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Re:	IEEE 802.16m-07/005r2 – Call for Contributions on Evaluation Methodology and Key Criteria for P802.16m - Advanced Air Interface
Abstract	This proposal captures specific system evaluation methodology for VoIP service under IEEE802.16m system.
Purpose	For discussion and approval by TGm. VoIP is a dominant service for wireless access network and such service holds distinguished characteristics and QoS requirement. In order to better evaluate system performance for such service, a specific system evaluation methodology for VoIP service is proposed in this document.
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System Evaluation Methodology for VoIP Service

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1. Background: challenge of supporting VoIP service for OFDMA air interface

Voice service is a dominant and profitable service for wireless access operator. In all-IP wireless access network, performance of VoIP service is a critical factor for the success of business deployment. Therefore, support of VoIP service should be carefully and thoroughly considered.

There is a stringent requirement for latency and latency jitter for VoIP service to achieve a satisfying performance. Moreover, VoIP service has a low and variable data-rate if silence suppression and packet header compression are employed. In general, each VoIP packet is in the order of one to several hundred bits, so efficient transmission of these packets over OFDMA air interface becomes a big challenge, especially in uplink transmission.

IEEE 802.16m system shall be flexible and efficient to support required services defined by ITU-R, e.g. VoIP. More specifically:

- Required QoS for VoIP shall be provided including end-to-end latency and error performance etc.
- Higher VoIP capacity and lower VoIP latency than current S3G system (e.g., LTE/AIE) shall be provided.

However, 802.16e and 802.16j do not provide specific evaluation methodology for VoIP service. In order to better evaluate system performance for such service, a specific system evaluation methodology for VoIP service is proposed in this document. Section 2 highlights specific simulation scenario for evaluation of VoIP service. Section 3 covers traffic models for system simulation. Section 4 captures main components of delay statistics. Section 5 addresses VoIP performance metrics for evaluation. Finally, the annex presents a list of parameters of key models and configurations.

2. Simulation Scenario

Mobile-to-Mobile Scenario

In most simulation scenario for VoIP service, separate land-to-mobile (DL) and mobile-to-land (UL) simulations are recommended to be employed. However, for voice conversation, mobile-to-mobile call is a very common usage case. In order to evaluate end-to-end delay for such common case, mobile-to-mobile simulations should be considered, in addition to the land-to-mobile (DL) only and mobile-to-land (UL) only simulations. Figure 1 gives an example scenario of mobile-to-mobile call for system evaluation.

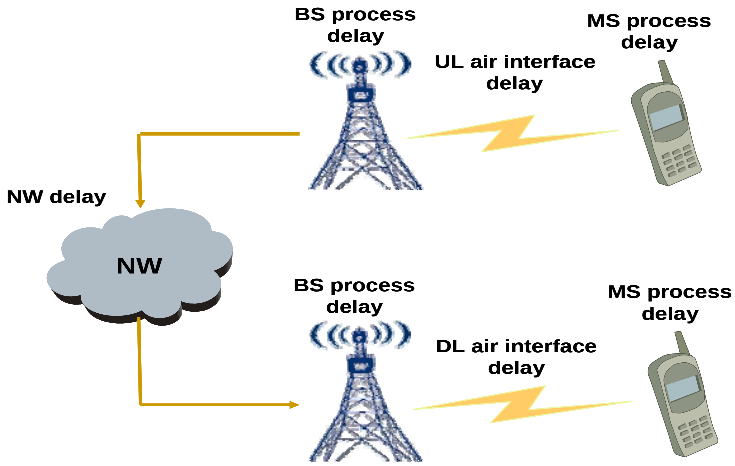

Figure 1: An example scenario of mobile-to-mobile call

The system consists of a network of 19 hexagonal cells with six cells surrounding the center cell in the first tier and 12 cells surrounding the center cell in the second tier. For the default case, each cell has 3 sectors. 57*N_{sub} mobile subscribers are randomly dropped over the 57 sectors such that each sector serves N_{sub} subscribers.

Each mobile subscriber corresponds to an active user session that runs for the duration of a drop. A drop is defined as a simulation run for a given set of subscribers over a specified number of time frames, N_{Frames}. At the beginning of each drop, the subscribers are associated with a specific BS and sector, henceforth referred to as the serving BS. The association is based on both the path loss and shadow fading between the subscriber and the BS, which are fixed for the duration of the drop.

Among all 57*N_{sub} subscribers, 57*N_{sub}/2 randomly selected pairs are assigned for each drop. The pair of subscribers consists of transmitter and receiver of one VoIP session, which is fixed for the duration of the drop.

In order to allow a fair comparison of different proposals, we suggest defining a pre-set layout file, which includes BS position, subscriber position, pair selection, traffic source file, channel type and related parameters.

With the progress of standardization, relay assisted scenario shall be taken into consideration.

Backward Compatible Scenario

In order to evaluate backward compatibility with legacy 16e system, scenario with a mixture of 16e and 16m subscribers is to be investigated. In such scenario, 50% of the subscribers in a sector are 16e subscriber, and the rest 50% are 16m subscriber. The performance of 16e subscribers in this scenario is used to compare with that under 16e only scenario.

Mixed Traffic Scenario

Coexisted different traffic type scenario should also be taken into consideration. For simulations with VoIP traffic only, each subscriber either transmits or receives an audio flow. For simulations with mixed VoIP traffic and other traffic type, only full-buffer flows are considered for the other traffic type in the early stage. For example, the following user configuration may be used:

- 100 VoIP users/sector + 32 full buffer users/sector for TDD 10MHz

3. Traffic Models

There are a variety of encoding schemes for voice (i.e., G.711, G.722, G.722.1, G.723.1, G.728, G.729, and AMR) that result in different bandwidth requirements. In order to keep alignment of WiMAX system evaluation methodology, AMR codec is adopted[1]. The table below lists the key parameters for VoIP traffic model[1]:

Table 1. VoIP traffic model

Parameter	**Assumption**
Average Call Holding time	Exponential Distribution (μ = 210 sec)
Voice Codec	AMR (12.2kbps)
Frame Length	20 msec
Talk Spurt Length	Exponential Distribution (μ = 1026 msec)
Silent Length	Exponential Distribution (μ = 1171 msec)
Silence Suppression	Yes
Header Compression	Yes
Speech Activity	47.17%
DL:UL ratio	1:1

Link adaptation of AMR codec is disabled in order to evaluate performance under worst case. The Voice traffic model specifies only one rate during the ON state (talk spurt) of the AMR codec (12.2 kbps) and another rate for the comfort noise (AMR_SID) during the OFF state of the AMR codec. The 1/8th frame rate blanking is modeled by only transmitting the first 1/8th rate frame of each silence interval.

We also suggest defining a pre-set traffic source file with link adaptation of AMR codec enabled, such that full-scale and realistic performance can be evaluated.

PHS (Payload Header Suppression) is to be modeled at MAC layer to capture most common usage scenario. For example, model RTP/UDP/IP overhead as 4 bytes constant overhead. And for the simplicity, signaling traffic is not modeled.

4. Delay Statistics and Analysis

The table below lists the main components of mouth-to-ear delay for VoIP service.

Table 2. Main components of mouth-to-ear delay for VoIP service

Delay Component	Value	Comment
MS process delay	35ms*	Including source encoder, channel coding, modulation, and interleaver processing delay
UL air interface delay	Simulation Statistics	Including request delay, schedule delay, and HARQ retransmission delay

BS process delay	10ms*	Including channel decoding, demodulation, deinterleaver, and reorder processing delay
NW process delay	35ms*	Including network routing, scheduling, blocking, and retransmission processing delay
BS process delay	10ms*	Including channel coding, modulation, and interleaver processing delay
DL air interface delay	Simulation Statistics	Including schedule delay, and HARQ retransmission delay
MS process delay	50ms*	Including channel decoding, demodulation, deinterleaver, source decoding, and de-jitter processing delay

*Note: the value is open for discussion.

Target end-to-end delay is the sum of all the above components, which is used to study level of satisfaction to the service requirement. We suggested that 95% of the end-to-end delay CDF shall be less than or equal to 285 ms in accordance with the corresponding value specified in ITU recommendation.

5. Performance Metrics

The following describes the performance metrics for VoIP service when evaluating the system performance. For mixed data and VoIP scenario, the performance metrics for data service defined in [1] should also be used.

VoIP Specific performance metrics

- Spectrum Efficiency for VoIP Service (users/sector/MHz)

Spectrum efficiency for VoIP Service is defined as

$$\frac{N_{\text{sub, 5\% outage}}}{\text{Total Assigned BW/MHz}}$$

Where $N_{\text{sub, 5\% outage}}$ denotes the maxium number of VoIP service subscribers that can be supported in a sector with 95% of the end-to-end delay CDF less than or equal to the target delay value, for example, 285 ms for mobile-to-mobile scenario.

- Application Packet Erasure, Drop, and Success

An application packet sees one of three possible outcomes: erasure, drop, or success.

An application packet erasure occurs when any portion of the packet is lost on the channel, after any H-ARQ or other re-transmissions have failed.

An application packet drop occurs when no portion of the packet is ever placed in a physical layer packet for transmission.

An application packet success occurs when the entire packet is successfully transmitted.

At the end of the simulation run, application packets that have any portion of their data still in queue or in flight (i.e. not yet successfully delivered) are considered dropped.

If performance on air interface is to be studied, PER is defined as erasure rate. While end-to-end system performance is to be studied, PER is defined as total sum of erasure rate and drop rate.

- Application Packet Delay

The delay for a successfully delivered application packet is the time between packet arrival and successful transmission of the last bit in the packet. For erased or dropped application packets, delay is defined to be infinite.

When performing end-to-end VoIP simulations, the packet delay is the time between packet arrival at the uplink until successful transmission of the last bit in the packet on the downlink. The statistics value can be shifted by the constant delay equal to the sum of network and processing delay.

- Loss and Drop CDF

Statistics on proportion of dropped and erased packets are to be kept separately for audio packets for each subscriber. A CDF of each of the audio dropped packets over all subscribers is to be provided, and similarly for erased packets.

- Audio Tail Delay

For each subscriber, determine the audio tail delay point as follows. A delay parameter is defined, TargetAudFER (in percent). Find the smallest delay value D such that TargetAudFER percent or less of the audio packets have delay equal to or longer than D. If no finite D exists that satisfies this criterion (i.e. more than TargetAudFER percent of the audio packets are dropped or erased), then D is defined to be maximal allowed value. This delay value D is the audio tail delay associated with that specific subscriber, and each subscriber has exactly one such value.

When performing end-to-end VoIP simulations, the audio tail delay point is determined for each subscriber based on the total delay packets experience end-to-end, cumulative of both uplink and downlink delays.

We use the following value for VoIP for both DL and UL scenarios: TargetAudFER = 2%.

A CDF of the value of the audio tail delay over all VoIP subscribers is to be provided as part of the VoIP simulation results. An approach to get a coarse estimate of the end to end delay statistics is to convolve the DL delay PDF with the UL delay PDF. The resulting CDF can be shifted by the constant delay equal to the network and processing delays.

- Per-Subscriber Result Data

An ASCII text file is to be generated for each VoIP simulation run, which contains one line for each subscriber that runs VoIP in the simulation. The VoIP statistics described above are to be included for each such subscriber.

The following data is to be reported in an ASCII text file, one line per subscriber, in the following format.

Subscriber# PhyTHP PhyPER AudTHP AudFER AudTailDelay

Subscriber#: Reference Subscriber# for reporting subscriber statistics, from source configuration file

PhyTHP: Physical layer throughput (kbps)

PhyPER: Physical layer packet error rate

AudTHP: Total audio bits delivered divided by simulation time (kbps)

AudFER: Audio packet error rate (%)

AudTailDelay: TargetAudFER% tail delay point (msec)

- UL Traffic IoT Criterion

The Interference over Thermal (IoT) criterion will be used for the simulation of 16m system uplink. The IoT is computed per sector per subframe and is defined as

$$IoT_s = \frac{\sum_{i:BS(i)\neq s} E_{c,i,s} + N_0}{N_0}$$

Where *BS(i)* represents the best serving sector of subscriber *i*, $E_{c,i,s}$ is the total received power from subscriber *i* at sector *s*, and N_0 is the thermal noise variance.

The IoT statistics are collected from all sectors of the system. The uplink operation shall use an IoT criterion of a mean value of 6 dB . Target IoT should be achieved on the mean value with the 1% tail of the IoT distributions being only 1dB away from the target. Any value larger than 6dB also can be submitted such that the other performance criteria are not violated. The IoT for the default two receiving antenna mode is [(IoT_1+No)/No + (IoT_2 + No)/No]/2, where the total received signal power at antenna *i* is defined as IoT_i, i=1,2.

- Power Usage

Statistics on power usage and power efficiency is a key factor of system performance. The CDF of total instantaneous transmit power over all BSs on the downlink and total instantaneous transmit power over all subscribers on the uplink is to be provided.

Annex

Proposed baseline of simulation assumptions for network configuration

Parameter	Description	Assumption
Nc	Network Topology and number of cells	Hexagonal Grid, 19 cells
S	Number of sector per cell	3 (1, 3, 4, 6)
R	Site-to-Site distance	0.5-30Km (1Km)
K	Number of frequency allocations in the network	1, 2, 3, 4, 6
F_{BS}	Frequency allocation used in each sector	1, 2, 3, 4, 5, 6

Fc	Carrier Frequency	2.0-3.5GHz (2.5GHz)*[3]
BW	Bandwidth	10MHz(TDD)

*Note: the value will in accordance with definition in ITU Recommendation

Proposed baseline of simulation assumptions for base station configuration

Parameter	**Description**	**Assumption**
P_{BS}	RMS transmit power per sector/carrier	30-51dBm (43 dBm)
G_{BS}	Gain (boresight)	16 dBi
H_{BS}	BS Antenna Height	10-50m (32m)
θ_{BS}	Horizontal Antenna Pattern	70 deg @3 dB bandwidth, (s=6, 35 deg @3dB bandwidth)
	Vertical Antenna Pattern	None
G_{FB}	Front-to-back Power Ratio	25 dB
M_{TX}	Number of transmit antenna	1, 2, 3, 4
M_{RX}	Number of receive antenna	1, 2, 3, 4
d_{BS}	BS antenna spacing	λ/2, 4λ, 10λ,
NF_{BS}	Noise Figure (transmit & receive)	4-6 dB (5 dB)

Proposed baseline of simulation assumptions for subscriber station

Parameter	**Description**	**Assumption**
P_{SS}	RMS transmit power per sector/carrier	20-45 dBm (23 dBm)
G_{SS}	Gain (boresight)	0 dBi
H_{SS}	SS Antenna Height	1.5-7m (1.5m)
θ_{SS}	Horizontal Antenna Pattern	Omni
	Vertical Antenna Pattern	None
N_{TX}	Number of transmit antenna	1, 2
N_{RX}	Number of receive antenna	1, 2, 3, 4
d_{SS}	SS antenna correlation	0-0.7 (0.5)
NF_{SS}	Noise Figure (transmit & receive)	6-7 dB (7 dB)

Proposed baseline of simulation assumptions for propagation model

Parameter	Description	Assumption
Fc	Carrier Frequency	1.9-2.5GHz (2.5GHz)
PL	Path Loss Model	COST-HATA-231, Erceg
σ_{SF}	Log-Normal Shadowing Standard deviation	8-12 dB (8.9 dB)
ξ_{SF}	Site-to-Site Shadow Correlation Coefficient	0.5
d_{SF}	Correlation Distance of Shadowing	50m
σ_{N}	Thermal Noise Density	–174 dBm/Hz

Proposed baseline of simulation assumptions for multipath model

Type	Description	Speed	Assignment Probability
Flat fading, Ch-100	1 paths, Jakes fading spectrum	30 km/h	0.1
Flat fading, Ch-100	1 paths, Jakes fading spectrum	120 km/h	0.1
ITU VA Ch-104	6 paths, Jakes fading spectrum	30 km/h	0.1
ITU VA Ch-104	6 paths, Jakes fading spectrum	120 km/h	0.1
ITU PA Ch-102	4 paths, Jakes fading spectrum	3 km/h	0.3
ITU PB Ch-103	6 paths, Jakes fading spectrum	3 km/h	0.3

Proposed baseline of simulation assumptions for system modeling

Item	Assumption
Network Synchronization	Idea Synchronization of BS
Multiple Access	OFDMA, optional SDMA
Resource Allocation	Centralized at BS; early stage Proportional Fairness/Round Robin
Spatial Scheme Control	Early stage fixed per connection
Spatial Processing	SDMA; Beam-forming; SM; STC
Coding	CC, CTC, LDPC
Interleave	Random
Modulation	M-QAM with Gray Mapping, M=2, 4, 16, 64
DL Power Control	PC might be applied to individual connection, but overall transmit power should be constant unless interference avoidance techniques are carried out

UL Power Control	Perfect (signals of all simultaneously scheduled SS in the uplink arrive with the same average power at BS)
Modulation and coding rate selection	Determined based on average SINR of FEC block
Delay of CQI	2 subframe time
Protocol of HARQ	N-Channel SAW (sync. / async.)
HARQ Type	CC / IR
Maximum Retransmissions for HARQ	0, 4, 12
Delay of HARQ	Delay of ACK/NACK*; Delay of Retransmission*

*Note: the value is open for discussion and should be determined with further progress of standardization.

Reference

[1] WiMAX System Evaluation Methodology, V1.0, Created on 1/30/2007, WiMAX FORUM.