Evaluation and Improvement of VoIP Capacity for LTE

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***Abstract*—LTE-Advanced (LTE-A) proposed by the 3rd Generation Partnership Project (3GPP) has been accepted by the International Telecommunication Union (ITU) in 2010 as an IMT-Advanced (IMT-A) compliant 4G mobile radio system. Besides high spectral efficiency for data services, LTE-A had to prove its capability to support a large number of voice calls through Voice over IP (VoIP).**

Radio resource management for many VoIP calls continuously switching between active and inactive state is challenging. Control channel limitations prevent the dynamic scheduling of every VoIP connection in each 1 ms subframe. Semi-persistent scheduling enables channel quality aware scheduling accounting for control channel limitations.

We describe and evaluate by simulation five semi-persistent radio resource assignment algorithms focusing on boosting uplink VoIP capacity. Results show that high capacity gains are achieved under smart resource assignment strategies.

I. Introduction

Despite the increasing demand for packet data based communication, voice communication will always play an important role in wireless networks. Long Term Evolution (LTE) and LTE-Advanced [1] networks are fully IP-based and do not include a circuit switched domain for voice communication as known from GSM and UMTS networks.

Packet switching for voice traffic increases efficiency from multiplexing of connections to the same radio resource. The drawback is the limited capability of a packet switched network to guarantee Quality of Service (QoS) requirements.

4G candidate networks had to prove their performance within the IMT-Advanced (IMT-A) evaluation process specified by ITU-R [2] requiring IMT-A systems to support at least 40 voice calls per MHz bandwidth and cell. A cell must therefore be capable to carry user data and control traffic for 200 VoIP calls at 5 MHz system bandwidth. While the required data rate for voice communication is relatively small compared to other applications, voice traffic requires low packet latencies. As the number of VoIP calls in a system increases, packet loss and delay increase until the number of satisfied users drops below a given percentage defining the VoIP capacity. We study the influence of radio resource control algorithms on the number of satisfied users with focus on the uplink.

This paper is organised as follows: After introducing the IMT-A evaluation methodology to estimate the Voice over IP (VoIP) capacity of a system, we describe the problems related to voice traffic scheduling in Section II. In Section III we present multiple semi-persistent scheduling algorithms and evaluate their performance in Section IV. We present our conclusions in Section V and give an outlook on possible future work.

A. IMT-A VoIP Evaluation

The IMT-Advanced evaluation methodology for VoIP capacity is described in [2]. A two-state traffic model is used where the states correspond to a user either talking (active) or listening (inactive). A 248 bit VoIP Protocol Data Unit (PDU) is transmitted every 20 ms in active state resulting in 344 bit data of the MAC-PDU after adding lower layer headers and header compression. In inactive state a 40 bit Silence Insertion Descriptor (SID)-PDU is transmitted every 160 ms resulting in a 144 bit MAC-PDU. The probability for state transition is 0.01 resulting in a mean state sojourn time of $1/0.01 \cdot 20$ ms $= 2$ s.

According to [2], a VoIP user is satisfied if packet loss probability is less than 2 %. A packet not received within 50 ms is considered lost. System VoIP capacity is reached if less than 98 % of the users are satisfied.

B. Related Work

Semi-persistent scheduling for VoIP traffic in LTE is described in [3] where its superiority over dynamic scheduling under limited control channel resources is proven. The 3GPP has published its VoIP capacity results in [4] proving LTE-Advanced capable to outperform the IMT-A VoIP capacity requirement by a factor of two to three depending on the scenario. Similar results were found by the WINNER+ project evaluation group [5]. Although many baseline assumptions assuring comparable results are described, key factors like the link-to-system interface and a common scheduling algorithm, at least for simulator calibration, are not provided. The used implementations therefore differ across the partners of those evaluation groups, leading to a substantial variance in the results. The authors of [6] provide detailed information on the scheduling algorithm and link-to-system interface for their IMT-A compliant VoIP capacity evaluation. Dynamic resource allocation is used to determine the number of resources for each user depending on the channel quality. No uplink results are provided there. In [7] resource assignment algorithms are evaluated similar to the ones presented in this work, but the authors do not consider control channel capacity limitations, instead assuming dynamic scheduling of a large number of users possible in every subframe. The impact of limited control channel resources on system capacity is evaluated in [8] and [9].

Our contribution is the VoIP capacity evaluation of semi-persistent VoIP scheduling under realistic control channel constraints for both, uplink and downlink.

II. Problem Description

Figure 1 shows the Orthogonal Frequency Division Multiple Access (OFDMA) resource grid to be scheduled. In frequency domain 25 Resource Blocks (RBs) can be occupied by one user each. One or more consecutive RBs form a Transport Block (TB) containing data for one User Terminal (UT). Empty RBs between TBs or on the edge of the grid are named *spaces*.

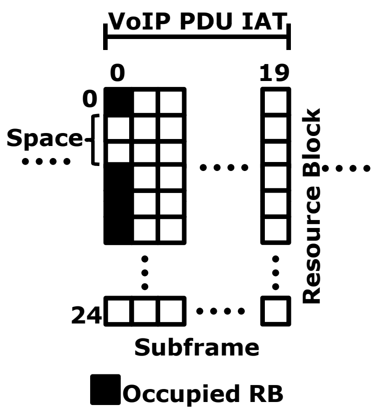

Fig. 1. Fragmented OFDMA resource grid for 5 MHz channel bandwidth.

A. Scheduling under Control Channel Capacity Limitations

If dynamic scheduling is used a Physical Downlink Control Channel (PDCCH) PDU per UT and direction is transmitted to grant radio resources. Depending on the used PDCCH format approximately 20 Control Channel Elements (CCEs) are available at 5 MHz bandwidth with 3 symbols per subframe reserved for control traffic. The amount of CCEs required to transmit a PDCCH PDU to a UT depends on the downlink channel quality at the UT. According to [8], 4 dB Signal to Interference and Noise Ratio (SINR) is sufficient to transmit the PDCCH content in one CCE. Lower SINR values require $2, 4$, or 8 CCEs.

To overcome control channel capacity limitations, semi-persistent scheduling was proposed exploiting the 20 ms fixed Inter-arrival Time (IAT) of VoIP PDUs [3]. Accordingly, periodic resources are granted on appropriate subframes, persistently, to a UT during active state, but SID PDUs and Hybrid Automatic Repeat Request (HARQ) retransmissions are scheduled dynamically, instead.

In downlink the PDCCH PDU informs UTs which RBs should be monitored for incoming traffic data. A group of UTs monitors the same RBs, to which VoIP traffic is multiplexed where only one UT is served at a time. In ideal case resources are granted once to a UT for the entire VoIP call duration. Further signalling is required if the RBs are changed.

In the uplink a fixed set of RBs must be granted to each UT. Granting resources requires a PDCCH PDU transmission whenever a UT switches to active state. Changing the assignment (relocation) requires PDCCH signalling. SID PDUs and HARQ retransmissions are scheduled dynamically.

B. Resource Assignment

UTs switching to active state need a persistent resource allocation. The TB size, the number of consecutive RBs forming the TB, depends on the channel quality of the selected RBs. RBs are chosen from the available spaces in the subframe. Released RBs from inactive calls can lead to fragmentation of RB spaces in a subframe. A modulation and coding scheme (MCS) is selected once when resources are granted and remains unchanged during reservation lifetime.

C. Time Varying Channel Conditions

During the mean talk spurt duration of 2 s the persistently granted resources should provide a sufficient quality for the MCS selected. If the channel quality gets better, resources are potentially wasted, since a smaller TB size would be possible. If the channel quality gets worse, transmission error probability increases. Data not decoded successfully must be retransmitted using HARQ. It might also be necessary to relocate RBs or to increase the number of reserved RBs if possible.

The interference level on a RB depends on the number of other nodes transmitting concurrently. Worst case in downlink is that the RB is used by all surrounding evolved NodeBs (eNBs). Channel quality can then be estimated by evaluating the quality of the reference symbols during control channel transmission when all eNBs transmit simultaneously. Measurements must be reported by the UT to the eNB.

Uplink interference at the eNB depends on the number of UTs transmitting on a RB. Additionally, it depends on the path loss between serving eNB and interfering UTs, which can vary by magnitudes depending on the UT position. As uplink resource allocations in surrounding cells change over time, the interference level may change in every subframe. It is not possible to estimate a worst case channel quality, at least not one that would be sufficient for communication. Uplink reference symbols can be used by the eNB to estimate the channel quality on each RB.

Small scale multi-path fading contributes to varying channel conditions creating a time- and frequency varying channel. This signal fading is correlated in time and frequency. The coherence time and frequency depend on the relative velocity between the nodes. Due to the correlation, past measurements and measurements from close by frequencies can be used to estimate current and future channel quality on a RB and neighbouring RBs.

D. Delay

A PDU remains queued until it is scheduled for transmission causing a scheduling delay. This delay may be increased by delays resulting from uplink resource requests informing the eNB scheduler about queued UT data. Transmission of the subframe requires 1 ms, and another 3 ms delay for decoding. Every HARQ retransmission causes an additional 8 ms delay for decoding, transmitting the Negative Acknowledgement (NACK) and retransmitting the TB.

Scheduling delays due to the lack of free resources and multiple HARQ retransmissions resulting from erroneous channel estimation can increase the delay beyond 50 ms, causing packet loss.

III. Scheduling Algorithm

A semi-persistent resource scheduler was designed and implemented complying with previously described VoIP scheduling problems. The main focus is put on uplink scheduling which is more challenging because of higher variability of uplink inter-cell interference.

A. Resource Assignment Strategies

Uplink channel quality is estimated by measuring the interference level on each RB at the eNB and a path-loss estimation of the respective UT. To further improve the accuracy, interference estimates for each RB are stored separately for each subframe within a 20 ms period matching the VoIP PDU IAT. For the downlink estimation, worst case SINR is assumed, collected from wideband measurement reports.

The goal is to select suitable resources from the available spaces of the resource grid. The set of potential TBs is created by the following procedure: The first RB in each space is selected and its SINR is considered. If it is sufficient for a MCS requiring only one RB, it is added to the set of potential TBs. Else the effective SINR of a TB spanning the first and second RB is evaluated. Mutual Information (MI) based averaging, as described in [10], is used. If the PDU fits the resources, the TB is added to the set of potential resources. Else the TB size is further extended until either the PDU fits or all RBs in the space have been checked without meeting the required quality. This is repeated for every RB in a space as potential starting RB of the TB.

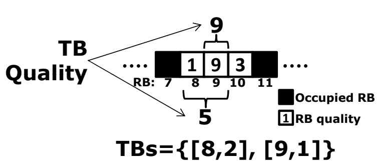

Fig. 2. Example candidate TB set.

Figure 2 shows an example with a three RB long space starting at RB 8 with different quality on each RB. If the average quality is above 4, a TB size of 1 RB is possible, else two RBs form a TB. RB 8 cannot provide sufficient quality for TB size 1, so RBs 8 and 9 are checked, which average quality 5 is sufficient to fit the data. The TB starting at RB 8 with size 2 is added to the set of potential TBs. Next the potential TB starting at RB 9 is evaluated. The quality is sufficient to form a TB of size 1. A TB starting at RB 10 is not possible since the end of the space is reached and the quality is below the threshold for TB size 1.

The strategy used to select a TB out of the set of candidates can then consider the TB size and the size of the resulting space(s) for its decision. The following strategies are applied:

First: Use the first TB in the set of potential TBs. This is usually the one with the lowest starting RB index. This results in higher interference at the first RBs in all cells while RBs with higher indices experience less interference.

Random: A random TB is chosen. This results in an equal distribution of interference across all RBs.

BestFit: Choose the TB that best fits a space. Therefore the difference between the length of the space and the length of the TB is minimised. This can help to reduce fragmentation by occupying entire spaces whenever possible. This can lead to favouring highly interfered RBs resulting in long TBs which will more likely occupy more RBs of a space.

LeastFit: Choose the TB that will fill its space least. The TB resulting in the largest difference between space length and TB length is chosen. The motivation is to create remaining spaces that are likely capable to fit future resource demands. This strategy favours RBs with low interference resulting in short TB sizes.

Smallest: Choose the shortest TB regardless of the length of the space. This strategy will select RBs with lowest interference.

It is sometimes possible to select a more robust MCS while maintaining the same TB size. This idea is described in [11] as *Best Robust MCS* and enabled in the scheduler.

B. Algorithm Description

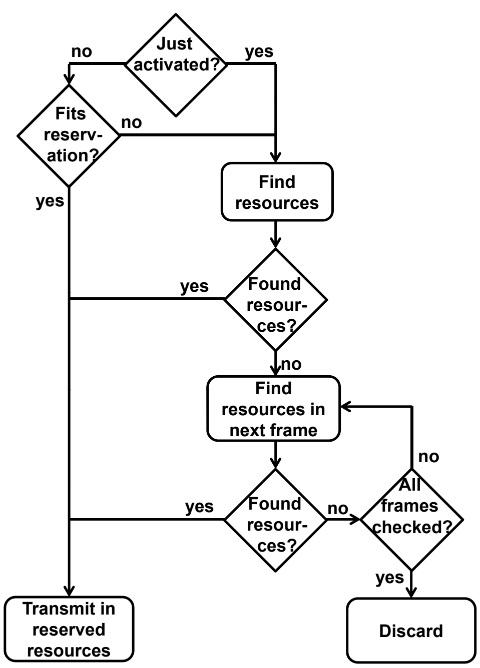

Fig. 3. Scheduling algorithm for active calls.

Figure 3 describes the scheduling algorithm performed for each persistent call during its active phase in each subframe. Calls established earlier are checked if the SINR on the reserved resources is still sufficient for the chosen MCS. The decision is based on measurements from previously received data from this call. The scheduler will try to find a new resource assignment if more RBs are needed since a more robust MCS is needed. This procedure is called a *Frequency Relocation*. This procedure is also performed when a call becomes active the first time or enters active state again:

The scheduler tries to find a sufficient amount of consecutive RBs in the current frame to carry the TB. The TB is selected according to the chosen selection strategy. If the current frame does not have enough free resources, the algorithm is repeated on the next subframe and so on, until all 20 subframes passing during the VoIP IAT period have been checked. This is called *Time Relocation*. If none of the subframes has sufficient free resources, the connection request or *Frequency Relocation* attempt fails and no resources are reserved for this call. At this point the system usually reached or is even beyond its capacity. Data that remains queued after a failed *Time Relocation* may be transmitted later using dynamic scheduling if enough control channel resources are available.

After scheduling the persistent connections, HARQ retransmissions are scheduled. They can either dynamically be served by the best, or any available RBs. Transmitting on the best resources results in better performance. Choosing the first available resources often results in very weak receive quality preventing successful decoding even after multiple retransmissions. Finally SID-PDUs are scheduled using the same TB selection strategy as applied for active calls. If the whole PDU does not fit in the remaining resources it is segmented until it fits. Queued segments are then transmitted in later subframes on free resources. Pending HARQ retransmissions and SID-PDUs are scheduled following a Longest Wait First strategy. Retransmissions are delted if more than 40 ms have passed since the initial transmission attempt.

IV. Simulation Scenario and Results

Results were obtained using the Open Source Wireless Network Simulator (openWNS) [12]. The simulator was previously used to evaluate cell spectral efficiency (CSE) and cell edge user spectral efficiency (CEUSE) for IMT-A evaluation [5] and was validated, analytically, for that purpose [13], [14]. We use the physical layer and channel model implementation *IMTAPhy* described and validated in [15].

A. Simulation Scenario and Assumptions

Traffic- and channel model and system deployment considered here match the ones described in the IMT-A evaluation guideline with some exceptions: An IMT-A compliant channel model implementation is used calculating the complex channel gain on each link for each antenna pair. This way receive filter and multi-antenna gains are included in the results. We do not consider a time-varying channel by setting UT velocity to 0 km/h. Presented results can therefore only serve as upper bound estimation showing the potential gains of the evaluated scheduling strategies. In this work we focus on channel quality variations caused by inter-cell interference. The challenges of voice traffic scheduling in a time varying channel including the limitations of uplink channel sounding and uplink path-loss estimation for power control are beyond the scope of this work and will be evaluated in a separate future publication.

Only seven eNB sites with three sectors each are deployed and wrap-around is used.

Base line assumptions for LTE [4] are used to model control traffic. This includes a 6 ms and 7 ms delay for downlink and uplink channel measurement, respectively. 3 out of 14 symbols are used for the downlink control channel supporting up to 8 uplink and 4 downlink PDCCHs. HARQ with Chase Combining is used. Two RBs are reserved for the Physical Uplink Control Channel (PUCCH) leaving only 23 RBs available for user data. A Maximal Ratio Combining (MRC) receiver is used and the antenna spacing at the eNB is set to ten times the wavelength of the center frequency according to base line assumptions from [4].

The 28 different MCSs described in the LTE standard [1] are used. The block error rates used to determine the minimum effective SINR for each MCS were derived using link-level simulations described in [16]. Figure 4 shows the TB size for voice and SID PDUs versus effective SINR for uplink transmissions. Downlink results are similar but depend on the overhead introduced by each antenna for reference symbols. Uplink power control is not used in the Indoor Hotspot (InH) scenario and all UTs transmit at 4 dBm. Fractional power control with parameters $\alpha = 0.8$ and $P_0 = 0.83$ dBm are used in the Urban Macro scenario, following the assumptions from [4].

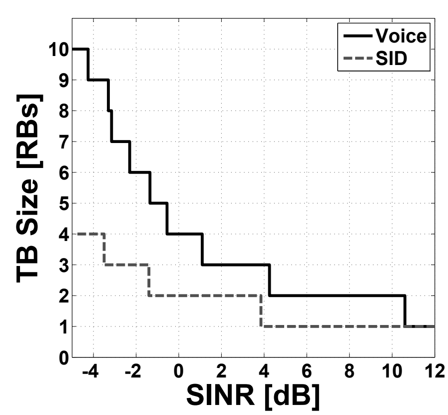

Fig. 4. TB size in RBs over SINR.

In the following, VoIP capacity is evaluated in the InH scenario [2] shown in Figure 5, which is a 50 m x120 m large room with two cells, and cellular Urban Macro (UMa) scenario with 500 m Inter-site distance (ISD). The UMa scenario is chosen because it yields the lowest capacity results according to [5] and [4].

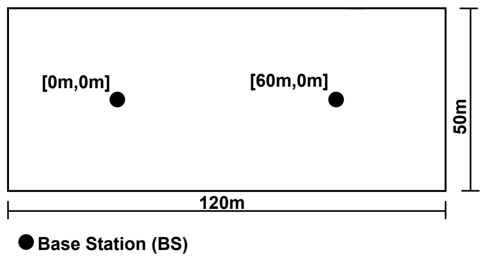

Fig. 5. IMT-Advanced Indoor Hotspot evaluation scenario.

B. Results

An upper VoIP capacity bound can be estimated analytically. There are $25 \cdot 320 = 8000$ RBs in 320 ms with 5 MHz bandwidth. During this time 8 voice and 1 SID PDUs are transmitted on average. In the best case each PDU is transmitted on 1 RB. The upper bound is then $\lfloor 8000$ RBs$/9$ RBs per call$/5$ MHz$\rfloor = 177$ calls/cell/MHz. Simulation assuming that all calls need one RB and an error free channel yields a capacity of 168 calls/cell/MHz. The theoretical capacity is decreased to $\lfloor 8000$ RBs$/17$ RBs per call$/5$ MHz$\rfloor = 94$ calls/cell/MHz if voice and SID PDUs are transmitted on 2 and 1 RBs, respectively. Simulations yield a capacity of 85 calls for this case.

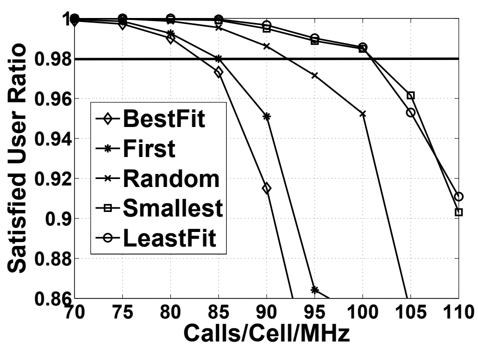

Fig. 6. Indoor Hotspot: Ratio of satisfied uplink users.

Figure 6 shows the ratio of satisfied users versus number of VoIP calls per cell and MHz in the InH scenario. TB selection strategies *BestFit*, *First*, *Random*, *Smallest*, and *LeastFit* are evaluated. Only 83 and 85 users can be served when applying the strategies *BestFit* and *First*, respectively, which do not consider channel quality. This is beyond the IMT-A requirement of 50 calls. The *Random* strategy does also not consider the channel quality for the scheduling decision but increases the capacity to 92 VoIP calls by spreading interference uniformly over the entire frequency range. The strategies *Smallest* and *LeastFit* allow up to 101 calls. Those strategies achieve higher capacities by using less interfered RBs and favoring ones with better channel quality.

As shown in Figure 7 the mean TB size is lowest with the strategy *Smallest*. In Figure 8 the BLER of the initial transmission versus number of VoIP calls is shown for the strategies

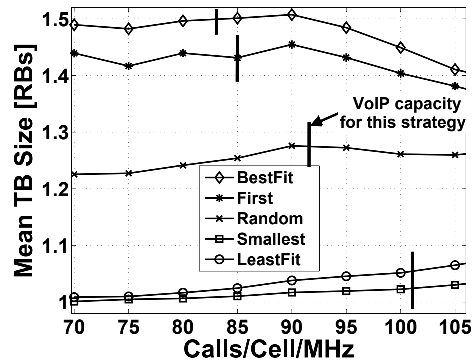

Fig. 7. Indoor Hotspot: Mean TB size.

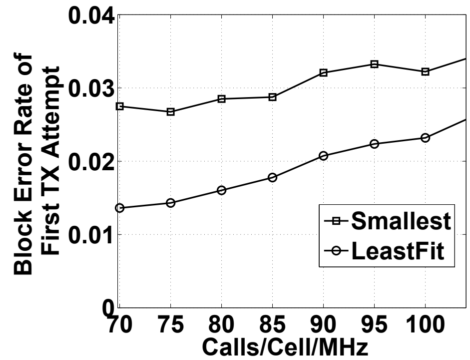

Fig. 8. Indoor Hotspot: Block error rate (BLER)

LeastFit and *Smallest*. The channel quality estimation is too optimistic resulting in more estimation errors for the strategy *Smallest* causing more HARQ retransmissions canceling out the benefit of the lower TB size.

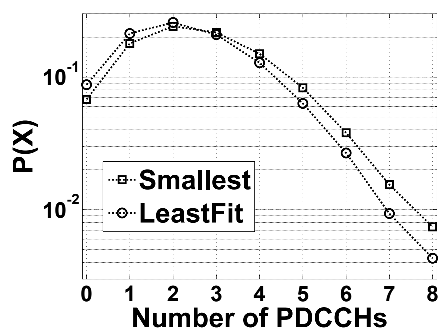

Fig. 9. Indoor Hotspot: Distribution of the number of uplink PDCCHs for 100 calls/cell/MHz.

Figure 9 shows the distribution of the number of PDCCHs used for uplink scheduling for the *LeastFit* and *Smallest* strategies. Results for 100 calls per cell and MHz are presented, where VoIP capacity is almost reached. PDCCHs are transmitted to signal persistent resource allocations when a call enters active state, for frequency relocations and for dynamically scheduled SID and HARQ transmissions. The maximum number of eight PDCCHs is only used for less than 1 % of the subframes. This proves that control channel resourced do not limit VoIP capacity if semi-persistent scheduling is used. The *Smallest* strategy requires more PDCCH resources because more HARQ retransmissions must be scheduled due to its higher BLER.

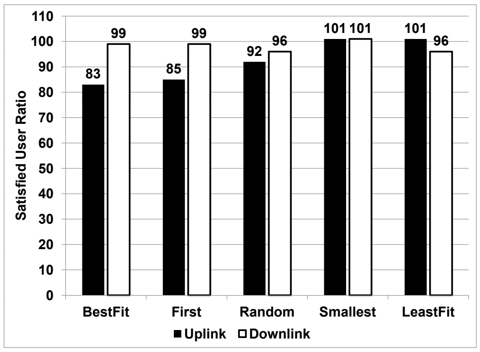

Fig. 10. Indoor Hotspot scenario VoIP capacity.

Figure 10 compares the VoIP capacities for the different strategies and also provides downlink results. All strategies perform almost equally in the downlink because differences in channel quality cannot be exploited with wideband channel measurements. The slightly different performance of the strategies is due to differences in resource fragmentation. Since all downlink results are higher than or almost equal to the uplink results, the uplink is limiting the overall VoIP capacity.

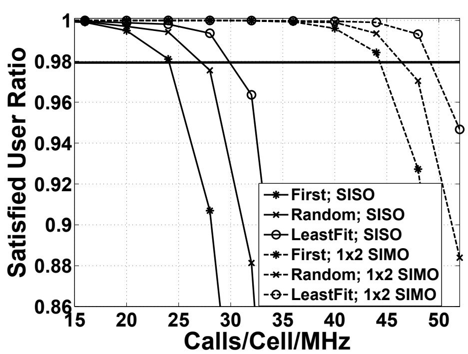

Fig. 11. Urban Macro: Ratio of satisfied uplink users for single- and multi receive antennas.

Figure 11 shows the uplink VoIP capacity for the Urban Macro scenario specified in [2] with one and two receive antennas. Results for *First*, *Random*, and *LeastFit* assignment are shown to compare performance with and without channel aware scheduling. The strategy *First* and *Random* satisfy 24 and 27 calls, respectively with a single receive antenna. VoIP capacity is increased by 25.0 % compared to the *First* strategy to 30 calls if *LeastFit* assignment is used. This is a higher gain than for the InH scenario where capacity was increased by 18.8 % relative to the *First* strategy. Still the IMT-A requirement of 40 VoIP calls per cell and MHz is not met.

VoIP capacity is increased significantly if a second receive antenna is used. 44, 46, and 49 calls can now successfully be served using the *First*, *Random*, and *LeastFit* strategy, respectively. The minimum IMT-A requirements are met. The capacity gain of the *LeastFit* strategy relative to the *First* strategy is decreased to 11.4 %. The reason is the improvement in SINR for many UTs causing more calls to be served requiring minimal TB size of one RB not allowing further improvement.

TABLE I
INDOOR HOTSPOT: VOIP CAPACITY FOR DIFFERENT ANTENNA CONFIGURATIONS.

	Antenna Configuration		
Strategy	1x1	1x2	1x4
First	83	101	109
LeastFit	101	117	122
Capacity gain	18.8 %	15.8 %	11.9 %

Table I compares the VoIP capacity of the *First* and *LeastFit* strategy in the InH scenario for one, two and four receive antennas and one transmit antenna (Single Output). The *LeastFit* strategy increases the capacity by 18.8 %, which is the same gain achieved by a second receive antenna. Using four receive antennas yields a capacity gain of 7.9 % relative to using the *First* strategy in a 1x2 configuration. The *LeastFit* strategy improves the capacity by 11.9 % for two receive antennas. A total VoIP capacity improvement of 44.5 % is reached comparing *First* scheduling strategy SISO to *LeastFit* scheduling strategy 1x4 SIMO performance.

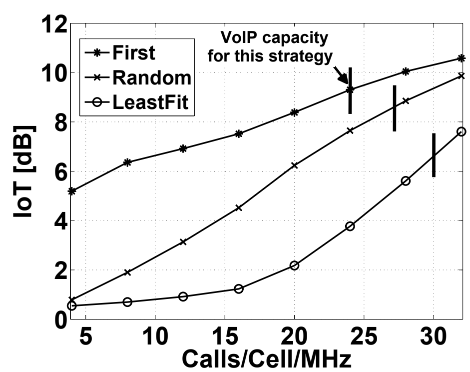

Fig. 12. Urban Macro: Mean Interference over Thermal (IoT) over number of calls for SISO case.

Figure 12 shows the mean Interference over Thermal (IoT) for the single antenna case. It is a measure for the performance of interference mitigation strategies. The interference increases as the number of calls is increased. The highest interference is experienced if the *First* strategy is used caused by the high utilization of the same RBs in all cells. The mean interference power is reduced if transmissions are distributed on the entire spectrum using the *Random* strategy. Interference aware scheduling performed by the *LeastFit* strategy results in lowest received interference power. This also proves that the capacity is not just increased by transmitting on RBs with better quality on the serving link.

V. SUMMARY, CONCLUSION, AND OUTLOOK

Challenges of VoIP scheduling for LTE are described and a suitable resource assignment algorithm for semi-persistent scheduling has been presented. Assigning the best resources rather than any other assignment increases VoIP capacity by almost 19 % from 85 to 101 successful calls per cell and MHz in the Indoor Hotsport scenario and one receive antenna. With 1x2 SIMO almost 16 % and 11.5 % VoIP capacity gain is archieved in the Indoor Hotspot and Urban Macro scenario, respectively.

Advanced transceivers providing higher data rate at low SINR as well as MIMO techniques can significantly increase VoIP performance by assuring a higher percentage of UTs can be served on only one RB. The more UTs are served with smallest possible TB size, the less capacity gains can be achieved by advanced scheduling techniques. Smart scheduling algorithms show their full potential in cases where SINR is low or when simple transceiver hardware is used.

As a next step we want to evaluate VoIP capacity in time-varying channels under realistic channel estimation constrains including downlink Channel Quality Indicator (CQI) and uplink Sounding Reference Symbols (SRS) limitations in time and frequency resolution.

Our future work then focuses on identifying the weakest users aiming to reduce their packet loss ratio. Therefore we distinguish between cell edge and cell centre users and apply appropriate scheduling algorithms for each group.

REFERENCES

[1] "Evolved Universal Terrestrial Radio Access (E-UTRA) and Evolved Universal Terrestrial Radio Access Network (E-UTRAN); Overall description ," 3GPP, Tech. Rep. 3GPP 36.300, September 2009.

[2] "Guidelines for Evaluation of Radio Interface Technologies for IMT-Advanced," ITU-R, Tech. Rep. ITU-R M.2135, November 2008.

[3] D. Jiang, H. Wang, E. Malkamaki, and E. Tuomaala, "Principle and Performance of Semi-Persistent Scheduling for VoIP in LTE System," in *Wireless Communications, Networking and Mobile Computing, 2007. WiCom 2007. International Conference on*, sept. 2007, pp. 2861 –2864.

[4] "Evolved Universal Terrestrial Radio Access (E-UTRA); Further advancements for E-UTRA physical layer aspects," 3GPP, Tech. Rep. 3GPP 36.814, March 2010.

[5] "Final Evaluation Report from WINNER+ on the IMT-Advanced Proposals in Documents IMT-ADV/6, IMT-ADV/8 AND IMT-ADV/9," ITU-R, Tech. Rep. IMT-ADV/22-E, June 2010.

[6] Y. Fan and M. Valkama, "Enhanced VoIP Support in OFDMA-based Packet Radio Networks," *Wireless Personal Communications*, pp. 1–24, 2011.

[7] Z. Wang, Y. Wang, D. Jiang, C. Tian, and D. Yang, "Scheduling and link adaptations for VoIP in TDD-LTE uplink," in *Wireless Communications, Networking and Mobile Computing, 2009. WiCom'09. 5th International Conference on*. IEEE, 2009, pp. 1–5.

[8] F. Capozzi, D. Laselva, F. Frederiksen, J. Wigard, I. Kovacs, and P. Mogensen, "UTRAN LTE Downlink System Performance under Realistic Control Channel Constraints," in *Vehicular Technology Conference Fall (VTC 2009-Fall), 2009 IEEE 70th*, sept. 2009, pp. 1 –5.

[9] Y. Fan, P. Lunden, M. Kuusela, and M. Valkama, "Efficient Semi-Persistent Scheduling for VoIP on EUTRA Downlink," in *Vehicular Technology Conference, 2008. VTC 2008-Fall. IEEE 68th*, sept. 2008, pp. 1 –5.

[10] K. Brueninghaus, D. Astely, T. Salzer, S. Visuri, A. Alexiou, and G. Seraji, "Link Performance Models for System Level Simulations of Broadband Radio Access Systems," in *Personal, Indoor and Mobile Radio Communications, 2005. PIMRC 2005. IEEE 16th International Symposium on*, vol. 4. IEEE, 2005, pp. 2306–2311.

[11] "Mechanisms for increasing the efficiency of MAC/PHY protocols," ICT-ROCKET, Tech. Rep., October 2009.

[12] D. Bültmann, M. Mühleisen, K. Klagges, and M. Schinnenburg, "openWNS - open Wireless Network Simulator," in *15th European Wireless Conference 2009 Electronic Proceedings*. Aalborg, Denmark: VDE Verlag GmbH, May 2009, pp. 205–210.

[13] M. Mühleisen, D. Bültmann, R. Schoenen, and B. Walke, "Analytical Evaluation of an IMT-Advanced Compliant LTE System Level Simulator," in *Proceedings of European Wireless 2011*. Vienna, Austria: VDE Verlag, Apr 2011, p. 5. [Online]. Available: http://www.comnets.rwth-aachen.de

[14] M. Mühleisen and B. Walke, "Analytical evaluation of lte uplink performance in the imt-advanced indoor hotspot scenario," in *Proceedings of PIMRC 2011*, Toronto, Canada, Sep 2011, p. 6. [Online]. Available: http://www.comnets.rwth-aachen.de

[15] J. Ellenbeck, O. Mushtaq, F. Sheikh, and S. Q. Shahrukh, "Calibrating an Efficient C++ Implementation of the ITU-R M.2135 Channel Model for Use in System-Level Simulations," in *Proc. International Workshop on Propagation and Channel Modeling for Next-Generation Wireless Networks (IWPCM 2011)*, Lyon, France, Mar. 2011.

[16] J. Ellenbeck, "System-Level Simulations with the IMT-Advanced Channel Model," in *LTE-A and Next Generation Wireless Networks: Channel Modeling and Propagation* , G. de la Roche, A. Alayon-Glazunov, and B. Allen, Eds. John Wiley and Sons, 2012 (to appear), ch. 14.