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Effect of Heavy Real-Time Service on Subscribers Number Based on 802.16e

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Abstract: Video conference service is an application that offers video and audio at the same time, and it is one of the real-time applications that deplete the capacity of any system significantly. Fortunately, the IEEE 802.16e standard offers a specific quality of service that can support the desired applications for a limited number of users as there is a tradeoff between the quality and quantity of system capability. In this paper, the number of subscriber units which can be served successfully by a single base station based on the IEEE 802.16e standard for different video conference applications has been evaluated in three cases depending on the circumstance of the end terminal case in terms of modulation and coding rate schemes: the worst case, the random case, and the perfect case. Due To our best knowledge, this paper considered as the first one deals with the capacity in these three cases in term of modulation schemes. The metrics of performance for this study are the end-to-end delay (ETE), the difference between the received to transmitted traffic, the delay variation for different video conference applications, and the amount of the free resources of the cell.

Keywords: IEEE802.16e, ETE Delay, QoS, Real-Time Polling Service (rtPS), Video Conference.

1. Introduction

Multimedia services, such as video conferences, interactive games, and IP television, are spreading dramatically in the recent years in our daily life. Due to the high demand for such services, high data rates need to be provided. To offer such high data rates, many wireless communication systems have been discussed [1]. WiMAX can be considered as a promising wireless system that can meet the users' high data rate demands. WiMAX is based upon the IEEE 802.16e standards to support fixed and mobile stations. In IEEE 802.16e, WiMAX adopts the OFDMA (orthogonal frequency division multiple access) as a way to support the flexibility and the high data rates [2].

In the case of video conference applications; the number of successfully served users by a single cellular cell is a critical factor to indicate the system performance. In other words, it is important to answer the following question: what is the maximum number of users that can be served under these three cases: worst, normal and perfect conditions?

To answer this question, researches in [3-9] trend to focus on one of these two cases: first, calculate the End to End delay (ETE delay) and jitter of video conference for a specific number of users. Second, analysis the schemes of Modulation and coding rate of WiMAX system to explain the difference between these schemes in case of real-time applications.

K. Kaur and V. Grewal in [3] presented scenarios for analyzing QoS of video conferencing in terms of QPSK, 16QAM, and 64QAM modulation schemes, and coding rate equal to 3\4 for four cells contained on fixed and mobile stations (48 stations). They assumed an acceptable End to end delay is less than 400 msec, they also dealt with each scheme of those modulations alone. In their paper they concluded that QPSK modulation scheme performance is better in case of traffic received, throughput, and delay, whereas the performance of 64QAM is good in traffic sent, load and packet end to end delay.

Haider. M. T, Shatha K. Jasim and Mustafa S. M. [8] studied and compared different kinds of modulation techniques like QPSK, 16QAM, and 64QAM of WiMAX system in Baghdad city. The result, which has been taken, based on Block error rate (BLER) delay, throughput and signal to noise ratio appeared that the maximum voice and video delay was found in QPSK type, whereas the maximum throughput of voice and video applications was found in 64QAM. Khan M. and Bandhu [9] evaluated the

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WiMAX performance system when the number of nodes and distance is increasing. They studied the network in two algorithms Round Robin (RR) and Strict Priority (SP) and analyzed the performance of throughput and Goodput. Applying these algorithms is very important to determine the maximum number of users in terms of data usage and calls. However, in order to provide continuous service in case of video conference application, it is vital to any provider to evaluate the point that caused system breakdown based on nominal conditions and users number. Yet, the applications of real-time affected with more than one factor related to the time. Furthermore, other factors in addition to application kind form whether the service is submitted successfully or not.

The overall problem statement of this manuscript is determining and knowing the number of users that can be successfully served with video conferencing service by one cell of the 802.16e standard before the breakdown and loss of service occurred, in three cases: perfect, worst and random which represents the real case so, we offer a criterion to evaluate the number of users before congestion occurs. This criterion depends on three points together at collecting the results. The first point is allocating a threshold for the time variation (jitter) of the application, when the value of jitter exceeds the threshold then the congestion of system capacity starts. The second factor is related to the end to end delay of the application (the time that elapses from the application layer of the source until the information reaches to the application layer of the destination, the values of the end to end delay and jitter are explained in section three. The last point is the ratio of transmitted traffic to received traffic. If the value of this factor increases then the lost traffic of the system will increase, the research considers the ratio of larger than 1%, which means that the system is going to collapse. This work evaluates the performance of the WiMAX cell in three cases of signal strength of the users: firstly, when the users have a perfect signal. Secondly, when the users face bad condition. The third case submits the normal case, when the users have random distribution where they will have various schemes of modulation and coding rate according to the standard of 802.16e.

This paper consists of following sections. Sections two describes the service classes of 802.16E. The proposed methodology is explained in section three, while section four presents the results and discussions. The last section provides the work conclusions.

2. IEEE 802.16E STANDARD

In this section the service classes and the video conference application have been described for IEEE 802.16e standard.

A. Service Classes of 802.16e Standard

In order to apply QoS to various applications, each connection link from a user to BS and vice versa is characterized by service class flow. IEEE 802.16e standard supports the following scheduling of service classes which corresponds to different traffic types: Unsolicited data grants (UGS), Real-time polling service (rtPS), non-real-time polling service (nrtPS), extended real-time polling service (ertPS), and Best effort (BE). This paper focuses on the service classes which endorse the real-time applications (UGS, rtPS, and BE).

UGS class generates fixed-size data periodically where the criteria of transmission opportunities for users do not depend on the request for transmission. Unlike UGS, rtPS class generates variable size data and allows users to specify the size of the desired grants depending on their requests. In the case of BE, the style of service class is different. It is allowed for each user to submit the request fairly. This type of service class gathers between the unicast requests from users to BS and unsolicited data grants [10].

B. Video Conference Application

A video consists of consecutive images that are displayed at a rate of images per second. These images that digitally encoded contains an array of pixels. The mechanism of representing the luminance and color is done by encoding the pixel into a number of bits. The nature of the video gives a dilatation in spatial. This is related to a given image or video, which consists of mostly white spaces. Moreover, there is an issue of temporal redundancy when a similarity might occur in two consecutive images. Thanks to compression techniques, which compensate this overhead of bits. However, the compression techniques make a balance between the quality of video and the consumption of systems capacity [11].

Video frames are compressed by using H.264 or H.265 encoder to create a compressed video bit stream. Firstly, each individual frame is broken up into blocks of pixels. Then, the blocks are analyzed for spatial and temporal redundancies among frames in order to take advantage of areas with no change on the origin [12]. The H.265 standard is an amendment to H.264, it can definitely submit better compression with the same picture size thereby enhance the quality. This is related to a high-efficiency video codec (HEVC) sequence which can occupy less transmission capacity than the equivalent to the H.264 video sequence. Consequently, resolutions should be higher than the H.264 video sequence [13][14].

Video conferencing application is one of the video types, it can be defined as a live connection among users in different far locations for communication purposes. This application includes video and audio. Unfortunately, the video conference consumes a high data rate where the bit rate varies from 100kbps for the low quality of video conference to 3Mbps for streaming HD (high definition) video [1]. Moreover, it is highly sensitive to ETE delay where there is no agreement about the optimum value of ETE for this application. Some references refer that ETE delay less than 400 msec is considered acceptable [3], while others say that ETE delay should be less than 150 msec such as [6] or below 100 msec [15].

Jitter or delay variation of sequential packets is an influential factor in a video conference, if the value of jitter is increased then there will be negative implications on this real-time application, it should not exceed 30 msec [6].

In addition to ETE delay and jitter, packet loss is another metric performance for video conference. It is preferred to not exceed 1% [5][6].

3. Methodology and Model Description

The point that indicates the phase of system falling depends on the characteristics of video conference application as well as the capability of the system itself. The method of this paper which has been cleared below as pseudo code considers four main aspects to indicate that the system provides the service of video conference without distortion: the first is that the ETE delay should be less than 100 msec, the second is that the max of jitter (delay variation) should be less than 30 msec, the third is that the percentage of the ratio of received traffic compared to send traffic of video conference application is less than 1%. The last one is the availability of free resources in the cell of the IEEE 802.16e standard to avoid the mechanism of the block to the new users. In many cases, the system succeeds in serving a specific number of users, but blocks any new requests so it is important to locate the free resources. The first three points are related to video conference application attribute, while the last one is related to video conference.

A. The Pseudo Code Algorithm

The algorithm that is followed to serve a new user in the system:

1. Initialization.
2. Firstly, BS computes the free resources of the system (symbols) based on the assumptions (Table I) according to the standard of 802.16e (the free capacity of the system).
3. Locate the type of scheduling (UGS, RTPS, or BE). Then, the users initiate the sessions with BS to pull the resources (symbols) in order to enjoy with the service of video conference.
4. The users employ the grant symbols from BS based on the strength of signal to noise ratio of each user. Where the scheme of modulation and coding rate for each user based on the magnitude of S/N at user station. High strength of S/N means high scheme of modulation and coding rate (more data rate) and vice versa.
5. The link between the user and BS will identify, then the information (the data of video conference) will transfer between them.
6. The successful connection of the link between BS and the user to provide the service of video conference depends on the threshold values of (the condition of: the value < the threshold): the end to end delay, jitter, and traffic received to traffic sent. Else, the service will go to failure.
7. For the new user, return to second point.
8. The end.

B. Model Description

The model under discussion consists of one cell which contains one BS connected to a server by 1 Gbps Ethernet cable. This BS takes the service of video conference from the server and gives it to users depending on the IEEE 802.16e standard.

Each user in the cell enjoys the serviceof video conference if the resources of BS is still available. The data rate value of the video conference is adopted to 384Kbps [16][17]. This value satisfies the minimum requirements of appropriate video conference applications (voice and successive images). With respect to users, the users are entered to BS one by one, these users were assumed to be fixed stations in the model.

The complete assumptions of the model are listed in Table I. It is worth mentioning that a suitable environment has been provided for the model to done correctly using OPNET modeler version 14.5.

C. Scenarios of Model Simulation

The implementation of model simulation falls into three scenarios. First represents the worst case where all subscribers use robust modulation (QPSK1/2). While the second case deals with perfect circumstances that qualify users to use a higher scheme of modulation and coding rate (64QAM3/4).

The last case is the random distribution, which illustrates the real distribution of users in the practical situation. In this case, each user is expected to deal with different signal to noise ratio based on the location to the base station and on the environment. For example, if a user is close to the base station the user can perform high modulation order, such as 64QAM. On the other hand, if a user is further away from the base station the signal will be lower and the link path could face more obstacles. To save the reliability in this situation, the modulation order decreases, QPSK.

TABLE I. MODEL DESCRIPTION

Parameter	Value
Profile type	OFDMA
Frame duration	5m sec
Service class type	rtPS
Minimum guaranteed data rate	384 kbps
Frame per second of video	30 frames/sec
Symbol duration	102.86 µsec
No. of subcarriers	2048
Duplexing technique	TDD
Bandwidth	20 MHz
UL subframe size (symbols)	50%
DL subframe size (symbols)	50%
Simulation time	10 min
No. of BS sectors	One
Propagation speed of cable	Min prop. speed

In randomly distributed model, six schemes of modulation and coding rate of the IEEE 802.16e standard are considered: QPSK1/2, QPSK3/4, 16QAM1/2, 16QAM3/4, 64QAM2/3, and 64QAM3/4, for up to 40 users, as shown in Table II.

TABLEII. DISTRIBUTION OF USERS' NUMBER OF RANDOM CASE

Modulation And coding scheme	Number of users								
	1	*5*	*10*	*15*	*20*	*25*	*30*	*35*	*40*
QPSK1/2	0	1	1	2	3	4	5	6	7
QPSK3/4	1	1	2	3	4	5	5	6	7
16QAM1/2	0	1	2	3	4	5	5	6	7
16QAM3/4	0	1	2	2	3	4	6	6	7
64QAM2/3	0	1	2	3	3	3	5	6	6
64QAM3/4	0	0	1	2	3	4	4	5	6

4. RESULTS AND DISCUSSION

A. ETE Delay Results

1) Results of the worst scenario

As mentioned before, all users, in this case, use the QPSK1/2 scheme to compensate for the effect of the weak SNR (signal to noise ratio). This case represents the capability of the cell to treat bad circumstances.

Fig.1 explains the relationship between the average value of ETE delay of video conference application and the increment of the number of users. The relevance value of ETE delay in case of video conference should be less than 100 msec to provide an acceptable service [15]. It is obvious that ETE delay climbs sharply after 12 users from about 35 msec to reach to greater than 600 msec at 13 users where the cell of IEEE802.16e fails to provide acceptable service for the application from ETE delay point of view.

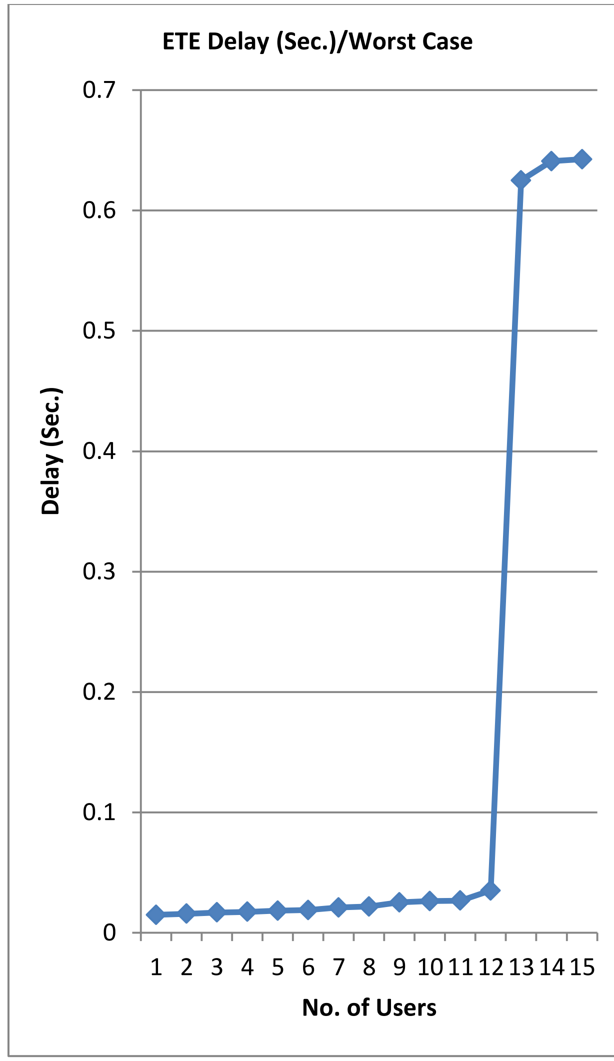

Figure 1. Result of ETE Delay In The Worst Case

2) Results of Perfect and Random Scenarios

ETE delay with the increment of users is shown in Fig. 2. In the case of random and perfect scenarios. It is noted that the result of the worst case is offered alone due to show results clearly without distortion.

In the perfect scenario, there is a great hop in ETE delay between 37 users and 39 users, whereas the magnitude of ETE delay at 37, 38, and 39 users are 22.1 msec, 22.7 msec, and 614.5 msec respectively. While in the case of a random scenario, the collapse of the ETE delay value starts at 21 users. The values of ETE delay in case of the random scenario at 20, 21, and 22 users are 22.5 msec, 152 msec, and 256.7 msec respectively.

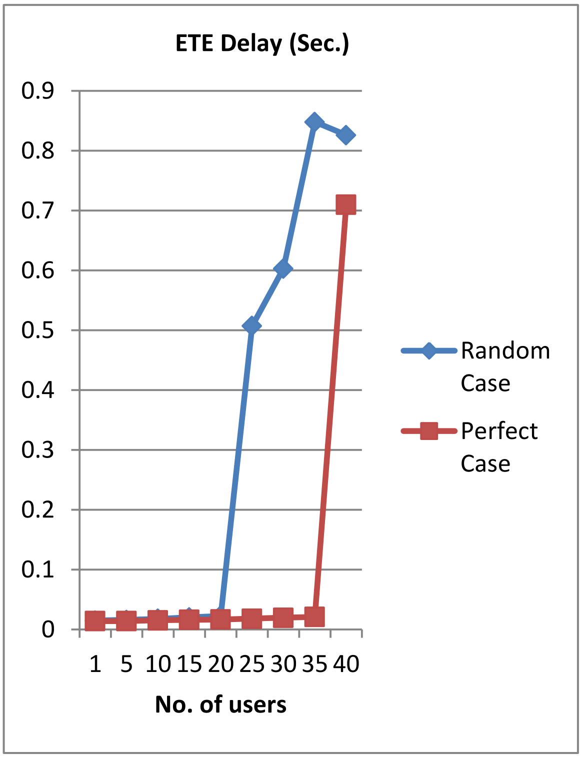

Figure 2. Result of ETE Delay In Perfect and Random Cases.

B. *Traffic Results*

1) *Results of the worst scenario*

Fig. 3 shows the average of the sent traffic and the received one in (bytes/sec) with the users' increment in case of all users work at QPSK1/2 modulation and coding scheme. There will be a noticeable difference between the sent traffic and the received one after 12 users. If we have 12 users, the average sent traffic and the received ones are about 1140415 bytes/sec and 1140322 bytes/sec respectively.

While in the case of 13 users the average sent traffic and the received ones become about 1222460 bytes/sec and 1206609 bytes/sec. It is clear that the ratio of the received traffic to the sent traffic at 12 users is 99.99% while in the case of 13 users becomes 98.7%. According to [6].

To provide an acceptable video conference service, the packet loss should be below 1%.

2) *Results of Perfect and Random Scenarios*

Fig 4. Sheds light on the relationship between the sent traffic and the received ones in cases of perfect and random scenarios. It can be inferred from the figure that, there is a point indicates the beginning of the loss of the received data. The system breaks down in cases of random and perfect scenarios start at 21 and 38 users respectively. From collected results, the ratio of the received traffic to the sent ones at 20, 21 and 22 users of random scenarios are 99.99%, 98.5%, and 97.5 respectively.

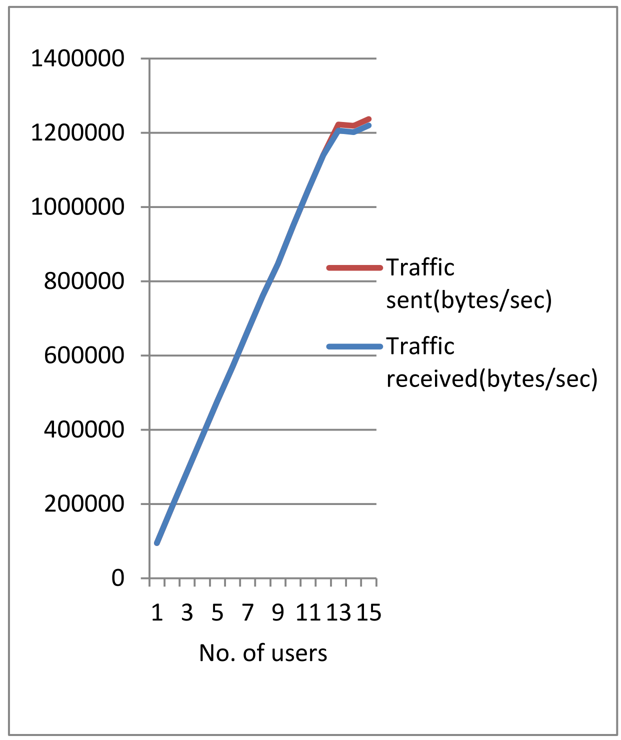

Figure 3. Sending and Receiving Traffic In The Worst Case

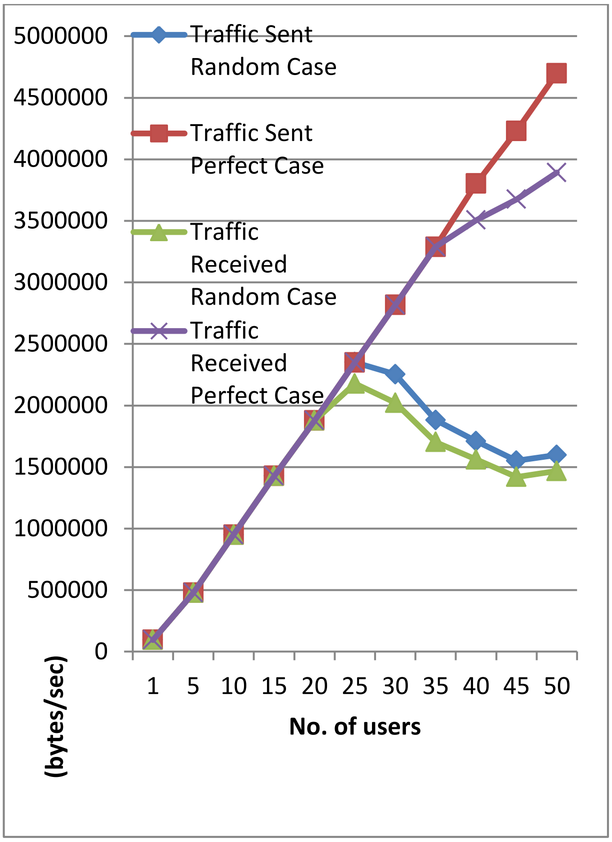

Figure 4. Sending and Receiving Traffic In Perfect and Random Cases.

In the case of the perfect scenario, the ratio of the received traffic to the sent traffic at 37, 38, and 39 users are 99.99%, 99.99%, and 94.1%, respectively.

C. The Results of Packet Delay Variation

Table III illustrates a comparison among the three scenarios at a nominal number of users to explain the state of the cell before and after the system breaks down from a jitter point of view. According to [6], the appropriate value for delay variation in case of the video conference is less than 30 msec.

TABLEIII. JITTER AMONG THREE SCENARIOS

Scenario	No. of users	Max of jitter
Worst	11	< 165 µs
	12	<220 µs
	13	<260 ms
Random	20	<200µs
	21	<450 ms
	22	<620 ms
Perfect	37	<150 µs
	38	<145µs
	39	<550 ms

The system collapse changes from one scenario to another. This is related to schemes of modulation and coding rate. The perfect scenario collapses at 39 users while the worst case falls at 13 users. However, the real situation of users' distribution explains the real state of cell capacity. In the random scenario, the capacity of cell is efficient up to 20 users. The performance of delay variation (jitter) acquires random behavior depending on the different types of modulation. The number of bits per each transmission batch changes from schemes of modulation and coding rate to another thus the behavior will be changed from scenario to another.

D. Results of Free Resources of the Cell

Fig. 5 and Fig. 6 explain the relationship among the three scenarios from the free resources of the cell with the increment of users' points of view. Free resources of the cell are based on the length of the frame, symbol duration, and bandwidth as well as other factors of the IEEE 802.16e standard. It can be measured by symbols per second. As a rule, the resources of the cell start from constant number then gradually decrease with users' increment.

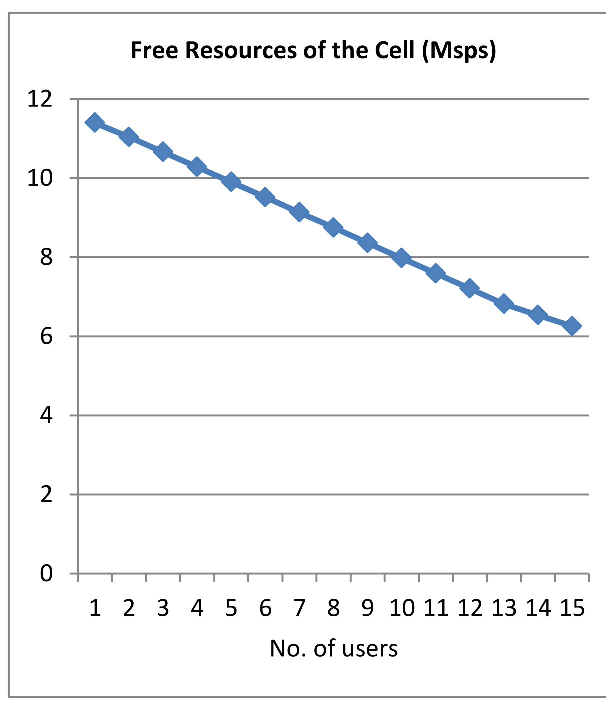

Figure 5. Free Resources of The Cell In The Worst Case.

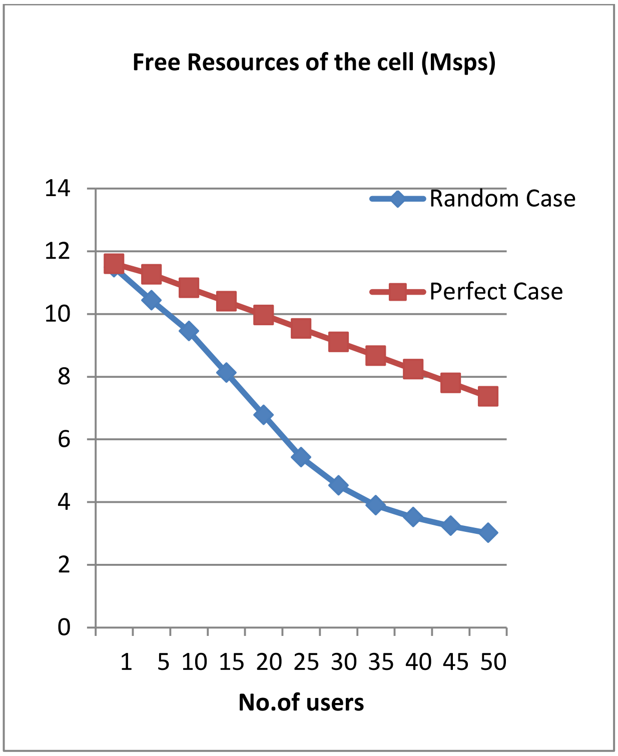

Figure 6. Free Resources of The Cell For Perfect and Random Cases.

The resource consumption depends on the scheme of modulation and coding rate as well as the minimum reserved bandwidth for nominal application. The video conference application is one of the most intensive resource applications. It is worth mentioning that, the value of the minimum reserved bandwidth of rtPS service class has a leading role in resource consumption.

Obviously, the consumption of resources with users' increment takes a linear relationship in the cases of worst and perfect because the users' schemes of modulation and coding are fixed. While in the case of random distribution, the relationship between the consuming and users increment is nonlinear due to the schemes of users are diverse.

E. Results Comparison among Service Classes of 802.16e

Table IV explains a comparison among three types of service classes (rtPS, UGS, and BE) in terms of average ETE delay, the ratio of the received traffic to the sent ones and the free capacity of the cell. The choice to comparison among these types of service classes is related to their relationship with real-time applications.

The results are collected with respect to the random scenario at 20 users. The aim of this comparison is to explain the action of service class on the video conference application.

The results show that the most convenient service class for video conference application is rtPS. The impact of resource exhaustion seems clear in the case of UGS, while BE is more resources conservative. On the other hand, rtPS appears suited to video conference application in all respects.

TABLEIV. COMPARISON AMONG rtPS, BE AND UGS

Service class	rtPS	BE	UGS
ETE Delay (ms)	22.59	721.36	1355.31
Free capacity of the cell (MsPs)	6.77	11.65	5.44
Jitter (less than)	290µ sec	535msec	352m sec
Received to send traffic %	99.99	90.16	91.7

5. CONCLUSION

The performance of IEEE 802.16e system cells as a function of the number of users has been evaluated. In this paper, random cases of users' distribution, which represents the realistic situation, has been considered. The paper considers the metrics of video conference performance that locates the successful support for users (end to end delay, the ratio of the received traffic to the sent ones, delay variation and availability of free resources for the cell of 802.16e which serves the subscribers). The results show that the cell successfully served 12 users in bad conditions for SNR before system congestion, a maximum limit of 38 users in the ideal conditions, and around 20 users in the case of random distribution. In the random case of users' distribution, it is recommended to nicety the minimum guaranteed reserved traffic rate of rtPS service class to minimum value, which can provide a successful and satisfying service of video conference application. This is related in case of increasing this value to consume the resources of the BS quickly to provide high-quality service.

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