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 due to the fact that the authors try to describe the traffic source and the protocol stack interaction with TCP 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 and appendix C of IEEE 802.16j-06/013 [2] with section 2 of this contribution ~~starting from section 2~~. We propose to put our traffic models description in Section 3 and remove Appendix C of IEEE 802.16j-06/013. 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. The full buffer model assumes that there is always data to transmit. This eliminates the traffic arrival statistic effect on system performance evaluation on PHY and lower MAC.

In the following sections, we will concentrate on traffic generation. Interaction of the generated traffic with the protocol stack 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 following topics will be discussed:

- Traffic model description using the generic packet data model;
- Traffic models:
    - HTTP [1][2][7]
    - FTP [1][2]
    - Near Real Time Video Streaming [1][2]
    - VoIP [1][3][4][5]
    - Gaming [1][6]
- Traffic mixes proposal.

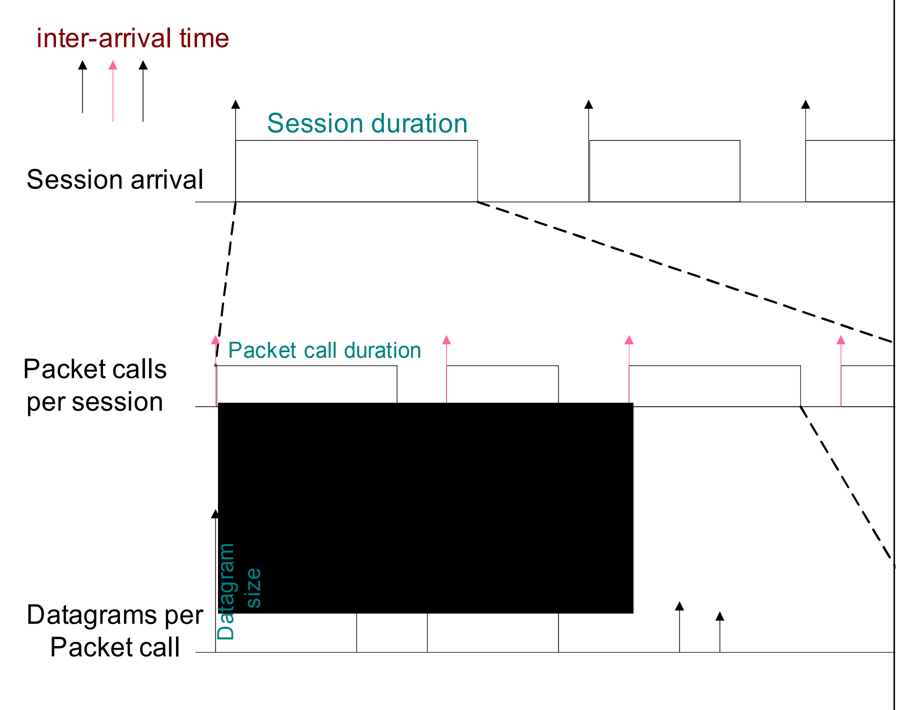
2.1 Traffic Generation

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 for each traffic model in the coming sections.

Figure 1. Bursty traffic generation model



2.2 Traffic models

The following traffic models are proposed:
HTTP;
FTP;
Near real time video streaming (NRT video streaming);
VoIP;
Gaming.

The traffic models apply to both DL and UL unless otherwise specified.

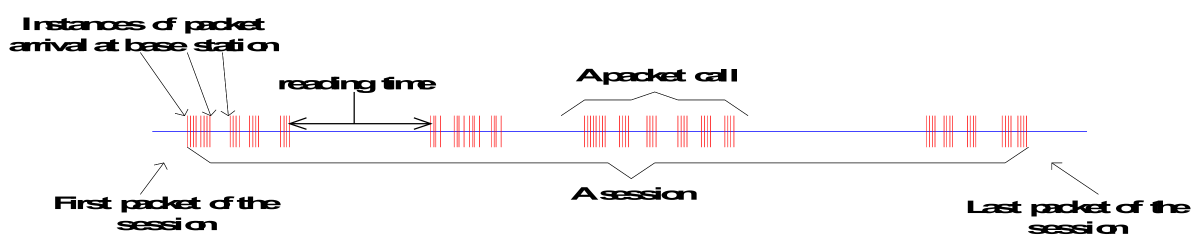
2.2.1 HTTP model

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 is determined by the number of pages per session and the page download and processing time.

2.2.1.1 HTTP traffic model characteristics

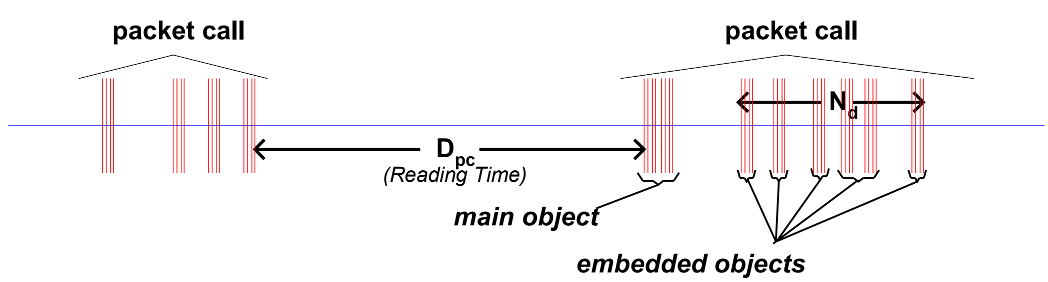
Figure 2 shows a typical web browsing session. Each session is divided into ON/OFF periods representing web-page downloads and intermediate reading times. Each web-page download is referred to as packet calls in Figure 2. During an ON period (packet call), users are requesting information. During an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

2.2.1.2 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 1 HTTP Traffic Model Parameters

Component	Distribution	DL Parameters	UL Parameters	
No. of Pages per session	Lognormal	Mean = 17, Std.Dev = 22	Mean = 17, Std.Dev = 22	$f_x \quad \frac{1}{\sqrt{2} \quad x} \exp \quad \frac{\ln x \quad ^2}{2 \quad ^2}$, $x \quad 0$
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	$f_x \quad \frac{1}{\sqrt{2} \quad x} \exp \quad \frac{\ln x \quad ^2}{2 \quad ^2}$, $x \quad 0$
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	$f_x \quad \frac{1}{\sqrt{2} \quad x} \exp \quad \frac{\ln x \quad ^2}{2 \quad ^2}$, $x \quad 0$
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 5.64, Maximum = 53 1.1, $k \quad 2$, $m \quad 55$	Mean = 4.229, Maximum = 53 1.1, $k \quad 2$, $m \quad 55$	$f_x \quad \frac{k}{x \quad ^{1}}, k \quad x \quad m$ $f_x \quad \frac{k}{m} \quad , x \quad m$ 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.

2.2.1.3 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. Users shall specify 50% (TBD) HTTP/1.0 and 50%(TBD) 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

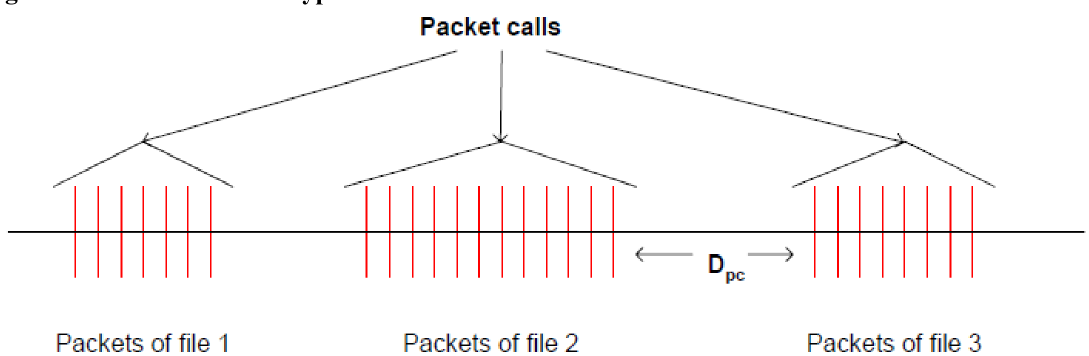
For DL FTP traffic, the arrival of a FTP user is modeled as a Poisson arrival process with arrival rate (TBD). The session duration can be modeled by a TBD distribution.

For UL FTP traffic, users shall arrive according to a Poisson process with arrival rate (TBD). The session duration is determined by the size of the file to be transferred.

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.

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 Minimum = TBD, 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).	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.006$

2.2.2.3 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 λ
Upload file size	Truncated lognormal; lognormal pdf: $f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 2.0899, \mu = 0.9385$ 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

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

2.2.3 Near real time video streaming (NRT video streaming) 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.

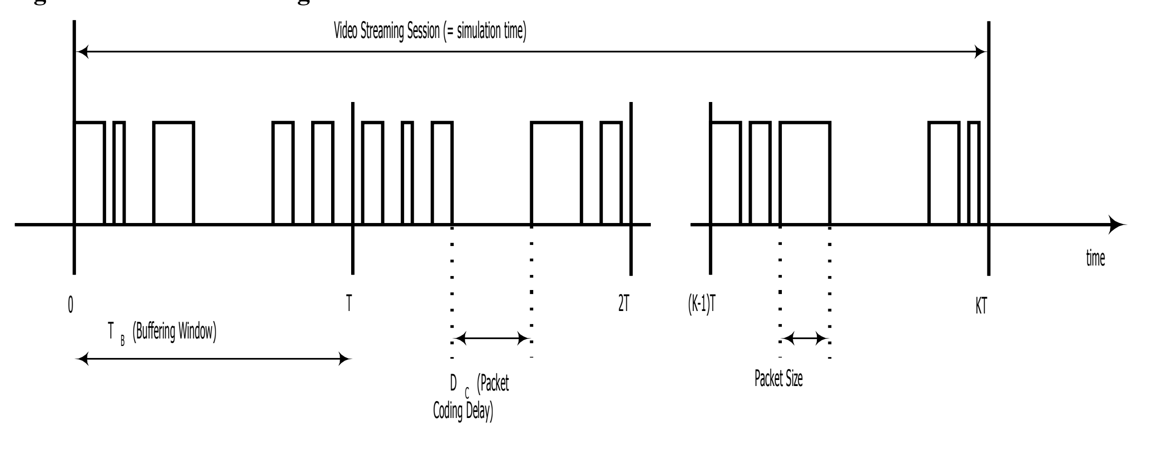
2.2.3.1 NRT video streaming traffic model characteristics for DL

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 NRT video streaming 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	Number of	Packet (slice) size	Inter-arrival time

	between the beginning of each frame	packets (slices) in a frame		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

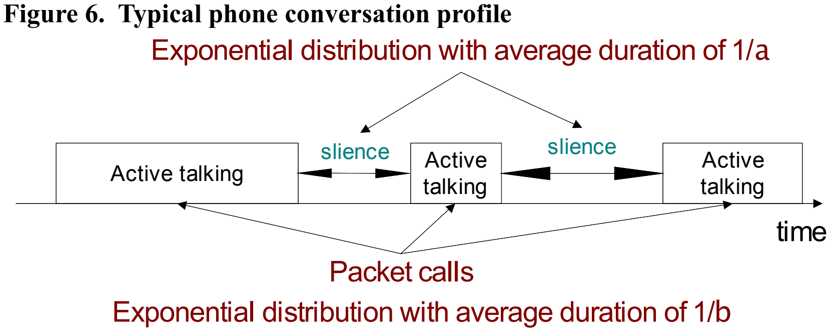
2.2.4 VoIP model

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.

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.

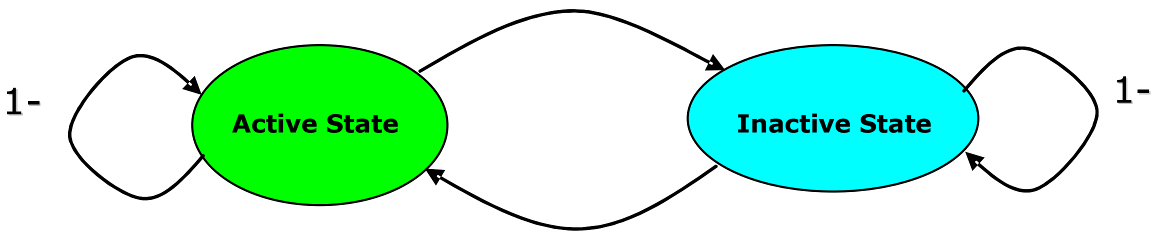
Table 5. Information on various vocoders

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

Among the various vocoders in Table 5, ~~To simplify the VoIP model, we propose the use of~~ a simplified AMR (adaptive multi-rate) audio data compression can be used ~~because it has the advantage of reducing the simulation time of VoIP model~~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 6 shows the VoIP packet size calculation for simplified AMR with or without header compression when using IPv4 or IPv6.

Table 6. 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	6 bytes	6 bytes	6 bytes	6 bytes

MAC Header				
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)

2.2.4.2 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	(=0.6)	
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.

2.2.5.2 Gaming traffic model parameters

Gaming traffic model parameters for DL and UL can be found in Table 8 [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 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 inter-arrival time	Extreme	Extreme	a=48ms, b=4.5ms	a=40ms, b=6ms	$f(x) = \frac{1}{b} e^{-\frac{x-a}{b}} e^{-e^{-\frac{x-a}{b}}}, b > 0$ $X = a - b\ln(-\ln Y), Y \in U(0,1)$
Packet size	Extreme	Extreme	a=330bytes, b=82bytes	a=45bytes, b=5.7	$f(x) = \frac{1}{b} e^{-\frac{x-a}{b}} e^{-e^{-\frac{x-a}{b}}}, b > 0$ $X = a - b\ln(-\ln Y) + 2, Y \in U(0,1)$ Addition of 2 in the equation is due to 2 bytes of UDP header size after header compression.

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. 3 cases of mixed traffic from Mix -1 to Mix -3 referenced in Table 9. 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 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%	0%	0%	0%
Traffic Mix 1 (TBD)	0.5 Nv	Remaining Data Users 100%	0%	0%	0%
Traffic Mix 2 (TBD)	0.5 Nv	Remaining Data Users 30%	30%	30%	10%
Traffic Mix 3 (TBD)	0.75 Nv	Remaining 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.
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