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Influence of Traffic Models and Scheduling on the System Capacity of Packet-Switched Mobile Radio Networks

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Abstract— **Accurate knowledge of the system capacity is necessary for planning of packet-switched wireless networks. Although packet services like FTP download or web browsing are not as critical with respect to delay as circuit-switched services, the user expects Quality of Service guarantees to be fulfilled. The assumed traffic model and the guaranteed Quality of Service as well as strategies for scheduling the users have impact on the system capacity. Estimates of the system capacity can be obtained by system level simulations. In this paper, it is shown that the system capacity for FTP traffic with a low subscribed data rate and a high number of users is comparable to the case when each user has always data to transmit and no QoS is considered. This is no longer valid if the number of users is decreased due to a higher subscribed data rate, or if the traffic model has a more bursty characteristic such as web browsing. This leads to a decreased average sector throughput which has to be considered in the network planning process.**

I. Introduction

In the planning process of wireless networks it is important to have an accurate estimate of the system capacity available. The system capacity depends, among others, on the traffic model, the Quality of Service (QoS) requirement and the resource allocation. These properties are different for circuit-switched (CS) and packet-switched (PS) networks [1].

PS networks, e.g., WiMAX and Flash-OFDM, use shared channel concepts with scheduling mechanisms. System capacity can no longer be derived based on blocking criteria as in Circuit Switched (CS) networks because channel allocation to the users is no longer fix. Different QoS requirements and traffic models as well as fairness in scheduling schemes have an important impact on system capacity. In particular, varying channel conditions can be utilised to increase the system performance by adapting the modulation and coding scheme (MCS) or by prioritising a user with good channel conditions during transmission [2, 3].

The focus of this paper is on the investigation of the system capacity of PS networks with Orthogonal Frequency Division Multiple Access (OFDMA) in combination with Time Division Multiple Access (TDMA) as it is used for WiMAX and Flash-OFDM and as it is also a promising candidate for 4G systems [4-6].

The system capacity is measured, e.g., by the sector throughput. An estimate of the system capacity can be obtained by system level simulations [7, 8]. In general, system level simulations require a high computational effort. Often simple traffic models like the Full Buffer Traffic Model (FBM) are assumed in order to reduce the computational complexity [9]. With the FBM, it is assumed that each user has infinite amount of data to transmit and no QoS is considered. In [1, 10], it is shown that this model leads to too optimistic sector throughput results compared to more realistic traffic models.

This paper focuses on the influence of traffic models, e.g. for file download and web browsing, and of different scheduling strategies on the sector throughput. These results are compared to the sector throughput obtained with the FBM. The results obtained in this paper form a basis which can be used in the planning process of PS wireless networks. If system capacity results for the FBM are available, estimates of the system capacity for real traffic models can be deduced based on the results of this paper.

The paper is structured as follows. Section II describes the assumed OFDMA/TDMA system model. Section III presents the recently proposed link to system level interface for the system level simulator which is suitable for OFDMA systems. In section IV, the assumed traffic models are presented. Section V describes the simulation environment. In section VI, the parameters for performance evaluation are given. Section VII shows the results of the investigation. Finally, conclusions are drawn in section VIII.

II. System Model

A broadband wireless access system with OFDMA in combination with TDMA as multiple access scheme is considered. The assumed system is not in detail compliant to any particular standard. However, the investigation and the results are in principle valid for any system of this type.

In the described system, the smallest resource unit (RU) is one subcarrier of one OFDM symbol. A number of RUs are grouped together. The grouping can be done in frequency domain (several subcarriers of the same OFDM symbol) and in time domain (subcarriers of different OFDM symbols). The RUs which belong to one group together are called scheduling unit (SU). A SU may have different dimension in time and in frequency domain, depending on how many RUs are grouped together. Each SU is allocated to one user for data transmission using one specific modulation and coding scheme (MCS) for the entire SU according to link adaptation.

The RUs are mapped onto physical subcarriers according to predefined patterns. To achieve frequency diversity, the subcarriers allocated to RUs of one SU are spread over the whole bandwidth leading to interferer diversity. The patterns are different for different base stations. This is similar to the permutation during the subchannel allocation in WiMAX or the usage of frequency hopping [5, 6, 11].

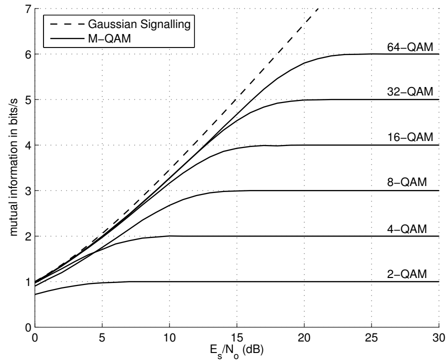

Fig. 1. Mutual information versus E_s/N_0 for QAM modulation

III. Link to System Level Interface

In system level simulations, the performance of the whole cellular system is investigated. Therefore, it would be too complex to model each single link in detail. Nevertheless, it is important to make a decision whether a transmitted packet is received successfully or not. A link-to-system level interface is needed to map signal to interference plus noise ratio (SINR) values calculated on system level onto results from link level simulations, e.g. block error probabilities. In [12, 13], a method is described how to get an estimate of the block error probability during transmission which is well-suited for OFDM based systems. The method is briefly described in the following.

During transmission, each subcarrier experiences a different SINR. The signal and the interference powers depend on pathloss, lognormal and fast fading which are modelled on system level. Interference occurs if the same subcarrier is used simultaneously in different cells. Receiver noise is modelled as Gaussian noise with constant power.

To estimate the performance of a single subcarrier, the mutual information is calculated for each subcarrier. Assuming Gaussian noise, with X the input signal from a given constellation and $Y = X + N$ the output signal over the AWGN channel, the mutual information is expressed by [14]

$$I(X,Y) = h(Y) - h(Y|X) \tag{1}$$

with $h(\cdot) = -\mathrm{E}[\log_2(p(\cdot))]$ the entropy function. A solution for higher order modulation schemes is complex and can be determined, e.g., by Monte-Carlo integration. Figure 1 shows the mutual information for QAM modulation versus the ratio E_S/N_0 of the energy per symbol and the noise power spectral density assuming equally probable input constellation points.

In the receiver of the system, the bits on the subcarriers are not considered separately due to one cyclic redundancy check for the SU. The whole SU is either correctly received or discarded in total. Furthermore, during encoding each information bit is distributed over the SU and only one particular MCS is used for the whole SU. To get one error decision for the SU, all subcarrier have to be considered together. Therefore, the mutual information of all subcarriers belonging to one SU is summed. This leads to an estimate of the performance of the complete SU.

A simple model has to be found for link level simulations which results in the same SU performance. Due to subcarrier permutation and coding over the whole SU, it is assumed that the SU behaves as if the mutual information is uniformly distributed over the subcarriers of the SU. Based on the average mutual information, an effective signal to noise ratio (SNR) is calculated for the SU. In the equivalent model, each subcarrier is assumed to have this effective SNR. Furthermore it is assumed that the noise is Gaussian distributed corresponding to link level simulations with AWGN channel. Based on this equivalent model, a decision is made whether the SU is correctly received or not. Therefore, link level results are necessary showing the block error probability versus SNR for transmission over AWGN channel for each MCS.

IV. Traffic Model

The characteristic of data traffic is bursty. There are periods with high amount of data and periods with low amount of data or even no data to be transmitted. Realistic system performance investigations should consider this behaviour. Therefore, traffic models have been defined, e.g. in [13, 15], generating bursty data traffic. In this section, the properties of the considered traffic models for FTP and HTTP are described.

In the traffic models, sessions are defined. Each session consists of packets. The start of a session is modelled by a Poisson process. In general, the rate for transmitted packets is adjusted by TCP depending on the throughput of the transmission, e.g., TCP slow start behaviour and TCP congestion control. These effects are not considered in this investigation. Rather, a subscribed data rate D_{subscr} is defined for each session. With S_{packet} the average packet size in bits of the traffic model, the average interarrival time between two consecutive packets is calculated by

$$T_{\mathrm{interarr}} = \frac{S_{\mathrm{packet}}}{D_{\mathrm{subscr}}}. \tag{2}$$

1) FTP Traffic Model: The FTP traffic model is taken from [13]. Each session consist of one file download. The size of the file is lognormally distributed with average value $S_{\mathrm{file,avg}}$.

The file is segmented into TCP packets with constant size $S_{\mathrm{packet}} = 1500$ byte. T_{interarr} is calculated according to Eq. (2). The probability distributions and parameters assumed in the investigation are given in [13].

2) HTTP Traffic Model: The HTTP traffic model is described in [15]. A session contains several Packet Calls. Between two consecutive Packet Calls there is an interval when no data is transmitted. This interval is called Reading Time. Each Packet Call consists of packets. The average number of Packet Calls per session is N_{pc}, the average Reading Time is T_{RT} and the average number of packets per Packet Call N_{p}. T_{interarr} is calculated according to Eq. (2). The parameters assumed in the investigation are given in [16].

V. Simulation Environment

Throughout this paper, an infrastructure based cell environment with reuse 1 is investigated. Each site contains three sectors with directional antennas. Two tiers of interfering cells are considered. It is assumed that the system is interference limited. Therefore, the site-to-site distance has no influence on the signal-to-interference ratio. The parameter settings of the simulation environment are given in Table I.

TABLE I
PARAMETER SETTINGS OF THE SIMULATION ENVIRONMENT

Parameter	Value
Number of sites	9 wrapped around on torus, 3 sectors per site, hexagonal deployment
Pathloss Model	Okumura-Hata
Slow fading	lognormal distributed, standard deviation 8 dB
Fading Channel Model	ITU Vehicular A
Base Station Antenna	directional, 30 m above ground
Mobile Station Antenna	omni, 1.5 m above ground
User positioning	uniform distribution, new positions for each snapshot
Mobile Station assignment	best server (incl. shadowing)
User Scheduling	Fair Resource (FR) or Proportional Fair (PF)
Snapshot Duration	10 sec

The system level simulator is based on a snapshot approach which combines properties of static and dynamic system level simulators. The duration of a snapshot is short enough to be able to assume constant pathloss and shadowing conditions for each user during the whole snapshot. Time variant fast fading channel conditions are considered within the snapshot.

It is important to know the system capacity for the traffic occurring in the busy hour because network planning is done for this situtation. In the busy hour, there are on average N_{BH} sessions active. The average session duration for FTP and HTTP sessions can be calculated by

$$T_{\mathrm{avg,FTP}} = \frac{S_{\mathrm{file,avg}}}{D_{\mathrm{subscr}}}, \tag{3}$$

$$T_{\mathrm{avg,HTTP}} = N_{\mathrm{pc}} \cdot N_{\mathrm{p}} \cdot T_{\mathrm{interarr}} + (N_{\mathrm{pc}} - 1) T_{\mathrm{RT}}. \tag{4}$$

The average number of sessions which are active at a given time instant is calculated by

$$N_{\mathrm{active,common}} = \frac{N_{\mathrm{BH}} \cdot T_{\mathrm{avg}}}{3600\ \mathrm{s}} \tag{5}$$

with T_{avg} the average duration for FTP and HTTP session as given by Eq. (3) and (4) respectively. If only a snapshot is considered, a model has to be found how to consider only a part of the session during the snapshot. There are sessions which are already running at the beginning of the snapshot. Furthermore, it is possible that new sessions start and that sessions terminate during the snapshot. The average number of active sessions at the beginning of the snapshot is given by Eq. (5). The probability of being in a certain state of the session at the beginning of the snapshot is calculated by the average duration of this state divided by the total duration of the session. For instance, the probability for being in the first reading time period of a web browsing session at the beginning of the snapshot is calculated by

$$P_{\mathrm{1stRT}} = \frac{T_{\mathrm{RT}}}{N_{\mathrm{pc}} \cdot N_{\mathrm{p}} T_{\mathrm{i}} + (N_{\mathrm{pc}} - 1) \cdot T_{\mathrm{RT}}}. \tag{6}$$

Therefore, the state at the beginning of the snapshot can be derived for each session. With the snapshot duration T_{snapshot}, the average number of starting session per snapshot is calculated by

$$N_{\mathrm{starting\ sessions}} = \frac{N_{\mathrm{BH}} \cdot T_{\mathrm{snapshot}}}{3600\ \mathrm{s}}. \tag{7}$$

VI. Performance Evaluation

The performance of the assumed broadband wireless access system is measured by sector and user throughput. The system capacity for a certain service type is reached if 5 % of the users do not achieve the QoS requirement for that service.

If $A_{\mathrm{sector,sum}}$ is equal to the amount of data transmitted without errors by a base station in one sector within the interval T_{snapshot}, the sector throughput is defined by

$$D_{\mathrm{sector}} = \frac{A_{\mathrm{sector,sum}}}{T_{\mathrm{snapshot}}}. \tag{8}$$

If $A_{\mathrm{user,sum}}$ is the amount of data a user transmits during the snapshot and T_{active} is the duration the user has data to transmit during the snapshot, the active session throughput of each user is calculated by

$$D_{\mathrm{user}} = \frac{A_{\mathrm{user,sum}}}{T_{\mathrm{active}}}. \tag{9}$$

In the FBM, each user is active during the whole snapshot ($T_{\mathrm{active}} = T_{\mathrm{snapshot}}$). In case of realistic traffic models, it is possible that a user has no data pending for transmission during time intervals of the snapshot. These intervals are excluded from the calculation of the active session throughput ($T_{\mathrm{active}} \leq T_{\mathrm{snapshot}}$).

In the following, the QoS requirement shall be derived. QoS requirements related to delay, delay variation and packet error rate are only considered in case of the realistic traffic models. In the investigation, delay is not measured. Instead, another model is assumed which is described in the following. A constant average arrival rate of new packets is assumed. Therefore, delay variation or packet error lead to varying user throughput. Short time variations are compensated by jitter buffers or tolerated by the user. The duration for which the user throughput may fall below the limit depends on the sensitivity of the service to delay. The interval is short for delay sensitive services and long for services which are not sensitive with respect to delay. The proposed model assumes that each user expects to achieve 10 % of the subscribed data rate as user throughput during a specific time interval. It is assumed that HTTP is more sensitive to delay than FTP traffic. The user throughput is measured continuously with a sliding window in periods of 1 s for HTTP and 5 s for FTP traffic. If the user throughput falls below the threshold of 10 % during a time interval of 1 s or 5 s, respectively, it is assumed that the QoS for this user is not reached. The consequence is that the user gets unsatisfied. In the investigation, even unsatisfied users remain in the system. There is no dropping considered, which leads to worst case system capacity values. It is assumed that the maximum sector throughput representing the system capacity is reached if 5 % of the user are unsatisfied.

VII. Simulation Results

This section presents the results of the investigation for transmission with the described traffic models obtained from system level simulations. Packet errors are modelled according to the described link-to-system level interface. During scheduling and link adaptation, a perfect knowledge of the actual SINR conditions is assumed. For link adaptation 12 different MCS are possible using QPSK, 16 QAM, 64 QAM and 256 QAM with coding rates between $1/6$ and $3/4$. A MCS is selected if a block error probability of less than 1 % can be guaranteed. The results show the change in the average, 10 %-ile and the cumulative distribution function (cdf) of the sector and the user throughput. In the following the FBM, HTTP and FTP model are considered. If the FTP model is used, it is distinguished between traffic with low subscribed data rate (LDR) and high subscribed data rate (HDR) using 100 kbit/s and 1 Mbit/s, respectively. The system capacity obtained when using a traffic model is compared to the system capacity obtained with the FBM. Here it is also distinguished between FBM with high number of users (HNU FBM) and low number of users (LNU FBM). The HNU FBM is used for comparison with LDR FTP and HTTP. In this case, the number of users in the FBM is equal to the number of users for LDR FTP. The LNU FBM is used for comparison with HDR FTP if only a small number of users is considered in the system.

In the following the results of the investigation are presented. For comparison reason the ratio of the achieved throughput when a more realistic traffic model is assumed and of the throughput with FBM is calculated. This is termed relative sector and relative user throughput in the following. The achieved throughput with HTTP and LDR FTP is divided by the throughput achieved with HNU FBM and the throughput achieved with HDR FTP is divided by the throughput achieved with LNU FBM.

The results for Fair Resource (FR) and different traffic models can be found in Table II. It can be seen that the sector throughput decreases if a realistic traffic model is assumed. While the decrease of sector throughput with LDR FTP compared to HNU FBM is around 20 %, with HTTP traffic the sector throughput is only one third compared to HNU FBM. Due to the reading time there are periods when only little amount of data can be transmitted.

The user throughput for FTP traffic is marginally higher compared to the user throughput achieved with HNU FBM. Due to starting and ending sessions there are more resources available for sessions remaining active than in the FBM when the users are active during the whole snapshot. In the HTTP model the average user throughput is 10 times higher than with HNU FBM due to the reading time periods. In these periods resources are not used by the session and may be allocated to other users.

Furthermore, it can be seen that only a small number of users per sector can be accepted for HDR FTP traffic if 5 % unsatisfied users should not be exceeded.

TABLE II

Performance results for Fair Resource Scheduling

Traffic Model		LDR FTP	HDR FTP	HTTP
Subscribed Data Rate [Mbps]		0.1	1.0	1.0
avg. Number of Active Users		20	1	8
Percentage of unsatisfied users [%]		5.3	4.5	5.3
avg.	relative sector	82.0	55.2	30.2
10 %-ile	throughput [%]	86.8	0	19.5
avg.	relative user	115	103	1046
10 %-ile	throughput [%]	113	139	912

Figure 2 shows the cumulative distribution function (cdf) of the sector throughput if FR scheduling is applied for different traffic models. The slope of the sector throughput cdf for LDR FTP, HTTP and HNU FBM is similar although the curves are shifted according to the performance observed in table II. It can be seen that with HDR FTP traffic, around 20 % of the sectors are not utilised due to low number of users. However there are also some sectors which achieve very high sector throughput. These sectors benefit from users with good channel conditions.

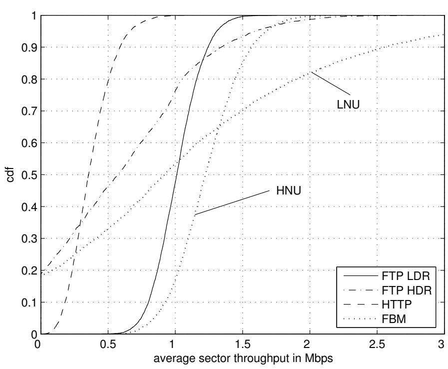

Fig. 2. Sector throughput distribution with Fair Resource Scheduling

Figure 3 shows the cdf of the average user throughput achieved during the snapshot. It can be seen that the throughput for LDR FTP and HNU FBM are almost the same. HTTP users are able to achieve a higher throughput due to the reading time when the users become inactive. This leads to a lower number of users transmitting at the same time so that each user may utilise a higher amount of resources when transmitting data and therefore achieving a higher throughput.

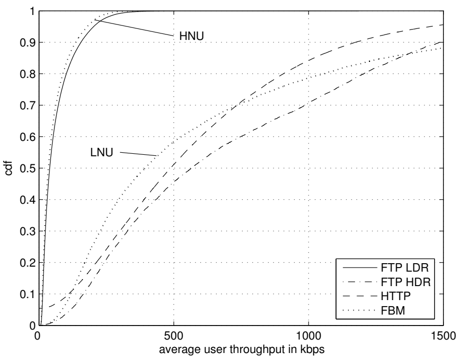

Fig. 3. User throughput distribution with Fair Resource Scheduling

Table III presents the results in case of Proportional Fair (PF) for the different traffic models. Again a degradation is observed for FTP and HTTP compared to the FBM. Figure 4 shows that the sector throughput increases up to 55 % with PF compared to FR. The slope of the sector throughput cdf is almost the same as already observed in Figure 2 for the different traffic models.

TABLE III

PERFORMANCE RESULTS FOR PROPORTIONAL FAIR SCHEDULING

Traffic Model		LDR FTP	HDR FTP	HTTP
Subscribed Data Rate [Mbps]		0.1	1.0	1.0
avg. Number of Active Users		28	1	8
Percentage of unsatisfied users [%]		5.0	5.6	5.3
avg.	relative sector	85.0	52.7	19.6
10 %-ile	throughput [%]	84.6	0	11.2
avg.	relative user	149	96.1	802
10 %-ile	throughput [%]	104	161	602

VIII. CONCLUSION

A system level simulation approach including modelling of packet traffic is derived in this paper which is used for investigation of the system capacity of broadband wireless systems. The system capacity is measured for different traffic models and compared to the FBM. It is shown that the FBM leads to too optimistic results for system capacity in terms of average sector throughput. Nevertheless this model is comparable to a high number of FTP users with a small subscribed data rate. The system capacity is decreased compared to the FBM scenario due to not utilised resources during the reading time period with HTTP traffic and a low number of active users per sector for FTP with high subscribed data rate. The results which are provided in this paper help to improve the network

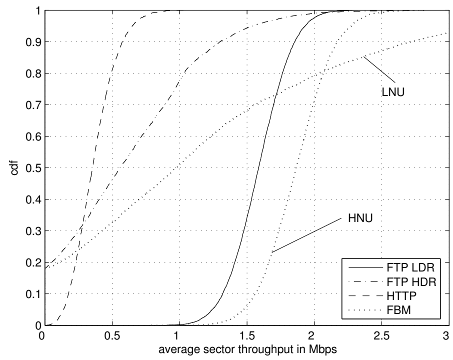

Fig. 4. Sector throughput distribution with Proportional Fair Scheduling

planning process. System capacity estimates for realistic traffic models can be derived if FBM results are available.

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