Evaluation of a new Scheduling Scheme for VoIP with Mobility in 3G LTE

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***Abstract*- 3G long term evolution is an all internet protocol based network and one of its main aims is to improve mobile multimedia services. This is achieved through streamlining the system for packet services. This leads to improvements in the form of higher bit rates, lower latencies, and a variety of service offerings. However, more challenging technical difficulties may be expected to arise when voice traffic flows over a long term evolution network. There is a major change in the way voice is transmitted in long term evolution network, in that it is transmitted in packets instead of through circuits. The fact that long term evolution is designed to support mobility, it is of great importance to analyse the effect of mobility to our new scheduling scheme for voice over internet protocol in long term evolution systems. In this paper, we analyse the effect of mobility to our proposed scheduling algorithm, voice over internet protocol optimisation scheduling algorithm while taking into account the quality of service parameters of voice traffic in long term evolution network. Using long term evolution-SIM simulation software, we evaluate the performance of our proposed scheduling algorithm and compare it with other scheduling algorithms in the literature such as; exponential proportional fair and proportional fair scheduling algorithms. Simulation results showed that mobility had a significant impact on the quality of service for voice traffic on all the three scheduling algorithms. Our proposed algorithm provided the best quality of service for voice traffic compared to the other two scheduling algorithms based on the packet loss ratio, delay, and throughput metrics.**

Keywords- LTE; Mobility; Scheduling Schemes; VoIP.

I. INTRODUCTION

3G long term evolution **(**LTE) was identified by the third generation partnership project (3GPP) as the preliminary version of next generation wireless communication systems because of its high data rates [1]. This mobile cellular communications technology provides a maximum 100Mbps downlink and 50Mbps uplink when using 20 MHz bandwidth [2]. In the downlink physical layer, LTE uses Orthogonal Frequency-Division Multiple Access (OFDMA) radio technology to meet the LTE requirements for spectrum flexibility and enables cost-efficient solutions for wide carriers with high peak rates. In the uplink, LTE uses a pre-coded version of OFDMA known as Single-Carrier Frequency-Division Multiple Access (SCFDMA) in order to compensate for a drawback with normal OFDMA of a high Peak-to-Average-Power Ratio (PAPR) [3]. Wireless technology has expanded from voice only to high-speed data, multimedia applications, and wireless internet [4].

LTE requirements for high data rates are achieved by the fact that this technology is only designed for packet switched networks (PSN); hence, there is no need for the circuit switched mode. However, this design brings with it more technical challenges especially for voice services. Voice over internet protocol **(**VoIP) services are both delay and packet loss sensitive. The biggest challenge of VoIP over LTE is to deliver Quality of Service (QoS). Normally, users would expect voice with the same quality as that provided by circuit switched networks. However, traffic delivered over PSNs is subject to delay and packet loss [5]. A major issue with VoIP over LTE is that 3G LTE adopts a different method of resource transmission from other cellular systems like Code Division Multiple Access (CDMA).

3G LTE uses Physical Resource Blocks (PRB) as its transmission unit. PRBs can be defined as the basic unit with both frequency and time aspects [6]. Basically, the base station of 3G LTE, known as eNodeB has a fixed number of available PRBs according to their allocated bandwidth and it is supposed to assign PRBs repeatedly at every Transmission Time Interval (TTI) [2]. Another issue with VoIP over LTE is that LTE systems also tend to support a very high mobility of up to 350 km/h [7]. LTE aims at providing a fast and seamless handover from one cell (source cell) to another (target cell) [8]. However, the consequences of the handover procedures in LTE systems depends entirely on the type of application that is being used, for example some applications would tolerent a short interuption while others would not. In this paper, the application that is being used is VoIP, so it is crucial to evaluate the QoS of VoIP for the high mobility. Different techniques have been introduced in recent years in order to overcome the challenges of voice over LTE [2] [1] [5]. However, QoS for voice over LTE is still a big challenge taking into account the mobility features of LTE, fading channels of wireless links as well as delay and packet loss sensitive voice characteristics. Our contributions in this paper are:

- Analyse the effect of mobility to our proposed scheduling algorithm; VoIP optimisation scheduling algorithm **(**VOSA) while taking into account the QoS parameters of voice traffic in LTE
- Evaluate the performance of our proposed scheduling algorithm VOSA with mobility features and compare it with other scheduling algorithms developed in [9] such as: exponential proportional fair **(**EXP-PF) and proportional fair (PF) scheduling algorithms.

The simulation results were generated using the open source LTE system simulator called long term evolution-SIM (LTE-SIM) [9]. It models different uplink and downlink scheduling strategies in multicell/multiuser environments; taking into account user mobility, radio resource optimization, frequency reuse techniques, the adaptive modulation, and coding (AMC) module. It also includes other aspects that are relevant to the industrial and scientific communities.

The rest of the paper is organised as follows: Section II discusses the general aspects of VoIP. Section III describes different scheduling algorithms used in this paper. Section IV describes the system model, scenario setup, handovers and mobility patterns. Section IV presents the simulation results and performance evaluation. Section V reviews the main conclusions, and introduces the future work.

II. VoIP

A. *Brief Description*

VoIP is a way of transmitting voice traffic as data packets over an IP network. Voice traffic is first transformed into digital signals then it is compressed and broken into a series of packets. These series of packets will later be reassembled and decoded at the receiver. Voice digitizing and encoding can either be done before or concurrently with packetization [10]. This technology has grown rapidly due to different factors such as: low cost, the integration of voice and data traffic over the existing networking infrastructures, etc.

VoIP is transmitted over a packet-switched network rather than the circuit-switch network protocols of the PSTN [11]. Initially voice and data were transmitted using two different networks but with the introduction of VoIP technology, they can both be transmitted using the same network infrastructure. VoIP reduces costs by avoiding the use of traditional PSTN. With VoIP technology the high cost for long distance calls and international calls transported over the circuit switched network can be reduced by transporting voice calls over low cost flat pricing packet switched network [12].

The advantages of VoIP such as integrated services and flexibility has attracted more customers as well as companies from circuit switch networks to packet switch networks. The fact that VoIP uses a packet-switched network means that the QoS provided by VoIP is not as good as that of circuit-switched network. This is due to the fact that real-time traffic such as voice is affected by technical issues like end-to-end delay or latency, jitter, and packet loss, hence adversely affecting the quality of voice [13]. Unique treatment should be given to voice traffic as it is vital for it to reach the destination in the quickest time [14].

Since our main aim is to analyse the effect of mobility to the QoS of voice traffic, it is of great importance to analyse some important parameters that describe QoS of voice in LTE networks. These parameters will be investigated in the next sub section.

B. *VoIP QoS Analysis*

The adaptive multirate (AMR) voice codec is one of the most popular voice codecs used in LTE. This codec provides 32-bytes voice payload every 20 milliseconds during talk-spurt period and 7-bytes payload carries a silence descriptor (SID) frame every 160 millisecond [1]. VoIP protocol stack that utilizes the real transport protocol (RTP) is encapsulated using user datagram protocol (UDP), and in turn is carried by IP. The combination of all these protocols requires a 40 byte IPv4 header or a 60-byte IPv6 header, but the overhead brought about by these headers causes serious degrading in spectral efficiency in supporting VoIP services. To solve this problem, an efficient and robust header compression (ROHC) technique is used. This technique solves the overhead problem by minimizing the size of the IP/UDP/RTP headers as little as 2 or 4 bytes using IETF RFC 3059 [6] [15].

One of the main characteristics of voice traffic is block/packet error rate that leads to packet loss and delay [7]. According to [16], the allowed maximum mouth-to-ear delay for voice is 250ms with the assumption that the delay for the core network is approximately 100ms, while the tolerable delay for radio link control (RLC), MAC buffering, scheduling, and detection should be strictly lower than 150ms as shown in Fig.1. Hence, taking into account that both end users are LTE users, tolerable delay for buffering and scheduling must be lower than 80ms. A delay of 50ms from eNB to UE has been chosen for the 3GPP performance evaluation metric limit to better account for variability in network end-to-end delays [7]. When voice packets are transmitted over a packet switched network, packets will be dropped due to error rate and packet delay exceeding the target latency. However with the occurance of the packet loss, voice quality is not affected if the error rate is less than outage threshold [7]. This means that the Qos of VoIP in LTE is limited by an outage limit, described in TR 25.814 [6] and was later updated in R1-070674 [17]. The outage limit means that error rate of VoIP users must be kept within 2%. The overall description of the QoS for voice users in LTE can then be defined as the maximum number of VoIP users that can be supported without exceeding a given threshold. At least 95% of total VoIP users should meet the above described outage limits [1].

III. SCHEDULING ALGORITHMS

Different scheduling algorithms have been introduced in recent years in order to overcome the challenges of voice over LTE.

In [1], the authors proposed an efficient LTE scheduler to increase the capacity of VoIP in E-UTRA Uplink. The proposed scheme modified the persistent scheduling algorithm proposed in [5] such that the resources of two VoIP users can be coupled, this brought about early termination gains without the need of additional control signals. The proposed efficient scheduling method employs a resource sharing approach. It also employs the random user pairing and best user pairing method to improve the capacity of VoIP services over E-UTRA Uplink. The results showed that the employment of their proposed scheduling scheme makes a larger available capacity than that resulting from the original persistent scheduling.

In [2], a Medium Access Control (MAC) layer PRB scheduling algorithm was proposed. The key ideas of this scheme are VoIP priority mode and its adaptive duration management. The VoIP priority mode assigns PRBs first to VoIP calls and it is also able to minimize VoIP packet delay and packet loss while the adaptive duration management is able to prevent the overall system performance degradation. The proposed MAC scheduler in this paper allocates PRBs in a round robin way and the scheduling order is determined according to the following factors; the queue length and Signal to Interference Noise Ratio (SINR) of each call, so the larger the factor values are the earlier the corresponding call is scheduled. Their results show that when the VoIP priority mode is not used, the packet drop rate rises rapidly as the number of VoIP call increases. On the contrary, when using the VoIP priority mode, the drop rate remains at low level around 1 % in spite of the increase of VoIP calls. However, this proposed mode has got a possible negative effect. It might degrade the efficiency of the eNodeB resource utilization because VoIP calls are seldom able to fully utilize the allocated PRB capacity.

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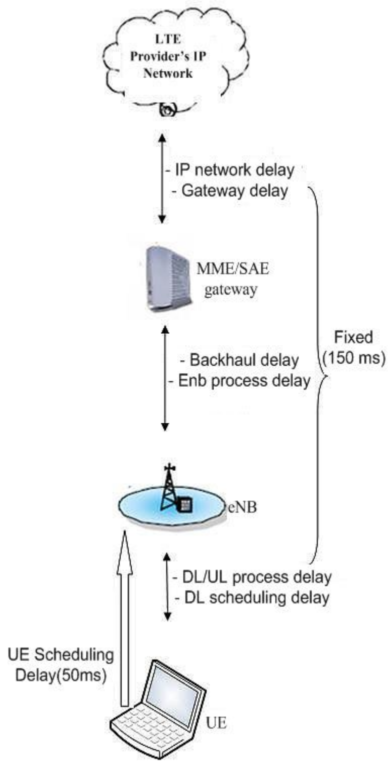

Figure 1. End To End Delay Components in E-UTRAN (LTE)

In [5], a new semi-persistent scheme of MAC scheduling, which adaptively prioritize VoIP traffic by not forcing other traffics to starve, was proposed. The proposed scheme, which combines VoIP priority mode with user coupling, allows utilizing the system capacity efficiently. The priority mode duration is adaptively controlled using the channel condition and two users are coupled to share the resources. The controlling user is also determined dynamically using minimum information as possible. The priority mode works by assigning resources to VoIP traffic in a priority basis at the same time controlling the duration of this mode dynamically according to the channel conditions in order to avoid starvation of other services in the same time. The scheme also allows user coupling where two users share the resources allocated to them by the eNodeB therefore offsetting the low resource utilization while in priority mode.

This is achieved by allowing two users who have different channel conditions to share resources. Basically, the proposed scheme consists of two parts; user coupling and link authority change. User pairing method takes care of pairing VoIP users according to the channel conditions so that the pairing results in the most efficient usage of resource. The link authority change ensures that the user in need of resource at any point of time has the authority on the link to get the fair share. The authority adaption phase is executed using the Acknowledgement / No acknowledgement (ACK/NACK) channel of the users. Each user monitors his couple's ACK/NACK channel and changes the authority status if the signal on his own channel and his pair's channel is different, i.e., 'his signal is ACK and his pair's signal is NACK or vice versa'. On the other hand, the authority remains same if the signals of both the channels are same. Their results show a great improvement over the original persistent mode.

In [9], different scheduling algorithms were developed and the main ones included; PF and EXP-PF. PF scheduling algorithm focused on maximizing the total network throughput as well as assuring fairness among flows. It allocates resources based on two factors; experienced channel quality and the past user throughput [19]. This scheduler uses the metric descrbed as the ratio between the instantaneous available data rate and the average past rate with reference to the i-th flow in the j-th flow subchannel. This can be depicted in equation 1 below.

$$W_{i,j} = \frac{r_{i,j}}{R_{i,j}} \tag{1}$$

where W_{ij} is the scheduler metric, R_i is the estimated average data rate and r_{ij} is the instantaneous available data rate.

EXP-PF scheduling algorithm basically aimed at increasing the priority of real-time flows as opposed to non-real-time flows. In other words, the flows with head-of-line packet delay very close to the delay threshold [20]. Its metrics were calculated as follows;

$$W_{i,j} = \exp\left(\frac{\alpha_i D_{HOL,i} - X}{1+\sqrt{X}}\right)\frac{r_{i,j}}{R_{i,j}} \tag{2}$$

and

$$X = \frac{1}{N_{r,t}} \sum_{i=1}^{N_{r,t}} \alpha_i D_{HOL,i} \tag{3}$$

with Nr,t being the number of active downlink real-time flow.

Considering a packet delay threshold T_i, the probability α_i is defined as the maximum probability that the delay $D_{HOL,I}$ of the head-of-line packet delay exceeds the delay threshold. Therefore α_i is given by

$$\alpha_i = -\frac{\log \alpha_i}{T_i} \tag{4}$$

With all these techniques in the literature, QoS for voice over LTE is still a big challenge taking into account the fading channels of wireless links as well as delay and packet loss sensitive voice characteristics. Based on the algorithm in [2], we propose a new scheduling algorithm called VOSA, with the aim of improving the performance of voice traffic over a 3G LTE network.

At the same time it reduces the negative impact that may be caused by the introduction of the new algorithm on the entire system's performance. Details of VOSA can be found in [18]. This algorithm is activated at every TTI by considering if there is a VoIP call and if the duration period of the new algorithm has not exceeded the limit. To determine the duration of our new algorithm, we use the adaptive method proposed in [2]. This method provides limits to *VOSA* by adaptively changing between a pre-specific minimum and maximum according to the ratio of dropped packets. Higher drop ratio means that there are many ongoing VoIP calls, and hence it is necessary to increase the limits to allow more consecutive TTIs to be dedicated to VoIP calls. On the other hand, low drop ratio implies that QoS of VoIP calls are satisfied at decent levels, and thus it is safe to reduce the duration of the algorithm and serve other service in the network. It should be noted that the adaptive method used here considers only dropped packets due to many ongoing VoIP calls (Call congestions). However, packet loss can also be due to different factors such as fading channels, interferences, etc and these factors were put under consideration while scheduling other kinds of traffic in our network.

Our scheduling scheme is designed by making modification to the algorithm in [2]. Basically, the *VOSA* allocates PRBs to VoIP calls based on the arrival time metric. Once the PRBs allocation is done, the scheduling order of the calls is determined by the following factors: Quality feedback (QF) and queue length (QL) of each call. The better the factor values are, the earlier the corresponding call is scheduled. In our algorithm we use the following equation:

$$D_f(i) = Q_{feedback(i)} * Q_{length(i)} \tag{5}$$

where $Q_{feedback\ (i)}$ and $Q_{length\ (i)}$ are the quality feedback and queue length respectively. Equation (5) implies that the better the wireless link and the longer the queue length, the earlier the corresponding call is scheduled to have the PRBs. This is calculated at every TTI. The details of the proposed algorithm consist of two parts. The first part of *VOSA* describes the PRBs allocation at every TTI. The second part is the adaptive method to control the duration of the proposed algorithm, detailed in [2].

In short, scheduling starts at every TTI. The activation of the scheduling algorithm is determined by whether there exists a VoIP call or not, and whether the count of the consecutive scheduling algorithm enabled TTIs does not exceed the limit. Otherwise the normal mode is set to schedule non-real time traffic based on similar factors as in equation (5) and the count is reset. VoIP calls are assigned one PRB at a time. Calls with long queues and better wireless link are served first. It continues in the same routine as long as there are remaining PRBs and there are calls having data to send. If there are remaining PRBs after VoIP call scheduling, the remaining PRBs are allocated to other services in the same way as the normal mode. This prevents PRBs from being wasted.

VOSA scheduling algorithm is summarised in 8 different steps;

1. Indentify the traffic type whether voice or any other traffic
2. Determine the user metrics (QF,QL)
3. Find the user with the highest user metric as defined in equation 5
4. Consider the set of available resource blocks RBs N_{avail_RB}, at every start of the algorithm , $N_{avail\_RB} = \{1,2,\ldots\ldots\ldots, N_{RB}\}$ to be allocated to number of user K
5. Assign the resource block N* to the user K* with the highest user metrics value such that $N_{RB,}K^* = N_{RB,}K^* \cup \{N^*\}$
6. Schedule the user K* first
7. Delete the user K* and resource block N* from their respective lists
8. Repeat all the steps until all users are scheduled and if more resource block exists then allocate them to other traffic types.

The fact that VOSA focuses on channel quality and queue length metrics at physical and MAC layers improves the QoS of Voice calls in LTE. Using equation (5), VOSA performs scheduling at the MAC layer and schedules Voice calls first before assigning PRBs to other traffic in the network. However, the starvation of other traffics in the network is controlled by the adaptive method used. In other words, VOSA is only deployed when there are voice calls to schedule and if it does not exceed its limits.

IV. SYSTEM MODELLING AND SCENARIO SETUP

A. *PRB Characteristics*

In this sub-section, we introduce the characteristics of PRBs, described as the transmission resources in LTE. LTE systems consists of both a time and a frequecy plane. The time plane is divided into 1 ms TTI which consists of two slots of 0.5 ms to form 1 ms sub frames, where each sub frame contains 7 OFDMA symbols. In each TTI, there are 14 OFDMA sysmbols, where 2 symbols out of 14 are reserved for uplink pilot transmission, while the other 12 symbols are used for data and control information transimission. TTI can be defined as the minimum allocation unit in the time domain [21].

If we consider the frequency plane, the minimum allocation unit is the PRB, where each PRB contains 12 subcarriers of 15 Khz bandwidth each. The number of OFDMA symbols in a resource block depends on a cyclic prefix being used. All these can be depicted in Fig. 2. It must be noted that VoIP packets must be transmitted per TTI and they can occupy one or more PRBs [5]. The amount of data bits that can be transmitted by one PRB depends on the link between the eNodeB and the user mobile terminal. This is due to the fact that 3G LTE uses adaptive modulation and coding (AMC), in order to change modulation and coding schemes depending on the wireless link conditions.

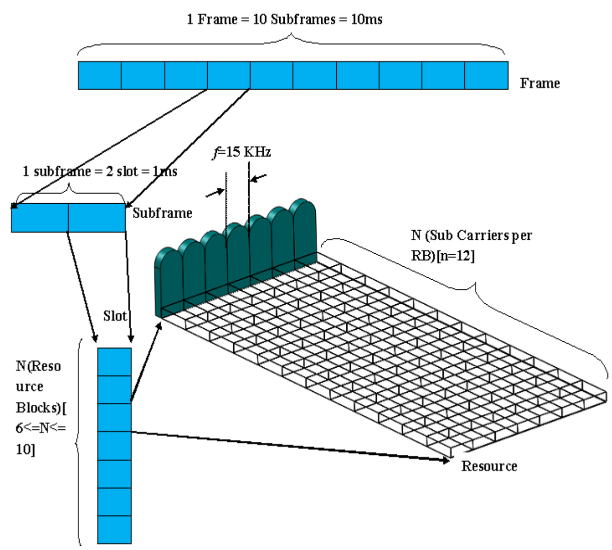

Figure 2. The structure and allocation of the eNodeB transmission resources symbols.

B. *Scenario Setup*

The network is made up numerous cells and different network nodes such as; the EnodeB, mobility management/gateway (MME/GW) and user equipements (UEs). All the simulations were run in a diamond-pattern scenario with 19-3-sector sites that totaled to about 57 cells. Most of the simulation parameters are presented in the table 1 below. VoIP flows are generated by the traffic generator in LTE-SIM called VoIP application. This application generates G.729 voice flows.VoIP traffic was designed with an ON/OFF Markov chain. The ON period has an exponentially distributed mean value of 3s while the OFF period has a shortened exponential probability density function with an upper limit of 6.9s as well as an average value of 3s [22]. Throughout the ON period, the source sends packets of 20bytes every 20 ms, thus implying that the source data rate is 8 kb/s. On the other hand, throughout the OFF period the rate is zero as we assume the presence of voice activity detector.

Three different scheduling algorithms were used in all simulation scenarios, these are: our proposed VOSA as well as EXP-PF and PF developed in [9]. In one simulation scenario there was no mobility features implemented so the user speed was set to zero, this is equivalent to static position. In another simulation scenario, we implemented mobility features and the user speed was set to 30 km/h which was equivalent to vehicular and the distance covered was set to 400 meters by default.

To quantitatively investigate the effect of mobility to the voice traffic including each scheduling scheme, we measured three important VoIP QoS metrics (Packet-Loss-ratio, Delay, and throughput) for all the scheduling algorithms and in both scenarios.

C. Handover and Mobility Pattern

One of the main aims of LTE systems is to provide a fast and seamless handover from one cell (source cell) to another (target cell) [8]. This is generally achieved due to the distributed nature of LTE radio access network architecture that consists of one node known as the EnodeB. According to the 3GPP release 8 specifications, handovers in LTE are hard handovers, implying that there is a minimum interruption in the services when the handover is performed.

During the handover process, VoIP users cannot be scheduled, they cannot transmit or receive any data. The only transmitted and received data is the signalling related to the handover procedure [7]. This can lead to additional delays to Packet Data Units (PDUs) hence affecting the user quality on each handover process. LTE utilises a UE assisted hard handover algorithm for mobility. The UE measures the downlink signal quality and sends the the measurement reports to the EnodeB either periodically or when an event triggers i.e., 'an interfering EnodeB becomes stronger than the current serving EnodeB'. Then the EnodeB will decide on the final handover based on the received measurement report. Normally, measurement averaging, handover margins, and timers are used to avoid excess handovers [23].

In the LTE-SIM simulator, two types of mobility models were developed, known as; random direction and random walk [24]. The user speed was selected between 0, 3, 30, 120 km/h, which were corresponding to static, pedestrian, and vehicular scenarios respectively. The user speed was mapped to a specific travel distance, i.e., 'a user travelling at 3km/h would cover 200 meters, a user travelling at 30km/h would cover 400 meters and the user travelling at 120km/h would cover 1000meters'.

If the random direction model is being used, the user chooses the speed direction at ramdom and keeps the same speed while moving towards the simulation boundary area. When the user reaches the simulation boundary area then new speed direction can be choosen. In contrary, when the random walk model is chosen, user chooses the speed direction at random but keeps moving at that speed for a specific travel distance depending on that speed. The user only changes the speed direction if the distance is convered or once the simulation boundary is reached [9].

V. SIMULATIONS AND PERFORMANCE EVALUATION

We used LTE-SIM to analyse the effect of mobility on our proposed scheduling algorithm (VOSA) and two other scheduling EXP-PF and PF. It should be noted that the details of VOSA can be found in [18]. The main contribution in this paper that was not introduced in [18] is mobility. We introduced mobility in our network and analysed it's effect on voice traffic in relation to the three scheduling algorithms in this paper.

The main reason for comparing our proposed scheduling algorithm with these other two scheduling algorithms is that, they use the same PRBs allocation as ours and similar simulation parameters except that they apply different metrics and the fairness factor. They were also used as the benchmark scheduling algorithms in the LTE-SIM simulator. This made our comparison more feasible.We measured the packet loss ratio (the rate at which VoIP packets were dropped during voice traffic transmission) while gradually increasing the number of VoIP users.

This is shown in Fig. 3. The packet drop ratio is measured and plotted on the Y axis as we increased the number of VoIP user steadily to the maximum of twenty users. As it can be seen in Fig.3, mobility had a significant impact on voice traffic in all three scheduling algorithms. There was a higher packet loss ratio in all the algorithms with mobility features compared to those without mobility features. However, there was less packet loss ratio in VOSA scheduling algorithm (with or without mobility features) compared to the other two scheduling algorithms.

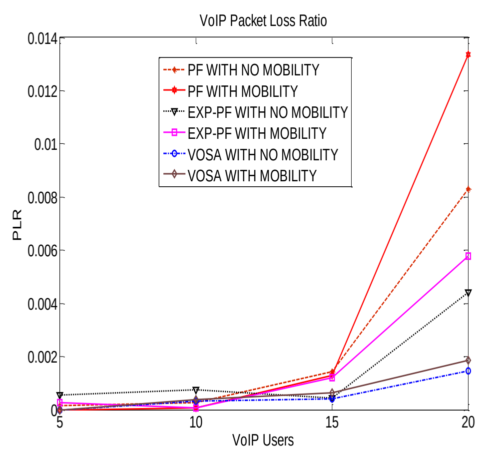

Figure 3. VoIP-Packet-Loss-Ratio Comparison

We also measured VoIP delay while gradually increasing the number of VoIP users. This is shown in Fig. 4. The VoIP delay is measured and plotted on the Y axis in seconds as we increased the number of users steadily to twenty. Similar to previous results, there was long delay in all the algorithms with mobility features and again VOSA had less delay than the other two scheduling algorithms. These two simulation results show that VOSA plays an important role in improving the QoS of Voice traffic in both scenarios.

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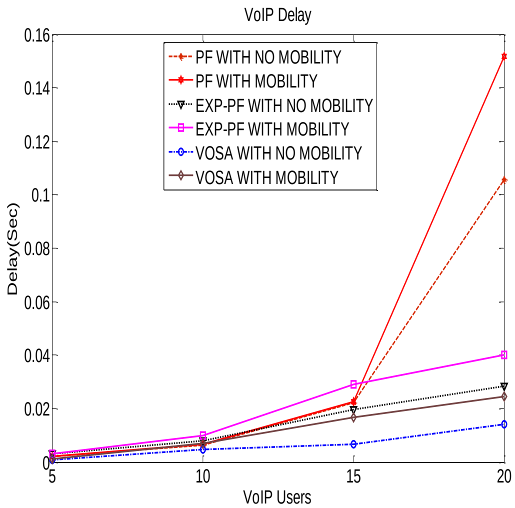

Figure 4. VoIP Delay Comparison

TABLE 1. SIMULATION PARAMETERS

Simulation Parameters	**Values**
Bandwidth	5MHZ
PRB Structure	12subcarriers,2subframes
TTI	1msec
Number of available PRBs	25
Modulations for AMC	QPSK
Number of sectors	3
Simulation time	1000 TTIs
User speeds	30km/h
Cyclic prefix	Normal
Mobility patterns	Random direction and random walk
Distance covered by user	400m
Scheduling algorithms	VOSA,EXP-PF, and PF
Cell radius	1 km

Apart from these two VoIP metrics, we also measured throughput while using all the scheduling algorithms and in both scenarios. This is shown in Fig. 5. As it can be seen, throughput decreased as the number of VoIP users increased in all algorithms and in both scenarios. This is mainly due to the fact that some VoIP packets were being dropped as the number of users were being increased, this resulted in the less utilisation of assigned PRBs.

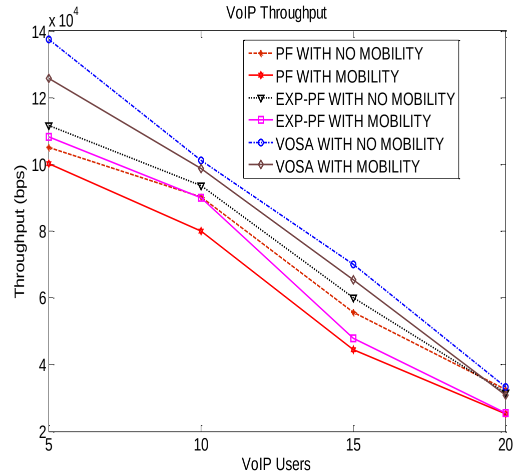

Figure 5. VoIP Throughput Comparison

VI. CONCLUSION AND FUTURE WORK

In this paper, we analysed the effect of mobility on voice traffic with three different scheduling algorithms. Using the LTE-SIM simulator, we were able to compare all the scheduling algorithms in two different scenarios. One scenario containing mobility features and another with no mobility features. Through simulations we found out that mobility had a significant impact on the QoS of voice traffic with all the three scheduling algorithms. However, our VOSA scheduling method provided better QoS of voice traffic than the other two scheduling algorithms by having short delay, less packet loss ratio, and slightly higher throughput. The main reason for the improvement in VOSA scheduling method is due to the fact that it does not cause any negative impact on the entire network's system performance. It is only deployed when there are VoIP calls to schedule and if it does not exceed its limits. These limits are determined by the adaptive method used. On top of that, the wireless link quality and the queue length factors used in VOSA played an important role in reducing the packet drop ratio and delay. In future work, we are working on determining the complexity and fairness of our proposed scheduling algorithm VOSA. Find the way of optimizing our scheduling algorithm such that we lower its complexity.

REFERENCES

[1] S. Y. Kim, “An Efficient Scheduling Scheme to Enhance the Capacity of VoIP Services in Evolved UTRA Uplink,” EURASIP journal of Wireless Communications and Networking, vol 2008, Mar. 2008, pp. 1-9, doi:10.1155/2008/732418.

[2] S. Choi, K. Jun, Y. Shin, S. Kang, and B. Choi, “Mac Scheduling Scheme for VoIP Traffic Services in 3G LTE,” IEEE 66th Vehicular Technology Conference, pp. 1441-1445, Oct. 2007, doi: 10.1109/VETECF.2007.307

[3] Rohde and Schwarz, "*UMTS* long term evolution (LTE) Technology introduction," A report by Rohde and Schwarz., pp. 1-30, Mar.2007.

[4] M. C. Chuah, and Q. Zhang, Introduction to Wireless Communications, US: Springer, 2006.

[5] S. Saha and R. Quazi, "Priority-Coupling-A Semi-Persistent MAC Scheduling Scheme for VoIP Traffic on 3G LTE," ConTEL, 10th International Conference on Telecommunications, pp. 325-329, Aug.2009.

[6] 3GPP, Physical layer aspects for Evolved UTRA, 3GPP Technical report 25.814, version7.1.0, pp. 1-135, Sep. 2006.

[7] T. Henttonen, K. Aschan, J. Puttonen, N. Kolehmainen, P. Kela, M. Moisio, et al, "Performance of VoIP with Mobility in UTRA Long Term Evolution," Vehicular Technology Conference, pp. 1-5, May 2008, doi: 10.1109/VETECS.2008.549.

[8] Qualcomm, LTE Mobility Enhancements, Qualcomm Incorporated, pp. 1-10, Feb.2010.

[9] G. Piro, L. A. Grieco, G. Boggia, F. Capozzi, and P. Camarda, "Simulating LTE Cellular Systems: An Open-Source Framework," IEEE Transactions on Vehicular Technology, vol.60, no.2, pp. 1-16, Feb.2011.

[10] B.Goode, "Voive over internet protocol (VoIP)," Proc.IEEE, vol.90, no.9, Dec.2002, pp. 1495- 1517, doi:10.1109/JPROC.2002.802005.

[11] K. Gonia, Latency and QoS for Voice over IP, SANS Institute- As part of GIAC practical repository, pp. 1-21, May. 2004.

[12] G. Zhang, and M. Hillenbrand, "Implementing SIP and H.323 signalling as Webservices," 30th Euromico conference proceedings, pp. 1-7, Mar.2004.

[13] B. Khasnabish, Implementing Voice over IP. John Wiley & sons Inc, pp. 1-224, 2003.

[14] U. Varshney, A. Snow, M. McGivern, and C. Howard, "Voice over IP," ACM Journal, Jun. 2002, pp.1 –8.

[15] IETF, Attribute list extension for the service location protocol,. IETF RFC 3059, pp. 1-35, Nov.2001.

[16] ITU-T, One way transmission time. ITU-T recommendation G.114, pp. 1-20, April.2003.

[17] TSG-RAN1, LTE physical layer framework for performance verification, 3GPP TSG-RAN1 R1-070674, pp.1-30, Mar. 2007.

[18] R. Musabe, H. Larijani, B. Stewart, and T. Boutaleb, "A New Scheduling Scheme for Voice Awareness in 3G LTE," IEEE Computer Society, Sixth International Conference on Broadband and Wireless Computing, Communication and Applications, Dec.2011, pp. 1-8, doi: 10.1109/BWCCA.2011.46.

[19] G. J. Choi and S. Bahk, "Cell-throughput Analysis of the proportional fair scheduler in the single-cell environment," IEEE Transaction Vehicular Technology, vol.56, no.2, April 2007, pp. 766-778, doi: 10.1109/TVT.2006.889570.

[20] R. Basukala, H. M. Ramli, and K. Sandrasegaran, "Performance analysis of EXP/PF and M-LWDF in downlink 3GPP LTE systems," 1st AH-ICI . Kathmandu, Nepal, Nov.2009, pp.1-5, doi: 10.1109/AHICI.2009.5340336.

[21] Roke, LTE eNodeB MAC Schedular, Roke Manor Research Ltd, Siemens, pp.1-4, Jan. 2009.

[22] C. Chuah, and R. H. Katz, "Characterizing packet audio streams from internet multimedia applications," Proc.IEEE, ICC, Aug. 2002, pp. 1199-1203, doi: 10.1109/ICC.2002.997040.

[23] J. Puttonen, T. Henttonen, N. Kolehmainen, K. Aschan, M. Moisio, and P. Kela, "Voice-over-IP Performance in UTRA Long Term Evolution Downlink," Vehicular Technology Conference, pp. 2502- 2506, May 2008, doi: 10.1109/VETECS.2008.551.

[24] T. Camp, J. Boleng, and V. Davies, "A Survey of Mobility Models for Ad Hoc Network Research," Wireless Communication and Mobile Computing (WCMC), Special issue on Mobile Ad Hoc Networking, pp. 483-502, Dec. 2002.