Scheduling and prioritization of packets are one of the most important and complex part of a radio access network such as LTE in case of real time services. Proper scheduling ensures important metrics such as QoS and reliability. One of the easiest solutions to the scheduling problem would be to prioritize VoIP packets; however, this has a very adverse effect of starving other traffic flows in case of high VoIP calls. As an improvement to this approach the authors of [4] have proposed a new model where the priority mode of VoIP traffics is adaptively controlled according to the channel condition. However, the improvement also suffers from low throughput conditions when the priority mode is active. This is due to the relatively small bandwidth requirement of VoIP traffic resulting in a low utilization of Physical Resource Blocks (PRB) allocated to it by evolved-Node B (eNodeB).

This paper proposes a new MAC scheduling scheme which is an improvement over the VoIP priority mode implemented in [4]. Our method combines user pairing with prioritization to efficiently utilize the allocation PRB. This results in a better throughput of the system and thus utilizes the total system capacity. The priority mode duration is adaptively controlled using the channel condition and two users are coupled to share the resources where the controlling user is also determined dynamically using as low control information as possible. This allows gaining a significant system improvement by improving upon the persistent scheduling without using much control signal.

This paper is organized as follows. Section II details the technicalities of carrying VoIP traffic over LTE while Section III discusses about different MAC scheduling schemes of LTE. Section IV describes the proposed scheme which is followed by simulation results in Section V. The concluding remarks are stated in Section VI.

II. VOIP OVER LTE

While carrying voice traffic over a packet-switched technology as LTE, special care needs to be taken. This section describes the basic concepts needed for understanding the intricacies of VoIP over LTE.

A. LTE Frames

In LTE system, both time and frequency plane needs to be cared about. The time plane is divided into 1 ms Transmission Time Interval (TTI) which consists of two 0.5 ms sub frames. Each sub frame consists of 7 OFDM symbols. 2 symbols out of 14 are reserved for UL pilot transmission, the other 12 symbols are used for data and control information transmission. TTI is the minimum allocation unit in time domain. If we consider frequency domain, then the minimum unit is called the Physical
Resource Block (PRB). Each PRB consists of 12 subcarriers of 15 KHz each.

VoIP packet must be transmitted per TTI. It can occupy one or more PRB. An example LTE frame is shown in Fig. 1. For more technical specifications about LTE please refer to [5].

B. Scheduling

One of the most important factors of LTE is scheduling when we consider VoIP traffic. Talking in phone is primarily delivered in spurt, which means; people talk for a period of time and keep silent for the rest. This is called talk spurt. The scheduler needs to behave in such a way that it can schedule the packets of VoIP in real time while a talk spurt is continuing and meanwhile in silent periods, it can schedule other traffics. In the next section various approaches to scheduling packets in LTE are going to be discussed in more detail.

III. MAC SCHEDULING SCHEMES IN LTE

In LTE, primarily two types of MAC scheduling have been proposed, namely, persistent and dynamic scheduling. As their names suggest, persistent scheduling works by scheduling the packets in a fixed basis, similar to the circuit switched fashion. It reserves the bandwidth at the beginning of a call and keeps the reservation until the call ends. On the other hand, dynamic scheduling takes advantage of the feedback information from the User Equipment (UE) about the link situation and schedules the packets thereby. In dynamic scheduling the packets transmission is fully dynamic at the expense of large amount of control signal which may hinder the delivery of data packets. To overcome the demerits of both the previous approaches, an intermediate approach has been devised, named as semi-persistent scheduling. In semi-persistent approach, the packets are not always scheduled dynamically but the decision is taken for a fixed amount of time in future. This method is less costly in the view of control signals and has is dynamic in nature as well.

A. Dynamic scheduling

As LTE is a packet radio system, the packets are usually scheduled using L1/L2 scheduling. VoIP packets can also be scheduled using the same procedure. From the resource usage point of view, this scheme is the most flexible scheme as it allows scheduling the packets exactly whenever they are needed. But the downside of this approach is that it needs a large amount of signaling. In a fully dynamic scheduling, the UE sends resource request in UL for each VoIP packet. The evolved Node B (eNodeB) allocates UL resources for every transmission and retransmission separately by the L1/L2 control signaling. With dynamic scheduling, the VoIP packet scheduling enjoys the full diversity of channel in both time and frequency domain with the expense of large control signaling.

B. Persistent Scheduling

Persistent scheduling means the scheduling of packets does not depend on the channel condition, rather a CS-like allocation of resource is made for each call and the allocation stays constant for the period of the call. Reducing L1/L2 control signaling in this way has been proposed in [6, 7].

IV. PROPOSED SCHEME

RRC signaling or some kind of enhanced L1/L2 signaling is used to allocate a sequence of TTI-RU chunks (time/frequency resource) as well as a fixed modulation scheme for a VoIP user. The allocation also includes resources required for Hybrid Automatic Repeat Request (HARQ) retransmission. As proposed in [8], the allocation can be so persistent that there will be no HARQ ACK/NACK but there will be a fixed number of retransmission for each packet which results in a fixed Forward Error Correction (FEC) scheme instead of an adaptive HARQ scheme. The main problem of persistent scheduling is the wastage of resource. In a very dynamic environment as the wireless environment, it is very hard to determine the exact number of retransmission and resource needed for a packet. It depends on channel quality, interference etc. Therefore, it is very hard to allocated fixed resource to each user and it may cause serious after-effects including very low-utilization of capacity.

C. Semi-Persistent Scheduling

The discussion about dynamic and persistent scheduling showed clearly that none of excessive control signal or total absence of control signal is suitable for VoIP users. Semi-persistent scheduling is a hybrid way of scheduling VoIP packets which uses a small amount of control signaling to determine the channel condition after every fixed interval and persistently schedules the traffic for the time in between. This approach has shown promises to be the best for VoIP traffic due to its controlled dynamic nature and use of small control signaling.
A. VoIP Priority Mode

The priority mode allocates VoIP calls before any other traffic. This approach has the downside of making other traffics to starve when the density of VoIP calls are high. To offset this problem, the duration for which the priority mode is active is controlled dynamically. While in priority mode, VoIP calls are given highest priority and any left over capacity is provided to other traffics. The PRB allocation scheme is designed by making modifications to Channel Adaptive Fair Queuing [10] and the mode described in [4]. While in VoIP priority mode, the scheduling order of the calls is determined by the queue length and the Signal to Interference-plus-Noise Ratio (SINR) of each call. The theory can be described using the following equation:

\[
D_f(i) = Q_{len}(i) \times \gamma_i
\]

Where \(Q_{len}(i)\) and \(\gamma_i\) is the queue length and the SINR value of ongoing call \(i\) respectively. Equation (1) indicates that the longer the queue and the better the channel condition is, the earlier a corresponding call is scheduled to have assigned PRBs.

On the other hand, as mentioned earlier, the duration of priority mode is controlled dynamically. To be more precise, it depends on the total drop ratio of the packets that is measured at the eNodeB. A preset minimum and maximum drop ratio is used and if the drop ratio is below the minimum threshold then the maximum count of priority mode duration is decreased due to the fact that there are enough resources to carry on with the call or there are enough PRBs given already to the call [4]. On the contrary, if the drop ratio is above the maximum, then the maximum duration is increased because it means that there are not enough resources allocated. If the drop rate is in between the \(\min\) and \(\max\) then the duration of priority mode is kept constant.

B. User Coupling

Simulation shows that the VoIP priority mode described above cannot utilize the full capacity of LTE while in priority mode due to the small size of typical VoIP packets. To overcome this problem in packet scheduling user coupling can be used where two users share the resource allocated to them. While in priority mode resource is allocated to several users by the eNodeB. But due to the unpredictable and ever changing nature of wireless network, the network condition for the users is never the same. It may happen that some users cannot use the full capacity of the allocated resource due to channel fading or poor channel quality. In that case, the resource is not utilized. In user coupling scheme, we propose to share resources of two users having opposite channel condition to offset the low resource utilization of one user by the other. This scheme is a modification of the one proposed in [9]. The proposal consists of pairing VoIP users to have early termination gain amongst users. It maintains the properties of original persistent method. The scheme consists of mainly two parts; user pairing 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 and 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.

C. VoIP Priority mode with User Coupling

The scheduling process functions by combining the above approaches to provide adaptive priority to VoIP traffics using only the drop rate information and then
pairing users in priority mode to offset the low resource utilization of VoIP depending upon the ACK/NACK channel of the users. The approach is not fully dynamic as it does not utilize all the control channel information and it is also not persistent scheduling due to the adaptive change of mode depending on the channel quality. Rather, we can call the scheme as semi-persistent which uses very small control channel information which poses as small load on the channel as possible and adds dynamicity to the process. The flow chart of Fig. 2 shows the total scheduling scheme.

While there is any VoIP traffic, the scheduler enters this mode of scheduling. If more than one user is using VoIP, the scheduler couples the users according to their channel quality. After coupling, resource requires for the pair is computed using a semi-persistent strategy. When the resource is reserved for the pair, it is distributed between them according to the channel quality while the scheduler observes the ACK/NACK channel. In case of ACK/NACK channel value for the pair has different values, and then the authority of the link is changed. This approach is continues until there are no VoIP packets to schedule.

V. SIMULATION AND PERFORMANCE EVALUATION

Simulations were conducted to analyze the proposed method for scheduling PRBs and its effect on VoIP QoS. The parameters used for simulation are provided in table 1.

<table>
<thead>
<tr>
<th>Simulation Parameter</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cell Radius</td>
<td>1 km</td>
</tr>
<tr>
<td>Bandwidth</td>
<td>5 MHz</td>
</tr>
<tr>
<td>PRB Structure</td>
<td>12 subcarriers, 2 subframes</td>
</tr>
<tr>
<td>TTI</td>
<td>1 msec</td>
</tr>
<tr>
<td>Number of available PRB</td>
<td>6</td>
</tr>
<tr>
<td>Max VoIP packet delay</td>
<td>20 msec</td>
</tr>
<tr>
<td>Max VoIP packet drop rate</td>
<td>5%</td>
</tr>
<tr>
<td>Min VoIP packet drop rate</td>
<td>2%</td>
</tr>
<tr>
<td>Modulation for AMC</td>
<td>QPSK, 16QAM, 64QAM</td>
</tr>
</tbody>
</table>

To analyze the performance of the proposed algorithm, we have performed two levels of simulation. The first one measures VoIP packet drop ratio using our method and compares it with persistent scheduling and absolute VoIP priority mode. The results are shown in Fig. 3. The result shows a great improvement over the original persistent mode. Whereas in persistent mode, the acceptable max VoIP packet drop ratio is crossed with only around 10 users, the priority modes works far better due to far less retransmissions of VoIP packets. On the other hand, priority mode with user coupling outperforms the absolute priority mode because whenever a user experiences bad channel quality due to fading or other channel conditions, it takes the authority of the channel and assumes greater part of resource which reduces the chance of possible retransmissions. According to the parameters of Table 1, the proposed algorithm showed little improvement over pure priority mode; to be more precise, with 50 calls priority mode drop ratio is 2.3% while priority with user coupling drop ratio is 1.1%. However, this is because the channel quality is good. With additional channel disturbance introduced, for 50 VoIP users, the proposed method have less than 2% packet drop ratio while the absolute priority mode has reached 5% mark.

On another set of simulations, where the throughput of eNodeB was measured, the VoIP priority mode performed badly due to the small VoIP packet size and small queue length of VoIP calls. But with the introduction of user coupling along with priority mode, the capacity of the system is utilized properly and the performance shows better than the persistent mode and VoIP priority mode, as shown in figure 4.

In summary, the proposed model, adaptive VoIP priority mode with user coupling, is able to meet the VoIP QoS requirements while being able to dramatically decrease the VoIP packet drop ratio and increase the total system throughput. The scheme has integrated adaptive user coupling with VoIP priority mode to overcome the shortcomings of the later.

VI. CONCLUSION

MAC scheduling plays a very important role in achieving real time QoS in VoIP service over a fully packet switched service such as LTE. To achieve a quality compared to CS service, traditional scheduling services such as dynamic and persistent scheduling has their own tradeoffs. For this reason, a completely new hybrid scheduling, aptly named as semi-persistent scheduling, has been invented. In this paper, we have proposed a new semi-persistent scheme for MAC scheduling in LTE where the original persistent scheduling is modified to use small amount of control channel information to change the scheduling parameters dynamically according to the channel condition. The proposed method uses a VoIP priority mode and integrates it with adaptive user coupling to better utilize the system capacity. Simulations were also performed where it is showed that the proposed scheme performed better in decreasing packet drop ratio and also increases total system throughput in bad channel condition.

REFERENCES


Figure 3: VoIP packet drop rate comparison

Figure 4: Total System throughput comparison


