Estimation of Effective Radio Resource Usage for VoIP Scheduling in OFDMA Cellular Networks

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***Abstract*— This work presents a method to estimate the amount of radio resource required to support a given number of Voice Over IP (VoIP) users in Orthogonal Frequency Division Multiple Access (OFDMA) based cellular systems. The proposed method is useful for the Call Admission Control (CAC) module, which needs to estimate the radio resource utilization before admitting new calls in the system. It is also useful in predicting the maximum number of VoIP users that can be supported for different cell scenarios and spatial user distributions. The proposed method is evaluated for Dynamic as well as Semi-Persistent (SP) scheduling algorithms. Results are presented for 1x2 Maximum Ratio Combining (MRC) and 4x2 Space Frequency Block Code (SFBC)-Frequency Switched Time Diversity (FSTD) modes. The 3rd Generation Partnership Project (3GPP)-Long Term Evolution (LTE) system parameters are used for performance analysis in this work.**

I. Introduction

Modern Orthogonal Frequency Division Multiple Access (OFDMA) based cellular systems, namely Worldwide Interoperability for Microwave Access (WiMAX) and 3rd Generation Partnership Project (3GPP)-Long Term Evolution (LTE) [1] are completely Internet Protocol (IP) based networks. The maximum achievable capacity of these systems are limited by the time and frequency selective wireless fading channel. Cross layer optimization techniques such as Link Adaptation (LA) and Packet Scheduling (PS) are some of the essential components which help in increasing the throughput of such systems.

The LA unit chooses the highest achievable data rate for a user under a given link condition, while meeting the Packet Error Rate (PER). The PS algorithm maximizes the sum user throughput by adapting the scheduling delay of each user as per the Channel State Information (CSI) provided by the LA unit [2].

Traditional circuit switched voice calls are supported as Voice over IP (VoIP) in 3GPP-LTE networks. LTE is expected to support a very high number of VoIP users while satisfying the required Quality of Service (QoS). In case of VoIP, the QoS is defined by a maximum allowed end-to-end delay and packet loss rate. VoIP traffic is characterized by random arrival of small packets with stringent delay requirements. On the other hand the maximum VoIP capacity in terms of number of users is heavily influenced by the Control Channel (CCH), as CCH allows only a limited number of packets to be dynamically scheduled in each Transmit Time Interval (TTI) over the air interface [3], [4]. In wireless networks, the PS for VoIP therefore must be designed to match the random VoIP packet arrivals to the channel variations of each user in the best possible way so as to maximize the number of supportable VoIP users while satisfying the stringent QoS and CCH restrictions.

Dynamic scheduling strategy for VoIP for 1x2 Maximum Ratio Combining (MRC) mode is found to support up to 272 users [5]. Since this method uses Sub-Band (SB) Channel Quality Indicator (CQI) based scheduling by taking advantage of frequency domain link adaptation, the Physical Data Control Channel (PDCCH) saturates before the Physical Downlink Shared Channel (PDSCH) bandwidth is fully utilized [4], [6]. Hence the capacity of such scheduling schemes is limited by the CCH overhead [7]. Semi-Persistent (SP) scheduling, which is based on Wide Band (WB)-CQI, has the potential to reduce the CCH overhead [8]. Up to 300 VoIP users can be supported using SP scheduling for 1x2 MRC [9], [10]. These mentioned VoIP capacities are for Urban Macro (UMa) cell scenario with Inter Site Distance (ISD) 500m. However, in Urban Micro (UMi) cell scenario, where Base Station (BS)/ evolved Node-B (eNB)s are placed at closer distance with ISD of 200m [11], the variation of interference is expected to affect the VoIP capacity. In contrast to these, one of the objectives of this work is to present the details of Dynamic and SP scheduling algorithms for VoIP traffic in UMi cellular environment for 1x2 MRC as well as for 4x2 Space Frequency Block Code (SFBC)- Frequency Switched Time Diversity (FSTD) modes. In this work, one complete VoIP packet is scheduled during one TTI for one user while maintaining Transport Block Size (TBS) size restrictions [1]. Packet bundling is not considered in this work.

The number of Physical Resource Blocks (PRB)s required, even for fixed size VoIP packets in LTE, vary with time and location of the users due to rate adaptation. Further, the PRB allocation in LTE for VoIP is in accordance with the Voice Activity Factor (VAF). A Silence Indicator (SID) is used to represent long periods of silence between talk spurts [11]. So, the radio resource utilization for VoIP is not directly proportional to the number of active users only unlike Global System for Mobile Communications (GSM) systems. In this work a method to estimate the average number of PRBs [12] required to make a VoIP call, as a function of the Signal to Interference plus Noise Ratio (SINR) of the user is presented. Using this, a method to predict the average number of required PRBs, in order to support a certain number

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of VoIP users given the user distribution (SINR) and the cell configuration is also proposed. A cell configuration is described by a set of parameters as given in Table I. Of these, some of the important factors which influence the VoIP capacity for a given bandwidth are the ISD and the spatial user distribution. The estimate of the number of required PRBs is an important input to the Call Admission Control (CAC) unit [12]. The concept of Effective Bandwidth (EB) was first introduced in [13] where analyses are done from queuing theory approach. In [14] author has discussed similar concept using Shannon's capacity model and evaluates call blocking probability but resource usage estimation of ongoing calls are not presented. Although International Telecommunication Union (ITU) specifies a 50:50 indoor to outdoor (I:O) user distribution, results for two special cases of 70:30 and 90:30 I:O user distribution is also presented to support the model developed in this work.

In LTE, the capacity in terms of number of satisfied users depends on users' SINR distribution. In case of VoIP when the number of users in the system exceeds a certain limit, the fall in the percentage satisfied VoIP users is catastrophic [10]. In other words, it is important for the CAC unit to estimate the average number of available PRBs per TTI in order to limit the admission of new VoIP users so as to prevent the rapid degradation of ongoing calls. This can be done using the methods described in the work. It is to be noted that in order to plan the VoIP capacity of a network, for a given cell configuration, a detailed system level simulation is required. Such simulations take long duration to produce results and often several trials are needed in order to reach a conclusion. The method described in this work can also be used to predict the maximum VoIP capacity for a given user distribution and antenna configuration without performing detailed system level simulations. Capacity estimation of VoIP as shown in [15] does not consider scheduling scheme or CCH limitation. Although VoIP capacity evaluation based on user latency is developed for OFDMA in [16], yet it is not usable as Adaptive Modulation and Coding (AMC) is not considered. The VoIP performance for different access techniques shown in [17] is from link spectral efficiency perspective, where system simulations are ignored. Analysis for VoIP capacity is also presented in [18], but it is applicable to Wireless Local Area Network (WLAN) systems.

The rest of the paper is organized as follows. Section II presents the system model, Section III describes the scheduling algorithms and VoIP capacity limits are shown in Section IV. The method to estimate number of required PRBs per TTI is discussed in V. Finally, results for PRB estimation are given in Section VI and conclusions are presented in Section VII.

II. CELLULAR SYSTEM MODEL

To evaluate the performance of the algorithms and models, methodology as suggested by ITU [11] is followed. The simulation configuration used is as specified by ITU [11] and 3GPP [1], [19]. Inter-cell interference is generated by all the base stations considering full traffic load. The UMi [11] test environment is considered for simulation with 50 % indoor and 50 % outdoor users. For usability of the prediction model two special cases with different I:O as discussed above are also considered. Propagation models and VoIP traffic model used in this work as described in [11]. The parameters are summarized in Table I. Calibration result is shown in Section V. Link to system mapping is done using Exponential Effective SINR Mapping (EESM) [20]. It is further assumed that Physical Downlink Control Channel (PDCCH) allows a maximum of 8 users to be scheduled per TTI dynamically [21]. The details of VoIP traffic as considered in this work is described in [11].

TABLE I
SYSTEM SIMULATION PARAMETERS

Parameters	**Values**
Network Layout	Hexagonal Grid with 19 Sites
Sectors per Cell	3
Scenario	Urban Micro [11]
ISD	200 m
Carrier Frequency	2.5 GHz
eNB Transmit Power	41 dBm
Antenna Configuration	1x2 MRC, 4x2 SFBC–FSTD
System Bandwidth	5 MHz
Sub-carrier Spacing	15 kHz
Number of PRBs	25 (12 Sub-carriers / PRB)
TTI Duration	1 ms (14 OFDM Symbols)
Control Signaling Overhead	3 / 14 OFDM Symbols
Distribution of Users	Uniform
Indoor to Outdoor Users (I:O)	50% : 50%
Channel Model	ITU-R M.2135
Rx Noise Figure	7 dB
User Velocity	3 kmph
Shadow Fading σ (dB)	4 for Outdoor, 7 for Indoor Users
Link to System mapping	EESM
Link Adaptation BLER Target	10 %
CQI	Resolution 1 CQI / 2 PRBs Reporting Delay 2 ms
HARQ	8 HARQ Channels per User Stop-and-Wait (SAW) Ideal Chase Combining (CC) with Max 4 Retransmission [22]

III. VOIP SCHEDULING

The generalized description of packet scheduling and radio resource allocation is shown in Figure 1.

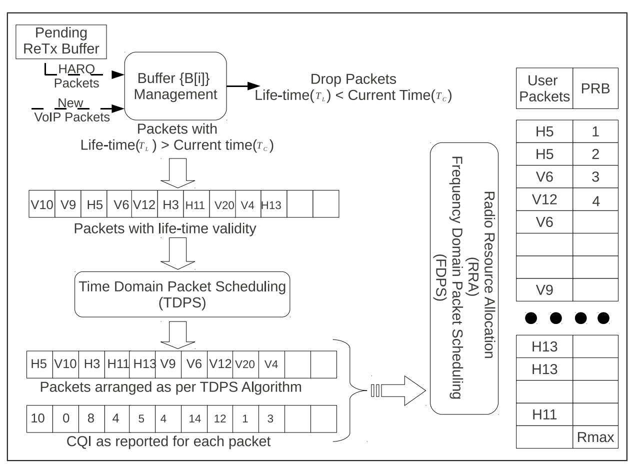

Fig. 1. Generalized VoIP Scheduler Architecture

A. Dynamic Scheduling

In Dynamic VoIP Scheduling each VoIP packet is scheduled based on SB CQI feedback. The scheduling algorithm can be summarized as follows:

1) Store new VoIP packets and retransmission packets in the input buffer $\{\mathrm{B[i]}\}$, for $i = 1, 2, 3...B_{Max}$. Where B_{Max} is the maximum number of users in the buffer.
2) Drop packets whose $\mathrm{T_L} < \mathrm{T_C}$, where $\mathrm{T_L}$ is the lifetime defined as the absolute time within which the packet must be transmitted out of the BS and $\mathrm{T_C}$ is the current time OR whose number of transmissions $\mathrm{N_{ReTx}} > \mathrm{N_{ReTx_{Max}}}$, where $\mathrm{N_{ReTx}}$ is the number of transmission for that packet and $\mathrm{N_{ReTx_{Max}}}$ is the maximum allowable retransmissions.
3) Select the packets which have at least one Hybrid ARQ (HARQ) channel available.
4) **Time Domain Packet Scheduling (TDPS)**
   a) $\{\mathrm{Bs[j]}\} = \underset{ascend(\mathrm{T_L})}{\mathrm{SORT}}(\{\mathrm{B[i]}\})$.
   b) Send $\{\mathrm{Bs[j]}\}$ to Frequency Domain Packet Scheduling (FDPS).
5) **FDPS (Dynamic Scheduling)**
   a) Inputs required:- Sorted buffer from TDPS $\{\mathrm{Bs[j]}\}$; PRB list $\{\mathrm{R[m]}\}$, for $m = 1, 2, ...R_{Max}$.; CCH list $\{\mathrm{CH[n]}\}$, for $n = 1, 2, ..CH_{max}$; SB-CQI feedback for k^{th} user on s^{th} SB $\{\mathrm{CQI_{SB}[k,s]}\}$
   b) Schedule dynamic user's packets
   for $\{j = 1$ to $Bs_{Max}\}$ **do**
      i) If $\{\mathrm{R[m]}\}$ and $\{\mathrm{CH[n]}\}$ are free then go to step 5(b)ii, otherwise go out of the loop.
      ii) Select the packet to schedule $\mathrm{P_{Sch}}$ = Bs[j].
      iii) Select Modulation and Coding Scheme (MCS) and form appropriate Transport Block (TB) for $\mathrm{P_{Sch}}$. Allocate PRBs from $\{\mathrm{R[m]}\}$.
         A) If $\mathrm{P_{Sch}}$ is a new packet, select the best available PRBs $\{\mathrm{R[m]}\}$ as per the $\mathrm{CQI_{SB}[k,s]}$ which matches the packet size.
         B) If $\mathrm{P_{Sch}}$ is a retransmission packet, use the MCS as in the first transmission.
      iv) Remove the allotted PRBs from $\{\mathrm{R[m]}\}$. Fill control channels $\{\mathrm{CH[n]}\}$ as per requirement.
      v) $j = j + 1$
   end for

B. Semi-Persistent (SP) Scheduling

In SP scheduling, the talk spurt VoIP packets are scheduled persistently while HARQ, SID and VoIP packets at the start of talk spurt are scheduled dynamically as described in the last section. In contrast to Dynamic scheduling WB CQI is used for SP scheduling algorithm. FDPS algorithms for SP scheduling are given below while Steps 1 to 4 of Section III-A remain same.

FDPS (Semi-Persistent Scheduling)

1) Inputs required:- Persistent Table $\{\mathrm{P[t_r,p]}\}$, for $p = 1, 2, ...P_{Max}$, where t_r is the relative time index and P_{Max} is the maximum number of users present in the table at t_r Refer Figure 2; Sorted buffer from TDPS $\{\mathrm{Bs[j]}\}$; PRB list $\{\mathrm{R[m]}\}$, for $m = 1, 2, ...R_{Max}$; CCH list $\{\mathrm{CH[n]}\}$, for $n = 1, 2, ..CH_{max}$.; WB CQI feedback for k^{th} user $\{\mathrm{CQI_{WB}[k]}\}$

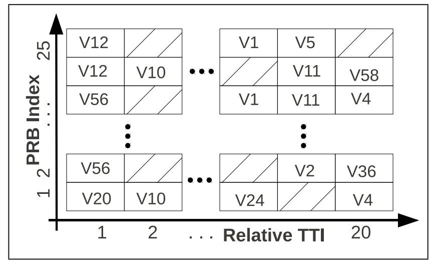

Fig. 2. Table for Persistent Users

2) Schedule persistent user's packets (**No CCH is used**)-
   for $\{p = 1$ to $P_{Max}\}$ **do**
   a) Select the packet to schedule $\mathrm{P_{Sch}}$ = $\{\mathrm{P[t_r,p]}\}$.
   b) If $\mathrm{P_{Sch}}$ is present in $\{\mathrm{Bs[j]}\}$, then select MCS, as in the last transmission and form TB for $\mathrm{P_{Sch}}$. Allocate PRBs from $\{\mathrm{R[m]}\}$. Otherwise free up the entry in $\{\mathrm{P[t_r,p]}\}$.
   c) Remove the user from $\{\mathrm{Bs[j]}\}$.
   d) $p = p + 1$
   end for
3) Remove the allotted PRBs from $\{\mathrm{R[m]}\}$.
4) Schedule Packets which were not selected above
   for $\{j = 1$ to $Bs_{Max}\}$ **do**
   a) If $\{\mathrm{R[m]}\}$ and $\{\mathrm{CH[n]}\}$ are free then go to step 4b, otherwise go out of the loop.
   b) Select the packet to schedule $\mathrm{P_{Sch}}$ = Bs[j].
   c) Select MCS and form appropriate TB for $\mathrm{P_{Sch}}$. Allocate PRBs from $\{\mathrm{R[m]}\}$.
      i) If $\mathrm{P_{Sch}}$ is a new packet, allocate PRBs from $\{\mathrm{R[m]}\}$ as per the $\mathrm{CQI_{WB}[k]}$ which accommodates the packet. **Update** $\{\mathrm{P[t_r,p]}\}$**.**
      ii) If $\mathrm{P_{Sch}}$ is a retransmission packet, use the MCS as in the first transmission.
   d) Remove the allotted PRBs from $\{\mathrm{R[m]}\}$. Fill control channels $\{\mathrm{CH[n]}\}$ as per requirement.
   e) $j = j + 1$
   end for

C. Improved SP Scheduling

The given VoIP SP algorithm is further modified in such a manner where the scheduler monitors WB-CQI of persistent users and when the CQI reported changes then the scheduler reallocates the persistent users dynamically. In this case PRBs are utilized efficiently. Therefore if either of the two conditions below is satisfied then the packet is scheduled dynamically,

- Present WB CQI < MCS indicated by $\{\mathrm{P[t_r,p]}\}$.
- Present WB CQI $\geq$ MCS indicated by $\{\mathrm{P[t_r,p]}\}$ $+\Delta$.

where Δ is a factor which decides that when an user should be reassigned. In this work it is set to 2.

IV. Results of VoIP Scheduling

Table II shows the performance of Dynamic and SP scheduling in terms of maximum number of supportable users for 1x2 and 4x2 antenna configuration without packet bundling.

TABLE II
VoIP Capacity for 5 MHz

Scheduling Scheme	Supportable VoIP Users
Dynamic 1x2 MRC	275
Semi Persistent 1x2 MRC	310
Semi Persistent 4x2 SFBC-FSTD	355

V. Estimation of Av. Num. of Required PRBs

A method to estimate the number of PRBs required to make a VoIP call and hence a technique to estimate the maximum number of users supportable for a given number of available PRBs is described in this section. Figure 3 shows the average number of scheduled PRBs vs. G-factor [21] for a VoIP call for both SP (1x2 and 4x2) and Dynamic (1x2) scheduling algorithms. These results are obtained from a large number of system level simulation for full capacity, assuming normal cell configurations as mentioned in Table I. The relationship between the average number of allocated PRBs required to make a VoIP call with varying geometry can be expressed as

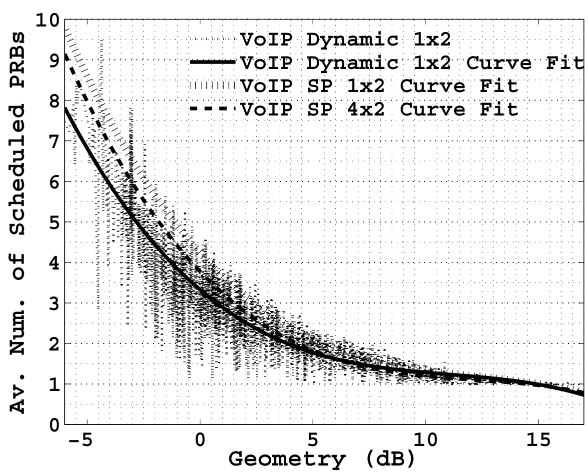

Fig. 3. Geometry vs. Average Number of PRB Scheduled

$$\mathrm{N_{PRB}}(\mathbf{G}) = \mathbf{a}.(\mathbf{G})^3 + \mathbf{b}.(\mathbf{G})^2 + \mathbf{c}.(\mathbf{G}) + \mathbf{d} \tag{1}$$

where Coefficients $\mathbf{a}, \mathbf{b}, \mathbf{c}, \mathbf{d}$ used in this expression are given in Table III and $\mathbf{G}$ is the downlink geometry (G-factor) of users. $\mathrm{N_{PRB}}(\mathbf{G})$ as described in (1) captures the effect of the link layer of LTE. Geometry of user k is,

$$\mathbf{G_k} = \frac{\mathrm{P_{Tx}^D . P_L^D . A_G^D}(\theta,\phi).\chi^{\mathrm{D}}}{\sum_{i=1, i\neq D}^{N_{Sec}} (\mathrm{P_{Tx}^i . P_L^i . A_G^i}(\theta,\phi).\chi^{\mathrm{i}}) + \mathrm{N_0 . N_F . B_w}} \tag{2}$$

where the variables in (2) are described in the Table IV.

Figure 4 shows the Cumulative Distribution Function (CDF) of downlink geometry in urban micro scenario for different

TABLE III
Coefficients for Different VoIP Scheduling

Scheduling Scheme (SS)	a	b	c	d
Dynamic 1x2 MRC	-0.0012	0.039	-0.48	3.3
Semi Persistent 1x2 MRC	-0.0012	0.045	-0.62	4.2
SP 4x2 SFBC-FSTD	-0.0012	0.045	-0.58	3.8

TABLE IV
Notations used in the expression 2

Variables	Description
$()^D$	Desired Link
$()^i$	Interfering Link
$\mathrm{N_{Sec}}$	Number of Sectors
$\mathrm{P_{Tx}}$	Transmission Power
$\mathrm{P_L}$	Path Loss
$\mathrm{A_G}(\theta,\phi)$	Antenna Gain
χ	Shadow Fading
$\mathrm{N_0}$	Noise Power Spectral Density
$\mathrm{N_F}$	Noise Figure
$\mathrm{B_w}$	Transmission Bandwidth

ratio of indoor to outdoor users. For I:O = 50:50 ratio the result is in agreement with 3GPP self-evaluation results [23], which is a calibration for the simulator developed.

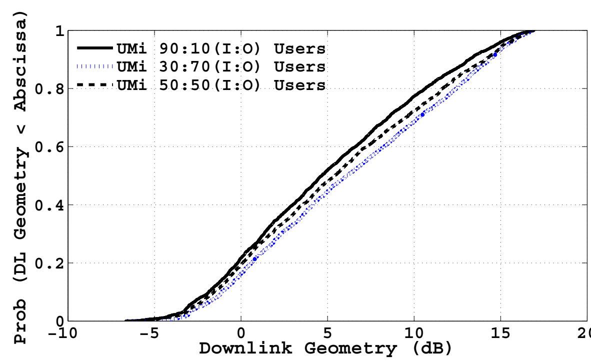

Fig. 4. CDF of Downlink Geometry for Different Indoor:Outdoor (I:O) User Distribution

The average number of PRBs required to sustain a VoIP call in the cell for a given user distribution and cell configuration (hence G-factor) and scheduling scheme is

$$\overline{\mathrm{N_{PRB}}} = \int_0^\infty \mathrm{N_{PRB}(x) p_G(x) dx}, \tag{3}$$

where $\mathrm{p_G}$ is the Probability Density Function (PDF) of geometry.

In order to estimate the required number of PRBs for a given number of VoIP users, an important parameter is Inter Scheduling Interval (ISCHI). ISCHI is defined as the average interval in number of TTIs between two scheduling events for an user. Therefore, in 1 second, the average VoIP packet generation rate R is given by-

$$\mathrm{R} = \frac{\mu_{\mathrm{TS}}}{\mu_{\mathrm{TS}} + \mu_{\mathrm{SP}}} \frac{1}{\mathrm{T_{On}}} + \frac{\mu_{\mathrm{SP}}}{\mu_{\mathrm{TS}} + \mu_{\mathrm{SP}}} \frac{1}{\mathrm{T_{Off}}} \tag{4}$$

where $\mathrm{T_{On}}$ is packet generation interval in ms during voice active state (Talk-spurt) and $\mathrm{T_{Off}}$ is the packet generation interval in ms during inactive state (Silence Period). In (4)

$\frac{\mu_{\rm TS}}{\mu_{\rm TS}+\mu_{\rm SP}} = \lambda$, while $\frac{\mu_{\rm SP}}{\mu_{\rm TS}+\mu_{\rm SP}} = (1-\lambda)$, where λ is the VAF. So, (4 can be written as-

$$\mathrm{R} = \frac{\lambda}{\mathrm{T_{On}}} + \frac{1-\lambda}{\mathrm{T_{Off}}} \tag{5}$$

Now assuming that the scheduler allocates PRBs to each packet coming in the scheduler, we can define average ISCHI as $\overline{\mathrm{T_{isi}}} = \frac{1}{\mathrm{R}}$. Taking 50% VAF and other parameters as given in [11], $\overline{\mathrm{T_{isi}}}$ equals to 35.56 (without HARQ overhead).

It is understood that if $\mathrm{N_{PRB_{Total}}}$ is the maximum number of PRBs available and $\overline{\mathrm{N_{PRB}}}$ is the average number of scheduled PRBs per user, then $\frac{\mathrm{N_{PRB_{Total}}}}{\overline{\mathrm{N_{PRB}}}}$ is the number of schedulable users (packets) per TTI. However, since maximum number of schedulable new users (packets) are limited by $\mathbf{N}_{\mathrm{u_{Max}}}$ (Maximum number of allocated packets in each TTI due to CCH limitation), therefore, number of users (packets) scheduled per TTI is given by $\min(\frac{\mathrm{N_{PRB_{Total}}}}{\overline{\mathrm{N_{PRB}}}}, \mathbf{N}_{\mathrm{u_{Max}}})$. Since, one user will be scheduled again only after $\overline{\mathrm{T_{isi}}}$ on an average, the number of TTI, therefore, the number of users supportable for a given number of PRBs and CCH limit is,

$$\mathcal{N}_{\mathrm{USup}} = \min(\frac{\mathrm{N_{PRB_{Total}}}}{\overline{\mathrm{N_{PRB}}}}, \mathbf{N}_{\mathrm{u_{Max}}}) \times \overline{\mathrm{T_{isi}}} \tag{6}$$

Except for full capacity dynamic VoIP scheduling, the function 'min' can be removed since the term $\frac{\mathrm{N_{PRB_{Total}}}}{\overline{\mathrm{N_{PRB}}}}$ is found to be less than $\mathbf{N}_{\mathrm{u_{Max}}}$ for all other cases. So

$$\mathcal{N}_{\mathrm{USup}} = \frac{\mathrm{N_{PRB_{Total}}}}{\overline{\mathrm{N_{PRB}}}} \times \overline{\mathrm{T_{isi}}} \tag{7}$$

The average number of PRBs required for $\mathcal{N}_{\mathrm{U}Total}$ VoIP users' geometry distribution is,

$$\mathrm{N_{PRB_{Required}}} = \frac{\mathcal{N}_{\mathrm{UTotal}} \times \overline{\mathrm{N_{PRB}}}}{\overline{\mathrm{T_{isi}}}} \tag{8}$$

considering CCH limitation is not present as described above.

VI. Results for Estimation of PRBs Required and VoIP Capacity Prediction

Results for estimating the number of required PRBs for a given number of users as well as predicting the maximum VoIP capacity for a given cell scenario using the method described in Section V is presented here. Evaluation parameters are described in Section II.

The parameters in Table III are obtained from maximum VoIP capacity using curve fitting from system simulation where I:O user distribution is 50:50. It is used for predicting the average number of PRBs used per TTI for a given number of VoIP users 8 as shown in Figure 5. We validate the developed model by considering two different scenarios with I:O users ratio of 90:10 and 70:30 [24]. This gives rise to two different geometry distributions as shown in Figure 4. Results for these scenarios are shown in Figures 6 and 7. It is seen that the predicted values are in agreement with the simulation results. Root Mean Square (RMS) value of error in terms of average PRB is 0.77. Results for predicting the maximum VoIP capacity using the models (6) and (7) is shown in Figure 8. The model is derived from the originating scenario i.e. where I:O = 50:50, and this is used to predict the capacity for different scenarios as described above. The predicted numbers are found to be reasonably acceptable with 1% error except for 90% Indoor users (1x2 MRC) where the error is 4%.

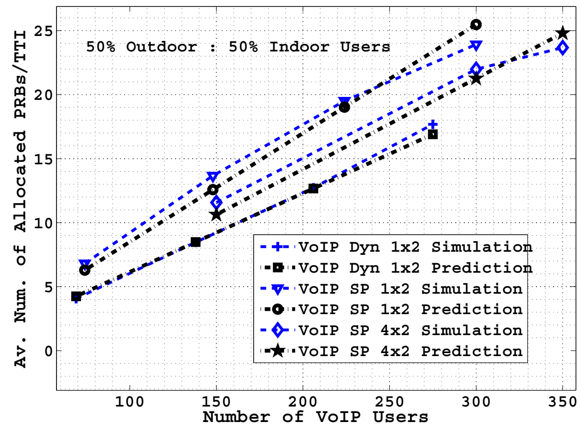

Fig. 5. Average Number of PRBs Allocated per TTI for varying Number of VoIP Users

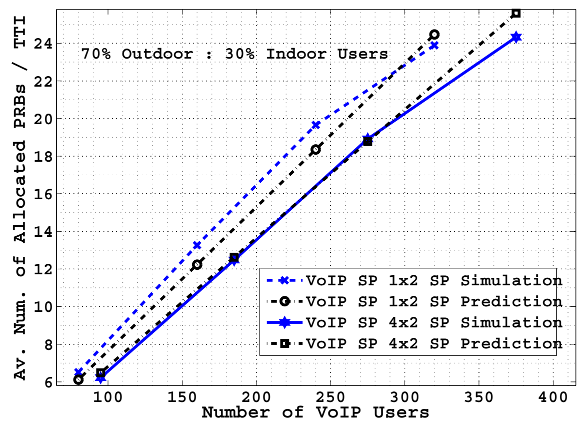

Fig. 6. Average Number of PRBs Allocated per TTI for varying Number VoIP Users (70% Outdoor Users)

VII. Conclusion

Detailed algorithm for Dynamic and SP VoIP scheduling are presented in this paper. The capacity limits without packet bundling in a 5 MHz LTE system is given. It is shown that Dynamic scheduling supports up to 275 users with 1x2 antenna configuration, whereas Semi-Persistent supports 310 users with 1x2 antenna configuration and 355 users with 4x2 antenna configuration. The method to predict the number of required radio resources (PRBs) for a given number of VoIP users is found to produce results which is within 1 PRB of the simulated numbers. Similarly, the maximum capacity prediction model is found to have a mean percentage error of 1.28. The results hold the key to successful CAC implementation.

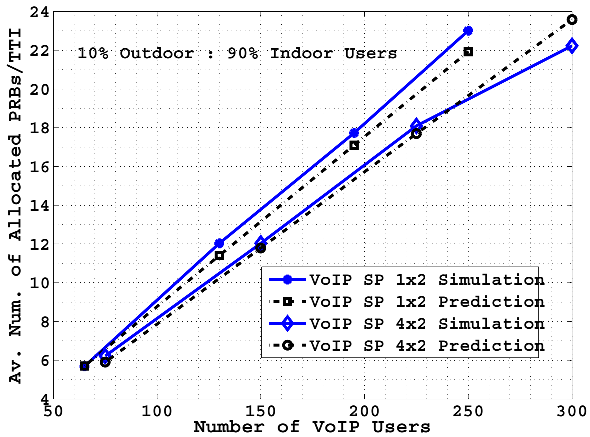

Fig. 7. Average Number of PRBs Allocated per TTI for varying Number VoIP Users (90% Indoor Users)

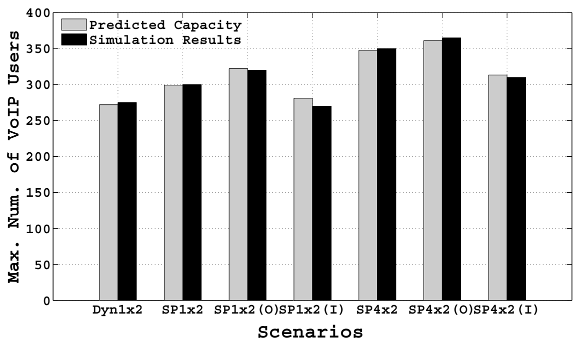

Fig. 8. Comparison of Maximum VoIP Capacity for Different Scenarios

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