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Scheduling for Multi-users Multiplexing Radio Voice Transmission for Enhancing Voice Capacity over LTE in PMR Context

Manh-Cuong Nguyen¹, Hang Nguyen¹, Eric Georgeaux², Philippe Mege², Laurent Martinod² and Alina Florea¹

Abstract—Despite the fact that Long Term Evolution (LTE) supports high-speed data transmission and supports different packet sizes by using Adaptive Modulation and Coding (AMC), LTE is not yet optimized for low bit rate voice communication, especially in case of using LTE in Professional Mobile Radio (PMR) context. In LTE, one pair of Physical Resource Blocks (PRBs) is the smallest User Assignment Unit. However, the smallest LTE packet size is still too large in case that low bit rate voice communication is transmitted in high Modulation and Coding Scheme (MCS). This reduces the voice capacity over LTE in PMR context. Therefore, we propose a new Multi-users Multiplexing Radio Voice Transmission method for enhancing voice capacity over LTE in PMR context. In this method, voice packets from different users are multiplexed into one LTE packet in the downlink transmission. In this paper, we propose a new scheduling method that can be used to transmit the multiplexed packets to corresponding users. This paper gives also a novel algorithm for classifying and grouping voice packets of mobile users having different MCS values. In case of low and medium number of users, the results show that the average voice capacity gain of the proposed method can rise up to 89.6%. In case of high number of users, the proposed method can improve the voice capacity by factor of 6.5.

Index Terms—scheduling, LTE, multiplexing, MCS, voice capacity, PMR

I. INTRODUCTION

Long Term Evolution (LTE) takes advantage of Orthogonal Frequency Division Multiple (OFDM) and Multiple Input Multiple Output (MIMO) to support high peak transmission rate (100 Mbps for the downlink (DL) and 50 Mbps for uplink (UL)). LTE becomes an evolution to the Fourth Generation (4G) broadband cellular communication [2] and a potential candidate for the next generation of Professional Mobile Radio (PMR), which is used for public safety work [4].

Despite the fact that LTE supports different packet sizes by using Adaptive Modulation and Coding (AMC), LTE is not yet optimized for small throughput; therefore, voice capacity in LTE is not increased as expected. There are several studies for increasing voice capacity over LTE. We can classify these studies into two main categories: reduction of data overhead and reduction of control overhead.

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The first category gives solutions for reducing the redundancy in the LTE packetization. In [3], the authors present a Robust Header Compression (ROHC) to compress the higher protocol headers (Internet Protocol (IP), User Datagram Protocol (UDP) and Real-time Transport Protocol (RTP)). With ROHC, the higher protocol headers can be compressed from 46 bytes for IPv4 and 66 bytes for IPv6 down to 1-3 bytes. The authors of [12] proposed a mechanism for reducing the Cyclic Redundancy Check (CRC) size from two to one byte without impacting the performance of system. [5] has presented a method for reducing the redundancy of the error correction part. In this article, different code rates of turbo code are used for bits having different protected levels.

The second category gives solutions to avoid the control channel limitations of fully dynamic (FD) scheduling. In semi-persistent scheduling (SPS) [6], the UE is pre-configured to retain certain scheduling information (resource block assignments, MCS ...) for a pre-defined period. During this period, a new allocation may be sent if the radio link conditions change. Therefore, the SPS can support more allocations with the same control resource. Transmission Time Interval (TTI) bundling [11], group scheduling [8] and packet bundling [10], can also be used efficiently to reduce the signal overhead. In TTI bundling, a few consecutive TTIs are bundled together and a single transport block is coded and transmitted by using a consecutive TTIs instead of being segmented in case of over-payload. TTI bundling reduces the L1/L2 grant and the number of HARQ feedbacks. In group scheduling, the mobile stations are clustered into group and the resource allocations are scheduled for each group instead of scheduling for each mobile station. In packet bundling two or more voice packets of one user are bundled into one L1 Protocol Data Unit (PDU).

However, existent methods pay attention mainly for the data overhead or signal overhead for high or medium voice payload. The gap between the LTE packet size and the small voice payload has less attention. In LTE, one pair of PRBs is the smallest User Assignment Resource Unit. The smallest LTE packet size is large when low bit rate voice communication is transmitted in high quality channel. In one general scenario, LTE uses 3MHz bandwidth, two transmission antennas, 3 symbols for Physical Downlink Control Channel (PDCCH), Advanced Multiband Excitation (AMBE) voice coder of 2450 bps, MCS index of 16 in PMR context, the size of smallest LTE packet is about 280 bits, three times bigger than the size of voice payload with size of 97 bits. This gap reduces significantly the

voice capacity of the system. In the PMR context, we found that if we could eliminate this gap, we can improve the voice capacity essentially. Therefore, we propose a new Multi-users Multiplexing Radio Voice Transmission method for enhancing voice capacity over LTE in PMR context. In this method, voice packets from different users are multiplexed into one same LTE packet. We built a new architecture for multiplexing voice packets from different users into one same LTE packet [9]. Our new Multi-users Multiplexing Radio Voice Transmission method allows reducing the difference between LTE packet size and voice payload to enhance voice capacity of LTE in PMR context. For multiplexing voice packets from different UEs we have to solve several issues. The first issue is how to send multiplexed packets to the corresponding UEs. The second issue is how to determine the number of voice packets that can be put in one LTE packet. The third issue is to find out the criteria for classifying and grouping the voice packets that can be put together in one same LTE packet in case that the UEs have different MCS values. In this paper, a new scheduling method to transmit the multiplexed packets to corresponding users is proposed. This paper presents also a new algorithm for classifying and grouping voice packets of mobile users that have different MCS values. The proposed scheduling method is realized by negligible modifications in the MAC scheduler. This new scheduling method needs not implement new elements in LTE architecture and no need to modify the structure of signal control in LTE. We introduce our new Multi-users Multiplexing Radio Voice Transmission scheduling and present a new algorithm for classifying and grouping voice packets of mobile users having different MCS values in section II. The performance of our method is evaluated in section III. Section IV will give the conclusion and some perspectives.

II. PROPOSED MULTI-USERS MULTIPLEXING RADIO VOICE TRANSMISSION METHOD

The general idea of this method is to put voice packets from different UEs into one same LTE packet. This allows reducing the different between the LTE packet size and the payload size so that we can reduce the average of resource blocks which are used to transmit voice payload of one UE. Assume that there are n UEs in a cell and they are classified and clustered into k group. Each group uses RG_j resource blocks to transmit the multiplexed payload of group. In this case, the proposed method have to find out a strategy for classifying and grouping to ensure that the number of used resource blocks of the proposed method $RG^{total} = \sum_{j=1}^k RG_j$ must be less than or equal to the number of used resource blocks in standard LTE $R^{total} = \sum_{i=1}^n R_i$ with R_i is the number of used resource blocks for UE_i in standard LTE.

A. Proposed Multi-users Multiplexing Radio Voice Transmission Scheduling

One of the most important issues to solve is how to transmit multiplexed packets to the corresponding UEs (see Fig 1). In LTE, the communication between UE and eNodeB is mainly

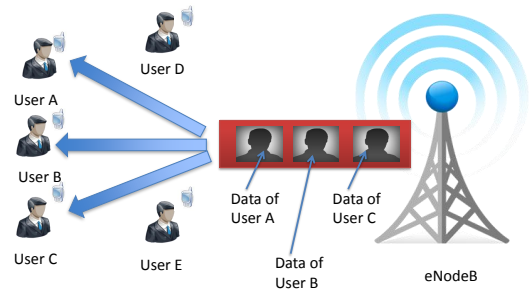


Fig. 1. Multicast issue in the proposed method

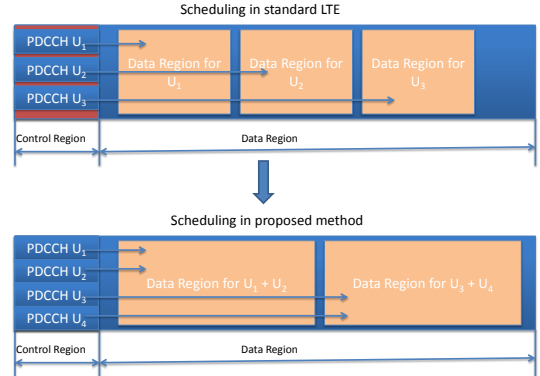


Fig. 2. Scheduling in the proposed method

unicast and broadcast. The unicast is used for carrying the user data or control information (resource allocation). Broadcast is used to send essential parameters for initial access such as downlink system bandwidth, system frame number ... In our case, we need a multicast mechanism. In this subsection, we will discuss the limitation of the evolved Multi-media Broadcast/Multicast Service for sending multiplexed packets and propose our new scheduling based solution.

1) *Limitation of evolved Multi-media Broadcast/Multicast Service (eMBMS)*: For transmission of the multiplexed packets to the corresponding UEs in the downlink, we need a multicast mechanism. In LTE, multicast is realized by concept of evolved Multi-media Broadcast/Multicast Service (eMBMS) [7]. However, eMBMS is used mainly for live video, audio streaming. Solution for voice packets is still a challenge. On the other hand, to apply eMBMS there are some new entities that must be added in the LTE infrastructure such as a Broadcast Multicast Service Center (BM-SC), eMBMS Gateway (MBMS GW), Multi-cell/Multicast Coordination Entity (MCE). Furthermore, in eMBMS all mobile users have the same content to receive. However, in our case, each UE has different content to receive and each UE must take out the corresponding part in LTE packet. Therefore, we proposed a new mechanism for multicast voice packet. This new multicast mechanism is realized by a new group scheduling method. In our scheduling method, we make modifications of standard scheduling in LTE to realize the multicast for multiplexed voice packets.

2) *Proposed Multi-users Multiplexing Radio Voice Transmission Scheduling for transmitting multiplexed packets:* In standard LTE, the resource allocation is realized at each TTI. The information of the allocated resource elements for UEs in the downlink is found in the PDCCH. The PDCCH uses the first 1, 2 or 3 OFDM symbols at the beginning of each sub-frame. The number of OFDM symbols in a sub-frame is indicated by the Physical Control Format Indicator Channel (PCFICH). Each PDCCH carries one Downlink Control Information (DCI) and the information about the identity of each UE by a Radio Network Temporary ID, or RNTI. The DCI carries the detailed information about allocated resource block for each user, the modulation and coding scheme and the other additional information such as HARQ process number, CQI request, and Transmit Power Control (TPC) command. DCI undergoes channel coding to form PDCCH. In standard LTE, each UE will search space that carries control information, finds out corresponding PDCCH by comparing its RNTI with RNTI in PDCCH [1]. Next, UEs read information in PDCCH to find out their allocated resources and modulation and coding scheme. In our case, the group classifier and group selector will classify and select voice UEs to send to MAC scheduler (see Fig 3). The criteria for classifying the UEs is MCS values of UEs. UEs have the same MCS values will be put in a same group. The group Selector will select UEs in a same group or groups with adjacent MCS values to create a list of UEs, choose MCS for group and send it to the MAC Scheduler. The details of algorithm for classifying and grouping UEs is presented in subsection II-B. In standard LTE, each PDCCH contains different resource allocation information for each UE. In our scheduling, MAC Scheduler will calculate resources that must be used and will assign the same resource allocation and the same MCS for all users in the group. n UEs in the multiplexed group will receive n different PDCCH channels having the same Resource Block Assignment (RBA) and MCS values (see Fig. 2). To do that, we define a new DCI format for the Multi-users Multiplexing Radio Voice Transmission. This format contains information about the Resource Block Assignment (RBA), MCS and position of UE in the group (see table I). The position field is used for the demultiplexing step on the receiver side.

TABLE I
DCI FORMAT FOR THE MULTI-USERS MULTIPLEXING RADIO VOICE TRANSMISSION

Field Name	Length (bits)	Comment
RBA	3 (1.4 Mhz)	Same for all UEs in the group
	5 (3 Mhz)	
	7 (5 Mhz)	
MCS	5	Same for all UEs in the group
Position in group	4 (1.4 Mhz)	Different for each UE
	6 (3 Mhz)	
	8 (5 Mhz)	

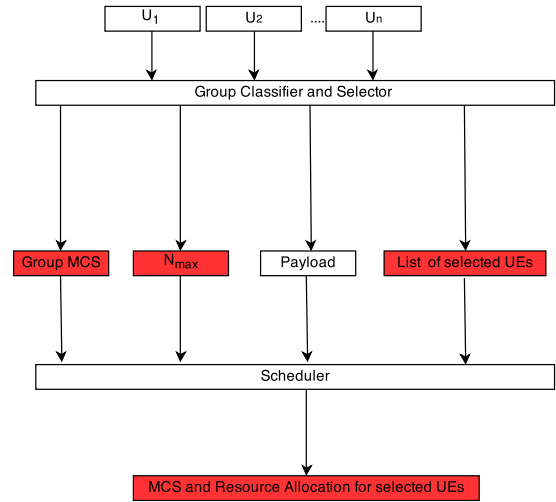


Fig. 3. Multi-users Multiplexing Radio Voice Transmission Scheduling

B. Different MCS values issue and algorithm for grouping UEs with different MCS values

In a cell, the channels usually have different CQI values. Therefore, UEs may be assigned different MCS values. Another issue to consider is how to determine the multiplexing size and the common MCS of group because this can affect the system performance and Bit Error Rate (BER) of UEs. To do so, it would be interesting to have a flexible algorithm that can cluster UEs having different MCS values. This subsection introduces the different MCS value issue and presents a new algorithm for grouping UEs having different MCS values.

1) *Different MCS values issue:* In LTE, the channel quality is evaluated by the channel quality indicator (CQI) value. CQI values are measured by the UEs and are sent to eNodeB. There are several factors that affect the CQI value. The CQI value depends on the noise and interference level of the channel and the quality of receiver. eNodeB receives the CQI values from the UEs and assigns the corresponding MCS values for UEs. For UEs that are assigned low MCS values, the number of RBs that must be used to transmit the same voice packet is greater than in the case of higher MCS values. Since with low MCS value, LTE needs more resources for detecting and correcting the errors.

One of the issues to be solved is how to group UEs that are assigned different MCS values. It means what MCS should be chosen as the common MCS of the group. This is the trade off between the BER of UEs and the system performance. If in a group the lowest MCS value is used, LTE will use more resource blocks for detecting and correcting the errors. This allows a better error protection, but the additional resource blocks will reduce the gain of the proposed method. On the contrary, if a higher MCS value is used as common MCS of the group, the error protection level will not be as good as expected for UEs that have low MCS value.

2) *Algorithm for grouping UEs with different MCS values:* To ensure the error protection and system performance, in this

paper, we propose a new algorithm for grouping and selecting UEs in case that UEs have different MCS values. The main idea is that UEs which have the same MCS value or adjacent MCS values will be selected for the same group. In each group, the lowest MCS value is chosen for calculating the group size. This value is considered the common MCS value for the same group. This can affect the voice capacity gain. However, in this case, the BER will be reduced because a higher error protection level is used. In addition, this algorithm not use the same size of group for all UEs. The size of group is changeable. It depends on the channel quality of each group. For UEs that have good channel quality (concentrate mostly in the center of the cell), this algorithm will assign a high value size of group. In contrast, for UEs with low channel quality, the size of the group is small (see Figure 4). This allows enhancing the system performance. Since for channels with high quality (high CQI values), LTE will assign high corresponding MCS values for UEs. In this case, the LTE packet size will be much larger than voice payload of UEs. Therefore, a bigger size of group will reduce the difference between the LTE packet size and the voice payload of group. In addition, this also increases the number of voice payload can be multiplexed in one same LTE packet to enhance the voice capacity gain. Algorithm for choosing UEs for scheduling is presented in Figure 5. In this algorithm, UEs will be sorted according to MCS values. The classifying and sorting UEs by MCS before implementing scheduling allows increasing efficiency of resource allocation because this reduces the dispersion of MCS in a group. In each group, the smallest MCS value will be selected for calculating the multiplexing size. The multiplexing size is a function of lowest MCS value of the group, voice payload and the LTE bandwidth. For a defined MCS, this algorithm uses the table 7.1.7.1-1 of 3GPP TS 36.213 V9.0.1 to find corresponding TBS index of MCS index. Next, this algorithm refers to the table 7.1.7.2.1-1 to find the maximum LTE corresponding packet size TBS_{max} for TBS index and the number of used resource blocks equal to the LTE bandwidth $N_{PRB}=N_{BW}$. The multiplexing size N_{max} is defined as the fraction between the maximum LTE corresponding packet size and the voice payload $TBS_{necessary}$.

$$N_{max} = \frac{TBS_{max}}{TBS_{necessary}} \quad (1)$$

To estimate the gain in this case, we first calculate the total number of resource blocks for all UEs in standard LTE. Assume that there are n UEs, UE_i needs RB_i resource blocks to transmit its voice payload. The total using resource blocks in standard LTE: $R^{total} = \sum_{i=1}^n R_i$

Assume that, we classify and group n UEs into k group. Each group uses RG_j resource blocks to transmit multiplexed payload of group. The total of resource blocks that are used for scheduling in our case: $RG^{total} = \sum_{j=1}^k RG_j$. In this case, the gain of the proposed method is defined as:

$$gain = \frac{(R^{total} - RG^{total}) * 100\%}{RG^{total}} \quad (2)$$

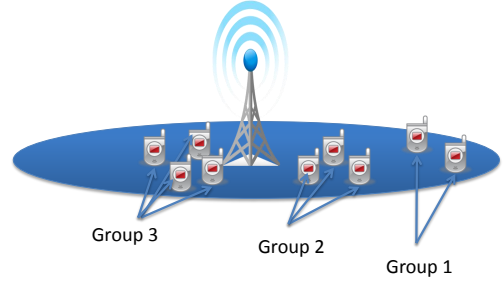


Fig. 4. Different MCS values issue and the general idea for grouping UEs

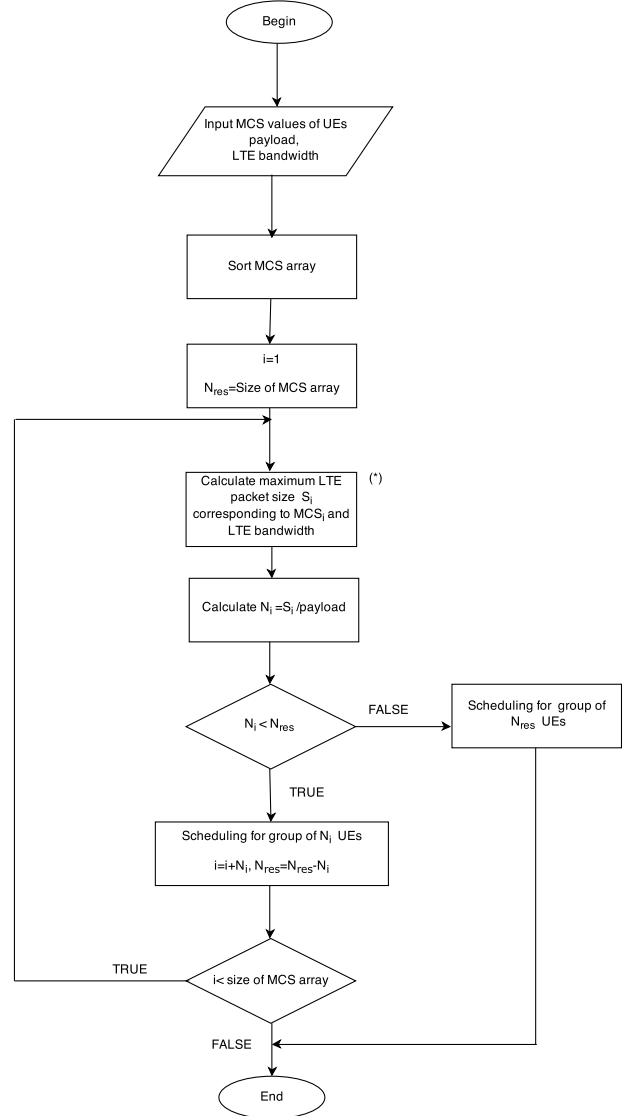


Fig. 5. Algorithm for classifying and selecting UEs for scheduling

III. PERFORMANCE EVALUATION

In our context, we study and evaluate LTE as a candidate for future PMR. In PMR, the voice communications are typically short. Voice capacity and robust channel protection

are important aspects of PMR. Therefore, the AMBE codec with low bit rate (2450 bps) and the low bandwidths of LTE (1.4 MHz, 3 MHz, 5 MHz), 24 bits CRC and Normal Cyclic Prefix are used for our evaluation. In the first scenario, we assume that number of UEs in a cell is high. In this case, we cluster only UEs having the same MCS values for evaluating the voice capacity gain of the proposed method for the corresponding MCS value. In the second scenario, UEs that have different MCS values can be grouped together. This scenario is used for evaluating the efficiency of the proposed method in case of low and medium number of UEs. In this scenario 50, 100, 200, 500, 1000 and 2000 UEs with random MCS values are used for 1000 simulation tests. We compare the average of used resource blocks in standard LTE and in our proposed method.

Table II, figure 7 and figure 8 present the efficiency of the proposed method for the second scenario. We found that the efficiency of the proposed algorithm depends on the number of UEs in one cell. The more UEs is in one cell, the higher voice capacity gain. The results show that when the number of UEs is low, our method works more effectively at low bandwidth and less effectively at high bandwidth. As the number of UEs is increased, our method is more efficient at high bandwidth. The reason is that our method uses the minimum MCS value to group the UEs to ensure the error protection and uses the biggest LTE packet for multiplexing the packets. Therefore, when the number of UEs is low, the dispersion of MCS values in a group is high when the packet size is large, this reduces the efficiency of resource utilization in our method. When the number of UEs is high, the dispersion of MCS values in a group is low, so with the bigger LTE packet size, the efficiency of resource utilization is enhanced.

Figure 6 present the voice capacity gain of our method in terms of number of users for 20 ms evaluation for the first scenario for 3 MHz and 5 MHz bandwidths. We can find that in the standard LTE, along with the increasing of MCS, the voice capacity is increased to its saturate MCS. In this figure, the saturate MCS is equal to 7 in 3 MHz and 5 MHz bandwidths. When the MCS is greater than the corresponding saturate MCS, the voice capacity of standard LTE cannot be enhanced. We also found that the graph of voice capacity in standard LTE has sections in which the voice capacity is not changed. However, the graph of voice capacity in our method is increased steadily with the increasing of MCS. This is because of the scheduling mechanism in LTE. In LTE, the smallest User Assignment Resource Unit is a pair of PRBs. When MCS is increased the LTE packet size is larger, but the size of voice payload is not changed so that one pair of PRBs can transmit maximum one voice payload from one UE. In addition, in standard LTE, it is difficult to choose a LTE packet size that is close to the size of voice payload. In our case, when the LTE packet size is expanded, we can transmit voice payloads from several UEs by a flexible multiplexing size therefore we can get higher voice capacity gain. In the best case (MCS value equal to 28 and LTE bandwidth equal 5 MHz), the proposed method can multiply the voice capacity

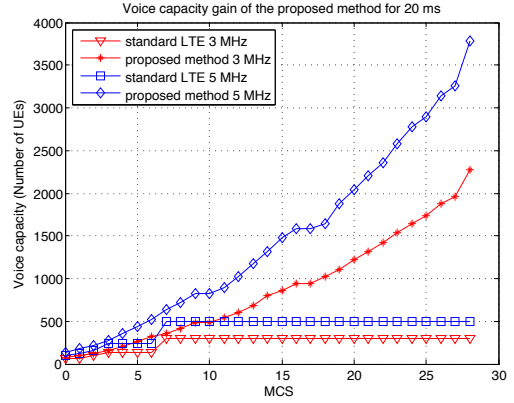


Fig. 6. Voice capacity gain of the proposed method for the first scenario

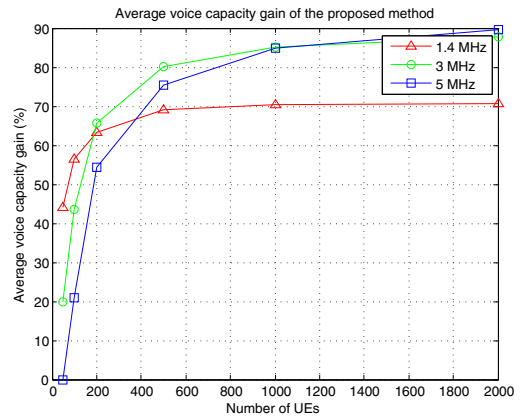


Fig. 7. Voice capacity gain of the proposed method for the second scenario

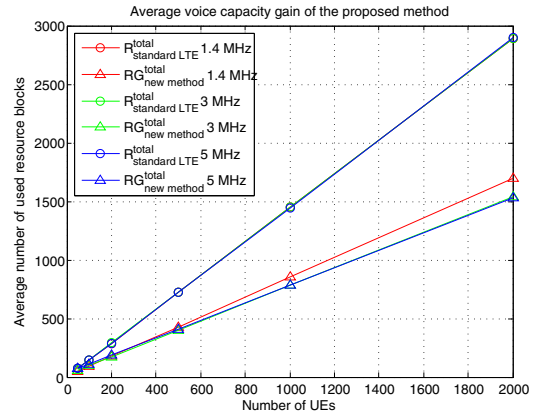


Fig. 8. Used resource blocks of the proposed method for the second scenario

by the factor of 6.5. In this case, the voice capacity of the standard LTE equal to 25 UEs and the voice capacity of the proposed method equal to 189 UEs.

IV. CONCLUSION

A new Multi-users Multiplexing Radio Voice Transmission method is introduced. In this paper, we proposed a new

TABLE II
AVERAGE VOICE CAPACITY GAIN OF THE PROPOSED METHOD FOR THE SECOND SCENARIO

Bandwidth	50 UEs	100 UEs	200 UEs	500 UEs	1000 UEs	2000 UEs
R^{total} 1.4MHz ¹	72	144	289	724	1450	2891
RG^{total} 1.4MHz ²	50	92	177	428	851	1693
Gain 1.4 MHz	44%	56.5%	63.2%	69.1%	70.3%	70.7%
R^{total} 3MHz ³	72	145	290	724	1448	2891
RG^{total} 3 MHz ⁴	60	101	175	402	782	1539
Gain 3 MHz	20 %	43.5%	65.7%	80.1%	85.1%	87.8%
R^{total} 5MHz ⁵	72	144	289	723	1446	2896
RG^{total} 5 MHz ⁶	75	111	187	412	782	1527
Gain 5 MHz	0%	29.7%	54.4%	75.4%	84.9%	89.6%

¹ Average number of used resource blocks in standard LTE for 1.4MHz bandwidth

² Average number of used resource blocks in the proposed method for 1.4MHz bandwidth

³ Average number of used resource blocks in standard LTE for 3 MHz bandwidth

⁴ Average number of used resource blocks in the proposed method for 3MHz bandwidth

⁵ Average number of used resource blocks in standard LTE for 5 MHz bandwidth

⁶ Average number of used resource blocks in the proposed method for 5MHz bandwidth

scheduling method to transmit multiplexed voice packets to the corresponding mobile users. This paper presented also a new algorithm for classifying and grouping voice packets of mobile users that have different MCS values. The proposed scheduling method is implemented by negligible modifications in MAC layer. Therefore, this new multicast does not require additional components in LTE architecture. In our scheduling method, different UEs in a multiplexing group can be scheduled for a same resource allocation. In case of low and medium number of UEs, the results show that the average voice capacity gain of the proposed method can rise up to 89.6%. In this scenario, we found that the greater number of UEs is, the higher the voice capacity gain. The efficiency of our method depends on the dispersion of MCS values, the LTE bandwidth and the number of UEs. In case of high number of UEs in a cell, the results show that the higher MCS, the higher the voice capacity gain. Our method gives positive results even in poor channel condition. Our method's performance is very high in case that the mobile users are concentrate at the center of cell with high channel quality. In the best case, the proposed method can multiply the voice capacity by the factor of 6.5. PMR is designed for emergency services where there is burst communication in a short period. In addition, the PMR service must be available at 100% of the time and emergency calls cannot be interrupted. Therefore, our method is useful and efficient for PMR context. With the same value of MCS, the larger the bandwidth, the higher the voice capacity gain our method can get. This shows the potential of our method with larger LTE bandwidth. In case of LTE-Advanced, with new carrier aggregation feature which allows creating very large bandwidths, the performance of our method is more interesting. This method also allows reducing the bit error rate (BER) in downlink transmission by exploiting the diversity of MIMO-OFDM for large bandwidth utilization and improving the performance of turbo code in LTE by increasing block

size, which is used for the input of turbo code.

However, in this paper, the limitation of the control plan is not yet considered. It might bring a new dimension for considering for the future work.

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