Project	IEEE 802.16 Broadband Wireless Access Working Group <**http://ieee802.org/16**>	
	Comments and proposal to replace traffic models in IEEE 802.16j-06/013	
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Re:	Response to chair's call for comments on Multi-hop System Evaluation Methodology (IEEE 802.16j-06/013)	
Abstract	We propose to replace the traffic model (section 3) of IEEE 802.16j-06/013 with our proposed traffic model starting in section 2 of this contribution.	
Purpose	Improve the traffic models in IEEE 802.16j-06/013.	
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1 Introduction

There is a lot of good information in the traffic model part of the Multi-hop System Evaluation Methodology (Channel Model and Performance Metric) document (IEEE 802.16j-06/013) [2]. However, we have found the following problems:

Some models are too complicated while others like VoIP are incomplete;
Gaming model is missing;
Section titles are confusing. Some sections are for UL but there are no specific sections for DL.

Hence, we propose to replace the traffic models in Section 3 of IEEE 802.16j-06/013 [2] with this contribution starting from section 2. We have added the missing models and re-organize the various sections in the traffic model of [2] to make it clear and easy to use.

2 Traffic models

For simulation aiming to test the PHY layer, use the full buffer model:

Without a traffic source:packets of various sizes can be injected to the frame directly
With a traffic source: FTP traffic with high enough arrival rate that can fill up all transmit buffer space.

In the following sections, we will concentrate on traffic generation for MAC and higher layer simulation. In addition, we assume that simulation software like OPNET is available which provides parameters for traffic models and TCP transport layer implementation. Users can then set the parameters for traffic models according to parameters specified here. For people with no such software access, we will provide references to document which provide the detailed TCP transport layer implementation and its interaction with the various traffic models.

The following topics will be discussed:

- Traffic model description using the generic packet data model;
- Traffic models:
  - HTTP
  - FTP
  - Near Real Time Video Streaming
  - VoIP
  - Gaming
- Traffic mixes proposal.

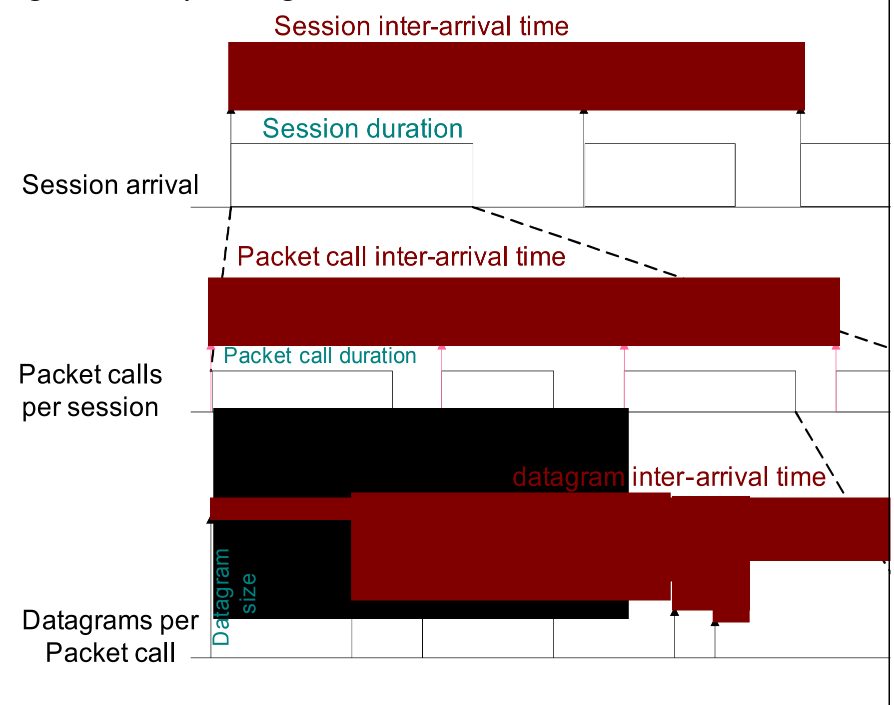
2.1 Traffic Generation

All traffic models 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. This describes the traffic arrival process in terms of when a user will arrive and how long the user will stay.
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.

The session, packet call and datagram inter-arrival times, session/packet call duration and datagram size distributions will be described in detail for each traffic model described in the coming sections.

Figure 1. Bursty traffic generation model



2.2 Traffic models

The following traffic models will be proposed:
HTTP;
FTP;
Near real time video streaming (NRTVS);
VoIP;
Gaming.

HTTP and FTP are referred to as data traffic while VoIP, gaming are real time traffic with strict delay and delay variation requirements. The traffic models apply to both DL and UL unless otherwise specified.

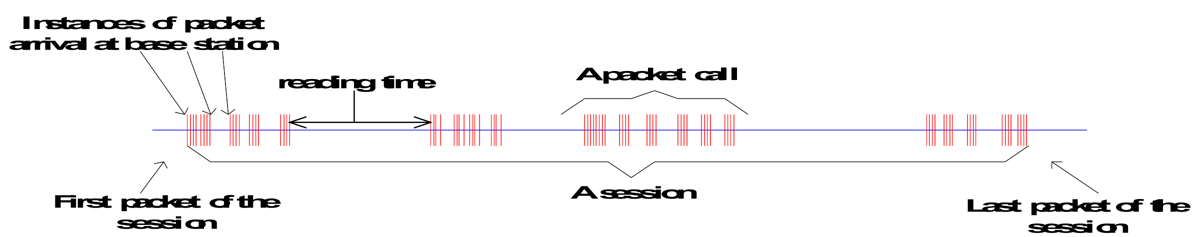
2.2.1 HTTP model

For HTTP traffic, the arrival of a web browsing user is modeled as a Poisson arrival process with arrival rate (TBD). This implies the session inter-arrival time is exponentially distributed. The session duration can be modeled by a TBD distribution.

2.2.1.1 HTTP traffic model characteristics

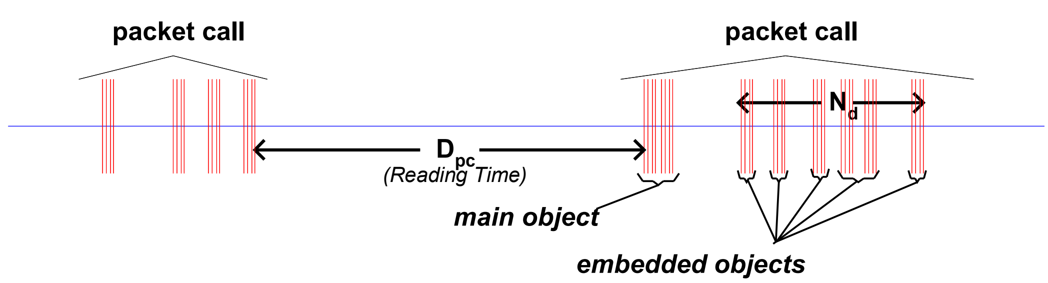
Figure 2 shows a typical web browsing session. Each session is divided into ON/OFF periods representing web-page downloads and intermediate reading times. Each web-page download is referred to as packet calls in Figure 2. During an ON period (packet call), users are requesting information. During an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

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.

2.2.1.2 HTTP traffic model parameters

The parameters for web browsing traffic are:

- 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 1 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 = 2 Mbytes	Mean = 9055 bytes Std. dev. = 13265 bytes Minimum = 100 bytes Maximum = 100 Kbytes	$f_x \quad \frac{1}{\sqrt{2}\quad x} \exp \frac{\ln x \quad ^2}{2 \quad ^2}, x \quad 0$ 1.37, 8.35
Embedded object size (S_E)	Truncated Lognormal	Mean = 7758 bytes Std. dev. = 126168 bytes Minimum = 50 bytes Maximum = 2 Mbytes 2.36, 6.17	Mean = 5958 bytes Std. dev. = 11376 bytes Minimum = 50 bytes Maximum = 100 Kbytes 1.69, 7.53	$f_x \quad \frac{k}{x^{\quad 1}}, k \quad x \quad m$ $f_x \quad \frac{k}{m}, x \quad m$
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 5.64 Max. = 53	Mean = 4.229 Max. = 53	1.1, k 2, m 55 Note: Subtract k from the generated random value to obtain N_d
Reading time (D_{pc})	Exponential	Mean = 30 sec	Mean = 30 sec	$f_x \quad e^{\quad x}, x \quad 0$ 0.033
Initial reading time (D_{ipc})	Uniform	N/A	Range [0,10]sec	$f_x \quad \frac{1}{b \quad a}, a \quad x \quad b$ $a \quad 0, b \quad 10$

Parsing time (T_p)	Exponential	Mean = 0.13 sec	Mean = 0.13 sec	$f_X \quad e^{\quad x}, x \quad 0$ 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.

2.2.1.3 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. For people with simulation software like OPNET, users shall specify 50% HTTP/1.0 and 50% 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.

2.2.1.4 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].

2.2.2 FTP model

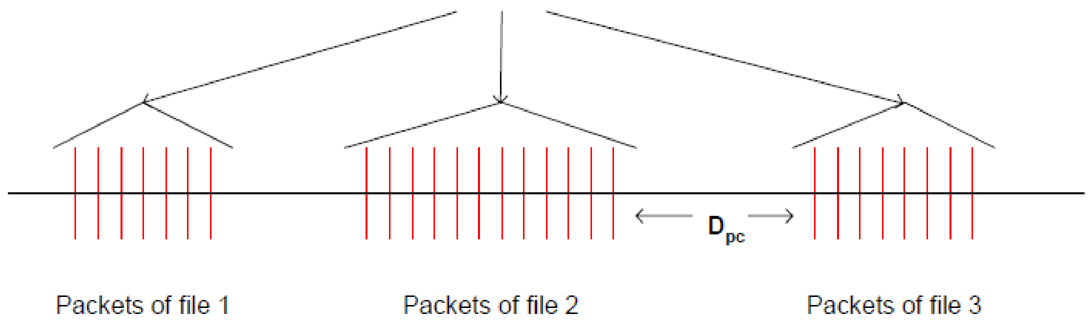
For DL FTP traffic, the arrival of a FTP user is modeled as a Poisson arrival process with arrival rate (TBD). The session duration can be modeled by a TBD distribution.

For the UL FTP traffic, there shall be 5 FTP users waiting to transmit at the beginning of simulation. Simulation software like OPNET shall be directed to perform call setup automatically. Other users have to do the call setup before transmitting. Afterwards, users shall arrive according to a Poisson process with arrival rate (TBD). The session duration is indirectly determined by the size of the file to be transferred. In addition, it is possible to have more users in a sector than the sector capacity can support. In this case, the new arrivals shall be blocked. The blocking rate can be recorded.

2.2.2.1 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. Once the file size is determined, the datagram size and arrival rate depends on the link capacity and underlying TCP implementation.

Figure 4. Packet trace in a typical DL FTP session



2.2.2.2 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 2.

Table 2. 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.	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.006$

2.2.2.3 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.

2.2.2.4 UL FTP traffic model parameters

The only traffic model parameter is the upload file size and can be found in Table 3.

Table 3. UL FTP traffic model parameter

Arrival of new users	Poisson with parameter
Upload file size	Truncated lognormal; lognormal pdf: $f_x \quad \frac{1}{\sqrt{2}\quad x}\exp\frac{(\ln x \quad)^2}{2\ \ ^2}\ ,x \quad 0$ 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

2.2.2.5 FTP and TCP interactions

For people who need to model the FTP and TCP interactions themselves, please refer to 4.1.4.2 of [1] for details.

2.2.3 Near real time video streaming (NRTVS) for DL

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. In addition, it is for DL direction only.

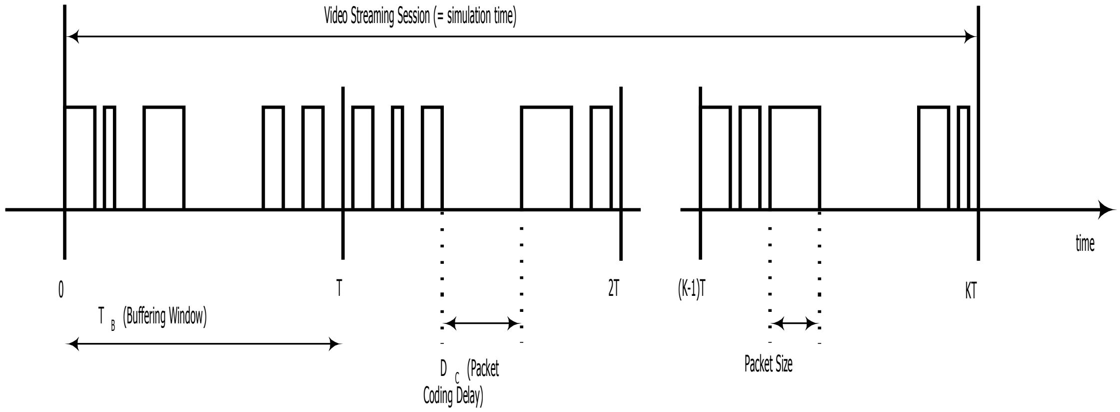
2.2.3.1 NRTVS traffic model characteristics for DL

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.

Figure 5. Video streaming traffic model



2.2.3.2 NRTVS traffic model parameters for DL

The packet sizes and packet inter-arrival rate can be found in Table 4.

Table 4 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

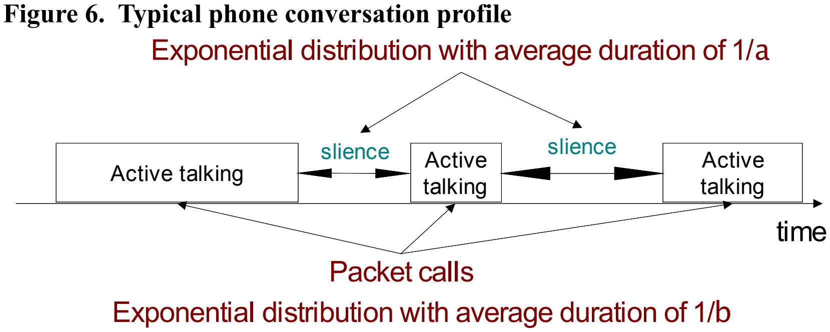
2.2.4 VoIP model

VoIP refers to real-time delivery of packet voice across networks using the Internet protocols. The arrival of a VoIP user can be modeled as a Poisson arrival process with arrival rate (TBD).. The session duration can be modeled by a TBD distribution. The VoIP model description is mainly taken from [1], [3], [6].

2.2.4.1 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.

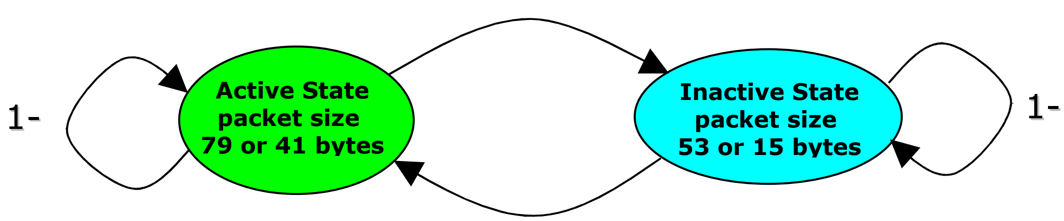
During the active state, packets of fixed sizes are generated at a regular interval. During the inactive state, we have chosen to generate comfort noise with smaller packet sizes at a regular interval instead of no packet transmission. The size of packet and the rate at which the packets are sent depends on the corresponding voice codecs and compression schemes. Table 5 provides information on some common vocoders.

Table 5. 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

To simplify the VoIP model, we propose the use of a simplified AMR (adaptive multi-rate) audio data compression. 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 using AMR



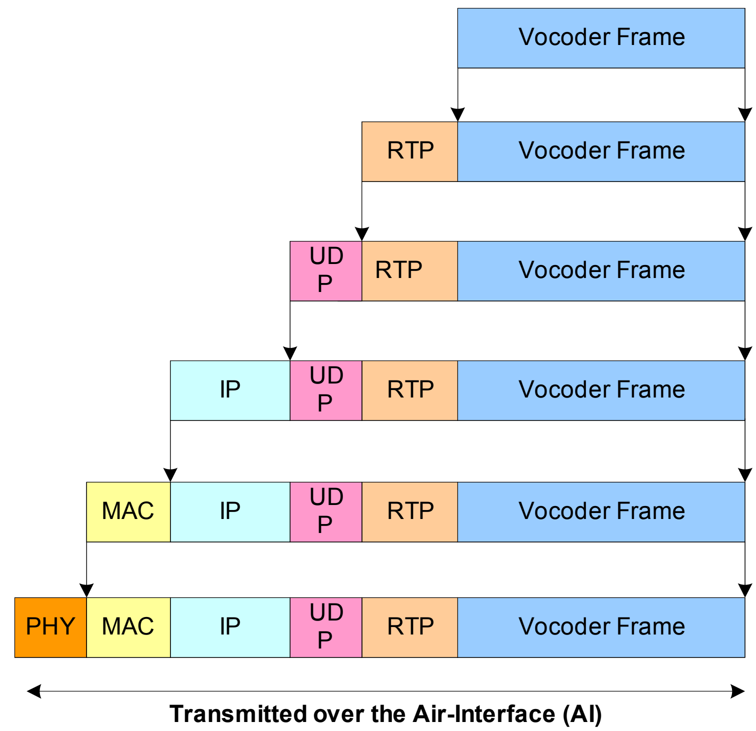
Without header compression, packets of 79 bytes are generated in the active state and packets of 53 bytes are generated in the inactive state. With header compression, packets of 41 bytes are generated in the active state and packets of 15 bytes are generated in the inactive state. Table 6 shows the VoIP packet size calculation with simplified AMR and Figure 8 shows the VoIP packet generation process.

Table 6. VoIP packet size calculation using simplified AMR

Description	**Without Header Compression**	**With Header Compression**

Voice Payload	AMR payload* 7bytes (inactive) / 33 bytes (active)	AMR payload* 7bytes (inactive) / 33 bytes (active)
Protocol Headers	40 bytes	2 bytes
RTP	12 bytes	
UDP	8 bytes	
IPv4	20 bytes	
802.16 Generic MAC Header	6 bytes	6 bytes
Total VoIP packet size	53 bytes (inactive)/79 bytes (active)	15bytes (inactive) / 41 bytes (active)

Figure 8. VoIP packet generation process [6]



2.2.4.2 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 7.

Table 7. 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	(=0.0215)	
Probability of transition from inactive to active state	N/A		

2.2.5 Gaming model

Gaming traffic is generated by users engaged in interactive gaming of multiple users in different locations via the internet. For gaming traffic, the arrival of a user is modeled as a Poisson arrival process with arrival rate (TBD). The session duration (time a user stays and play games) can be modeled by a TBD distribution.

2.2.5.1 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. The packet delay of a dropped packet is arbitrarily set at 180ms. In addition, a mobile network gaming user is in outage if the average packet delay is greater than 60ms.

2.2.5.2 Gaming traffic model parameters

Gaming traffic model parameters for DL and UL can be found in Table 8 [7]. 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 8. 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	$f(x)=\frac{1}{b-a}, a\le x\le b$
Packet 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	$f(x)=\frac{1}{b}e^{-\frac{x-a}{b}}e^{-e^{-\frac{x-a}{b}}}, b>0$ $X=a-b\ln(-\ln Y)+2, Y\sim U(0,1)$

2.3 Traffic mix proposal

To test various aspect of the system, we propose the following traffic mixes:

1. Data only, full buffer (data capacity referenced in Table 9);
2. voice only (voice capacity referenced in Table 9);
3. 8 cases of mixed traffic from Mix -1 to Mix -3 referenced in Table 9.

Table 9. Proposed traffic mixes

	VoIP	FTP	HTTP	n.r.t. video	Gaming

Voice Capacity	100% #users = Nv	0%	0%	0%	0%
Data Capacity	0%	100% #users = Nd	0%	0%	0%
Traffic Mix 1	0.5 Nv	Remaining Data Users 100%	0%	0%	0%
Traffic Mix 2	0.5 Nv	Remaining Data Users 30%	30%	30%	10%
Traffic Mix 3	0.75 Nv	Remaining Data Users 30%	30%	30%	10%

Nd, Nv are the number of users determined by considering the system and link capacity under various propagation conditions.

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.
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4 Acknowledgement

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