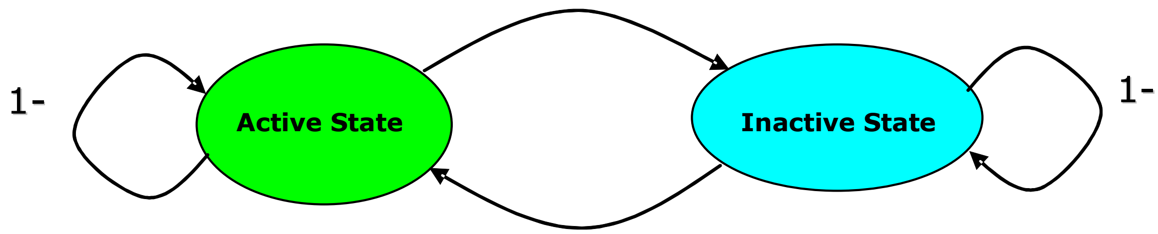
During the active state, packets of fixed sizes are generated at a regular interval. During the inactive state, we have chosen to generate comfort noise with smaller packet sizes at a regular interval instead of no packet transmission. The size of packet and the rate at which the packets are sent depends on the corresponding voice codecs and compression schemes. Table 6 provides information on some common vocoders.

Table 6 Information on various vocoders

Vocoder	**EVRC**	**AMR**	**G.711**	**G.723.1**		**G729A**
Source Bit rate [Kb/s]	0.8/2/4/8.55	4.75-12.2	64	5.3	6.3	8
Frame duration [ms]	20	20	10	30	30	10
Information bits per frame	16/40/80/171	95-244	640	159	189	80

Among the various vocoders in Table 6, a simplified AMR (adaptive multi-rate) audio data compression can be used to simplify the VoIP modeling process. AMR is optimized for speech coding and was adopted as the standard speech codec by 3GPP and widely used in GSM. The original AMR uses link adaptation to select from one of eight different bit rates based on link conditions. If the radio condition is bad, source coding is reduced (less bits to represent speech) and channel coding (stronger FEC) is increased. This improves the quality and robustness of the network condition while sacrificing some voice clarity. In our simplified version, we have chosen to disable the link adaptation and use the full rate of 12.2kbps in the active state. This will give us the worst case scenario.

Figure 7. Markov chain model of a VoIP source



Without header compression, AMR payload of 33 bytes are generated in the active state for every 20ms and AMR payload of 7 bytes are generated in the inactive state for every 160ms. Table 8 shows the VoIP packet size calculation for simplified AMR with or without header compression when using IPv4 or IPv6.

Table 8 VoIP packet size calculation for simplified AMR and G. 729

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	7bytes (inactive) 33 bytes (active)	7bytes (inactive) 33 bytes (active)	0 bytes (inactive) 20 bytes (active)	0 bytes (inactive) 20 bytes (active)
Protocol Headers	40 bytes / 60 bytes	2 bytes/ 4 bytes	40 bytes / 60 bytes	2 bytes/ 4 bytes
RTP	12 bytes		12 bytes	
UDP	8 bytes		8 bytes	
IPv4 / IPv6	20 bytes / 40 bytes		20 bytes / 40 bytes	
802.16 Generic MAC Header	6 bytes	6 bytes	6 bytes	6 bytes
CRC	4 bytes	4 bytes	4 bytes	4 bytes
Total VoIP packet size	57 bytes/ 77 bytes (inactive) 87 bytes / 103 bytes (active)	19 bytes/ 21 bytes (inactive) 45 bytes/ 47 bytes (active)	0 bytes (inactive) 70 bytes / 90 bytes (active)	0 bytes (inactive) 32 bytes/ 34 bytes (active)

2.1.4.2 VoIP traffic model parameters

During each call (each session), a VoIP user will be in the Active or Inactive state. The duration of each state is exponentially distributed. Within the Active/Inactive state, packets of fixed sizes will be generated at a fix interval. Hence, both the datagram size and datagram arrival intervals are fixed within a packet call. Parameters associated with the VoIP traffic model can be found in Table 1.

Table 1. VoIP traffic model parameters specification

Component	Distribution	Parameters	PDF
Active state duration	Exponential	Mean = 1 second	$f_x \quad e^{\quad x}, x \quad 0$ $1/Mean$
Inactive state duration	Exponential	Mean = 1.5 second.	$f_x \quad e^{\quad x}, x \quad 0$ $1/Mean$
Probability of transition from active to inactive state	N/A	(=0.6)	N/A
Probability of transition from inactive to active state	N/A		N/A

[Editor's note: Insert the following Section 2.1.5 to add the gaming model.]

2.1.5 Gaming model (UL and DL) [1][6]

Gaming traffic is generated by users engaged in interactive gaming of multiple users in different locations via the internet. A gaming session is defined as the time duration that a user plays a game and a gaming session occurs during the whole simulation period.

2.1.5.1 Gaming traffic model characteristics

The packet arrival time and the frame boundary are random and shall be simulated. Gaming packets are relatively small in size. Due to the interactive nature of gaming, packet delay must be short. Any packets that are generated and not transmitted at the PHY layer within 160ms shall be dropped.

2.1.5.2 Gaming traffic model parameters

Gaming traffic model parameters for DL and UL can be found in Table 9[6]. Largest Extreme Value distribution is used for random packet size generation. Since packet size has to be an integer, the largest integer less than or equal to X is used as the actual packet size.

Table 9 Gaming traffic model parameters

Component	Distribution		Parameters		PDF
	DL	UL	DL	UL	
Initial packet arrival	Uniform	Uniform	a=0, b=40ms	a=0, b=40ms	$f(x) \quad \frac{1}{b \quad a}, a \quad x \quad b$

Packet inter-arrival time	Extreme	Extreme	a=48ms, b=4.5ms	a=40ms, b=6ms	$f(x) = \frac{1}{b} e^{-\frac{x-a}{b}} e^{-e^{-\frac{x-a}{b}}}, b > 0$ $X = a - b\ln(-\ln Y), Y \sim U(0,1)$
Packet size	Extreme	Extreme	a=330bytes, b=82bytes	a=45bytes, b=5.7	$f(x) = \frac{1}{b} e^{-\frac{x-a}{b}} e^{-e^{-\frac{x-a}{b}}}, b > 0$ $X = a - b\ln(-\ln Y) + 2, Y \sim U(0,1)$ Addition of 2 in the equation is due to 2 bytes of UDP header size after header compression.

[Editor's note: Insert the following Section 2.2 to traffic mix model.]

2.2 Traffic mix proposal

To test various aspect of the system, we propose the following traffic mixes:
1. Five cases of HTTP, FTP, NRT Video Streaming, Gaming, or Voice only.
2. Three cases of mixed traffic from Mix -1 to Mix -3 referenced in Table 10. The percentage of the traffic mix in these 3 cases is expressed in terms of data capacity (i.e., bps) of a given targeted cell.

Table 2. Proposed traffic mixes

	VoIP	**FTP**	**HTTP**	**NRT video**	**Gaming**
Voice Only	100% #users = Nv	0%	0%	0%	0%
FTP only	0%	100%	0%	0%	0%
HTTP only	0%	0%	100%	0%	0%
NRT Video only	0%	0%	0%	100%	0%
Gaming only	0%	0%	0%	0%	100%
Traffic Mix 1 (TBD)	0.5 Nv	Remaining Capacity for Data Users 100% 0% 0% 0%			
Traffic Mix 2 (TBD)	0.5 Nv	Remaining Capacity for Data Users 30% 30% 30% 10%			
Traffic Mix 3 (TBD)	0.75 Nv	Remaining Capacity for Data Users 30% 30% 30% 10%			

Nv is the system voice capacity that satisfy outage criteria at system and user level.

3 References

[1] 3GPP2/TSG-C.R1002, "1xEV-DV Evaluation Methodology (V14)", June 2003.
[2]. IEEE 802.16j-06/013, "Multi-hop System Methodology (Channel Model and Performance Metric), 2006-05-19.
[3]. Chen-Nee Chuah, "A Scalable Framework for IP-Network Resource Provisioning Through Aggregation and Hierarchical Control", PhD dissertation, UC Berkeley, 2001.
[4]. IEEE P 802.20™ PD-09 Version 1.0, "*802.20 Evaluation Criteria – Ver. 1.0*," September 23, 2005.

[5]. IEEE 802.20 Working Group on Mobile Broadband Wireless Access, "*IEEE 802.20 Evaluation Criteria (EC) – Simulation of Common Traffic Types*," May-5-2005.
[6]. Hua Xu, Pranav Joshi, Eren Gonen, Y.C. Chen, Xiao Xu, "First Person Shooter Gaming Traffic Model for 802.16jMMR", IEEE 802.16j-06/094.
[7]. Bong Ho Kim, "WiMAX Forum-AWG: Real-World Traffic Benchmark", WiMAX Forum.