Capacity of Broadband CDMA Wireless Local Loop Systems

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Abstract—**Wireless Local Loop (WLL) is an emerging technology that allows rapid connection to the wired network from remote locations. A crucial issue for WLL is the system design so that the number of subscribers can be maximized while providing the required quality of service.**

In this paper, we consider a broadband wireless local loop with fixed point users and we evaluate the subscriber capacity for a slotted CDMA scheme which accommodates integrated voice, data, and video traffic. The impact of the system parameters on the WLL performance is studied via simulation and useful insight on how to efficiently design the network is obtained.

I. INTRODUCTION

As the demand for multimedia services and wireless access to the Internet intensifies, an extremely important issue that must be addressed concerns wireless admission control strategies which maintain Quality-of-Service (QoS) guarantees. Several admission control and multiple access protocols for slotted CDMA wireless systems have been proposed in the literature [1], [2], [3], [4], [5], [6], [7]; however, few works deal with capacity evaluation of broadband wireless systems.

In this work, a slotted CDMA packet access technique is studied to estimate the subscriber capacity in a broadband wireless local loop environment in the presence of integrated multimedia traffic. Wireless local loops (WLLs) use the radio channel to perform the "last mile" connection between the wired network and stationary or fixed users [6], [8], [9]. Since WLLs are expected to provide multimedia services with high quality of service, it is of crucial importance to estimate the capacity of such networks in terms of the number of simultaneous connections the system can support.

The objective of this work is an investigation on how to design a wireless network for radio access to fixed points, and how to manage it. Questions to be addressed include the estimation of the amount of traffic which can be served in a specific setting (e.g., under some assumptions on the user locations and on the QoS requirements), and the performance evaluation of a slotted CDMA scheme as the system parameters vary.

The fact that in the fixed-point WLL system considered users do not move has some implications on the analysis and design of such a system. In particular, fixed or slowly varying channel conditions may result in persistent impairment but also allow very accurate power control, also compensating for multipath effects. From the performance analysis point of view, the study of such a system may be significantly harder than for a mobile system. In fact, due to the decreased randomness of the environment, standard approximation techniques for the derivation of the signal to interference ratio in CDMA systems do not apply. Nevertheless, we have been using an approach which allows analytical developments at the physical layer while not conceding too much in terms of simplifying assumptions, with the objective of having an accurate and flexible tool for the evaluation of the physical layer performance. The physical layer model is used in a network layer simulator which incorporates the actual system operation and radio resource management.

The remainder of this paper is organized as follows. In section II we introduce the system scenario that has been used for this study, section III presents the voice and CBR video traffic model and section IV shows some results. The system model used to handle voice with silence activity detection and data traffic is introduced in section V, while the performances obtained for these classes of traffic are presented in section VI. Finally, section VII concludes the paper.

II. SYSTEM DESCRIPTION

A key enabler of a meaningful study is a detailed model of the physical layer. A thorough characterization of it (including frequency selective fading, shadowing, path loss, best station assignment, matched filtering and coding) results in computationally intensive efforts. In our study, the physical layer performance is the starting point for the evaluation of a whole system. Therefore, it is very important to find accurate analytical approximations to provide a faster way to do physical layer numerical analysis. Our approach combines analytical expressions with simulation. It is clear that a completely analytical study is infeasible, but our pseudo-analytical approach couples reasonable efficiency to good modeling accuracy.

More specifically, the following physical layer assumptions are made. The modeled propagation characteristics include path loss (following an inverse fourth-power decay with distance), log-normal shadowing, and frequency selective

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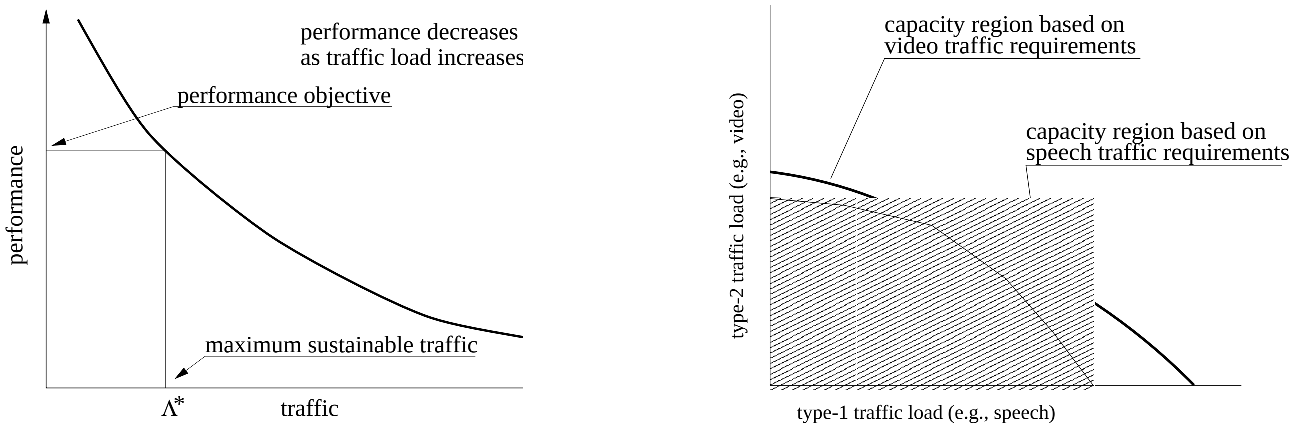

Fig. 1. Example of equivalent capacity curve for one and two classes of traffic. Some generic performance metric is plotted in terms of Erlang load in the network. A performance objective can be translated into a design decision.

Rayleigh multipath fading. Since we assume a fixed-point service, all propagation characteristics remain constant on a time scale larger than the time scale on which connections are established and torn down.

A packet is assumed to be correctly received if its Signal-to-Interference Ratio (SIR) exceeds a certain threshold, chosen according to some minimum BER requirement. In CDMA, perfect RAKE reception (i.e., perfect channel estimation) is considered. The use of an error correction code is assumed, which has the effect of reducing the processing gain of the spread spectrum modulation while decreasing the SIR threshold. The benefits of code orthogonality have been taken into account. The SIR expression has been derived by extending the formula presented in [10] to include into the interference term the effect of the chips that precede and follow the user's code. In this way, the formula holds also for a large delay spread of the channel. (Details of the derivation can be found in [11].)

The simulated scenario consists of 64 radio cells arranged in a uniform hexagonal grid. The whole structure is wrapped onto itself to avoid border effects. The available bandwidth is equal to 9.6 MHz, and a BPSK modulation is adopted. A number of buildings are uniformly scattered throughout the service area. Each building generates a certain traffic load, which is the aggregate traffic of all users in that building.

Buildings are assigned to the base station which provides the best channel conditions. Note that this may lead to dishomogeneous load in different cells, as described later. According to the signal strength at the receiver, each building determines the power to be transmitted, in such a way that at the base station the strength of the intended signal contribution is a constant (the same for all users at their own base station receiver). We applied this simple power control scheme on the uplink and no power control on the downlink. In fact, more complex solutions as those proposed in [7], [12] present an excessive computational complexity that would translate into exceedingly long simulations.

A great enhancement to the system performance is obtained when cell sectorization and availability of directive antennas at the remote user are considered. In this case, we created a new spatial channel propagation model to represent the effect of cell sectorization. We assumed that all the sectors that are located inside this region receive the whole power of the signal transmitted by the remote terminal, while the sectors located outside receive a signal power scaled down by a factor taken as a varying parameter in our model.

A. *Multiple Access*

In classic CDMA systems, channel division is only achieved through spread spectrum, i.e., an active user will continuously transmit, taking full advantage of the bandwidth spreading given by the ratio between the channel symbol rate and the user's own information rate. A possible alternative is to have a framed structure, composed of a number of slots, so that a user may not transmit in all of them. One simple example of this scheme can be a frame with two CDMA slots, with each user being allowed (under normal conditions) to access only one of them. The processing gain available to the user would be in this case half of what it could have been, but the average number of users simultaneously transmitting is also halved, providing roughly the same BER performance.

This slotted structure provides additional flexibility in terms of coexistence of traffic classes with different data rates. Also, opportunities for retransmission of packets whose SIR value is below the threshold could be provided, which is impossible in pure CDMA since users are already transmitting continuously. This feature has great significance when data traffic is incorporated in the study.

B. *Type of Generated Results*

Let us consider a single traffic class in isolation for the sake of discussion. Once a given set of parameter values has been chosen, it is possible to produce a curve like the one in the left plot of Fig. 1, where some generic performance metric is plotted as a function of the traffic load. Based on the specified objective, it is possible to determine the capacity region of the system, which is in this case a single number, Λ^*: any amount of traffic below that value will be served with the specified QoS, whereas any amount beyond that value will overload the network and violate the QoS specifications, e.g., it will result in a blocking probability larger than the objective.

When two classes of traffic are present at the same time, one more axis must be added, so that QoS is represented by a surface in the three-dimensional space. The contour of such surface for a given QoS produces a curve, which represents the two-dimensional capacity region of the system. That is, any combination of traffic loads which lies below the curve will produce adequate QoS, whereas any combination lying above the curve will violate the QoS specification. Since two classes of traffic are present, two such regions must be found, based on QoS objectives for the two classes (which need not be the same). The actual capacity region of the system will then be the intersection of the two, as illustrated by the shaded area in the right plot of Fig. 1.

III. Voice and CBR Video Traffic

In the first stage of the study, we focused on two classes of traffic, which exemplify speech and video connections. These two classes coexist in the system and share the available bandwidth.

The Quality-of-Service (QoS) metric at the network level is the probability that a connection request is blocked (not admitted). Both speech and video connections are taken as Constant-Bit-Rate (CBR) connections, i.e., once admitted they produce a stream of equally spaced packets. In this case, the system is similar to circuit-switching since admitted connections will be persistently using the assigned resources.

We assume that connections are statically assigned to a given slot upon arrival. Slots are assigned to connections according to interference measurements; if no slot is found such that after allocating the new connection an acceptable interference level is maintained for all connections, the new connection is blocked. The choice of static assignment is suboptimal, since a better interference averaging may be expected with dynamic reassignment of slots frame by frame. The major reasons for choosing static assignment are simplicity of implementation and the type of traffic assumed. It is expected that a major benefit of dynamic scheduling will accrue if retransmissions of corrupted packets are allowed.

Given the previous assumptions, it is possible, at least in principle, to predict whether or not the admission of a new connection will impair any of the existing connections in the system, and also whether or not the new connection itself can achieve the desired performance. If both these conditions are verified, then a connection should be admitted, whereas if either of them should be violated, the connection should be rejected. In this case, the dropping probability is zero by definition (The dropping probability is the probability that a packet is not delivered correctly. Here, since no retransmission is implemented, it is equal to the probability of packet error, i.e., the probability that the SIR threshold is not met). Note that the propagation parameters do not change during the lifetime of a connection, users do not move, and arriving connections are subject to the admission control test, so that QoS guarantees made at admission time hold true throughout the connection. The key parameter to be considered is therefore the blocking probability. Once a value of the blocking probability has been defined as the performance objective, it is possible to determine the capacity region of the system.

IV. Simulations for Voice and Video Traffic

For both speech and video connections call arrivals and departures are generated according to the Poisson arrival and exponential duration model with appropriate parameters. At each call arrival, the admission control test is performed to check whether admitting the connection violates QoS of any connections. Based on this QoS test the connection is either admitted or rejected. The statistics of the blocked connections is then computed over the whole network, thereby averaging over all buildings in all cells. For each pair of traffic loads in Erlangs (Λ_s for speech and Λ_v for video) the blocking probabilities experienced by the two traffic classes can be determined.

We considered a limited number of different scenarios, with independent user distributions and assignments. We first run a set of simulations in which the buildings were uniformly distributed in the service area with an average number of buildings per cell equal to 10. However, the actual number of buildings connected to each base station is a random variable due to random propagation conditions. We noticed that cells with a larger number of buildings connected to the base station have worse performance and also single disadvantaged buildings can bias the average results, so that careful planning is necessary in order to avoid bad locations. In order to obtain a fair scenario we run another set of simulations where exactly 10 buildings are assigned to each base station. The algorithm used to reach this goal is as follows:

- step 1: While assigning buildings to the base station, if the number of buildings connected to that base station exceeds 10, we identify the building belonging to that cell with the harshest shadowing and path loss;
- step 2: We generate a new instance of the multipath fading for this building, that corresponds to re-engineering extremely disadvantaged sites;
- step 3: If the new propagation conditions determine the assignment of the building to another base station, we keep on with the normal assignment procedure; otherwise steps 2 and 3 are repeated at most k times (namely k=8);
- step 4: If we fail k times in assigning the building to another base station, the new building is denied service and replaced with another one.

The baseline parameters considered are as follows: channel symbol rate $R_c = 9.6$ Mcps, speech information rate $R_s = 32$ Kbps, video information rate $R_v = 384$ Kbps, error correction code rate 1/2, delay spread 0.5 μs, corresponding to $L = 5$ channel symbols, SIR threshold corresponding to a bit error rate after decoding of 10^{-3}.

The considered objective values of connection blocking probability are in the range 10^{-3} to 10^{-2}.

The ratio between channel rate and information rate for video connections is $R_c/R_v \simeq 25$, and we keep that as the spreading gain of the spread spectrum modulation. However, after encoding the bit rate produced by a video source is doubled. In order to avoid having a too low processing gain, we keep it at 25, but on average we have two video packets per slot (this can be done by using two codes in parallel). With a processing gain of 25, an encoded voice connection produces a chip rate of 32 Kbps $\times 2 \times 25 = 1.6$ Mcps, i.e., about 1/6 of the available bandwidth. We therefore envision a slotted CDMA system with six time slots per frame, where in each frame we have 2 packets per time slot per video connection and one packet in one of the six time slots per voice call (possibly less if speech activity detection is used). The actual length of the frame is 20 ms, so that each voice packet carries 20 ms of speech, whereas each video packet carries $20/12 \simeq 1.67$ ms of video.

The traffic parameters that we used are summarized in Table I, while the main testbed simulation parameters are listed in Table II.

A. Results for CBR Voice and Video

The two-dimensional capacity regions for CBR are plotted in Fig. 2[1]. Plots are for the uplink direction, single antenna reception, and for the elaborate admission control rule based on interference tests. The four curves plotted are the capacity regions corresponding to blocking probabilities of 10^{-3} and 10^{-2} for speech and video. It is seen that the capacity regions corresponding to 0.01 blocking for video and 0.001 blocking for speech are close, giving a balanced design. On the other hand, if we wanted 0.001 blocking for both traffic classes, then video would impose a much stricter requirement, so that one would probably have to implement some mechanism to give it higher priority in order to avoid inefficient situations[2].

It is interesting to see that the general behavior of the capacity region boundaries can be approximated by straight lines in most cases, so that a good indication of the capacity region could be determined by considering the two classes in isolation, and by joining the two points so obtained.

Note that blocking requirements of 0.001 and 0.01 for speech and video, respectively, allow admission of an equivalent Erlang load Λ referred to speech traffic (i.e., computed by summing speech load and $12\times$video load[3]) between 30 and 50, leading to an average load of about $\Lambda \times 32$ kbps$/10$ Mcps$\simeq 10 - 15\%$, which is consistent with previously reported results for cellular CDMA.

Another interesting point is the large increase in the capacity obtained by relaxing the speech blocking objective from 0.001 to 0.01 (note, however, that this requires the objective for video to be relaxed accordingly).

It is worth investigating in some more detail the effect of worst-case buildings. We observed that the results in Fig. 2

[1] The results were obtained by running simulations as long as required to observe $2 \cdot 10^5$ connections.

[2] It can be argued that if a traffic class enjoys a QoS which is much better than required, some inefficiency is present in the system design.

[3] Note however that the results show that one Erlang of video traffic weighs a little more that 12 Erlang of speech traffic, due to differences in granularity and multiplexing efficiency.

TABLE I

TRAFFIC PARAMETERS USED FOR CBR TRAFFIC SOURCES

Parameter	Value
Frame Duration	20 ms
Speech Model	CBR
R_s	32 Kbps
Speech BER	10^{-3}
Video Model	CBR
R_v	384 Kbps
Video BER	10^{-3}

TABLE II

SIMULATION TESTBED

Bandwidth	9.6 MHz
Modulation	BPSK
Spreading Gain	25 or 50
Time Slots Per Frame	6 or 3
Channel Delay Spread	0.5 μs, 1 μs
Number of Cells	64
Buildings Per Cell	10 or 15
Sectors per Cell	1 or 3

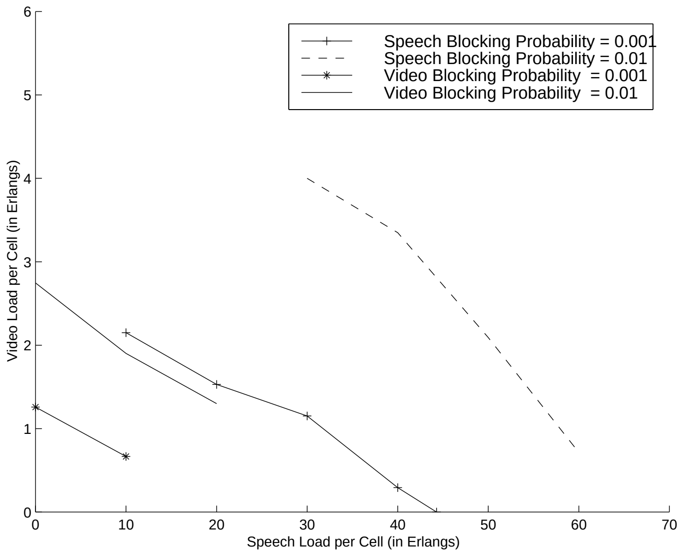

Fig. 2. Uplink capacity region for CBR traffic with buildings uniformly distributed.

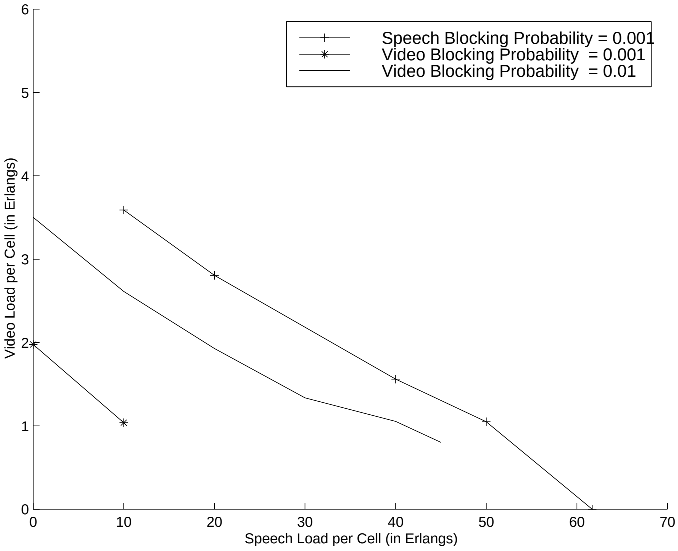

Fig. 3. Uplink capacity region for CBR traffic with a constant number of buildings per cell equal to 10.

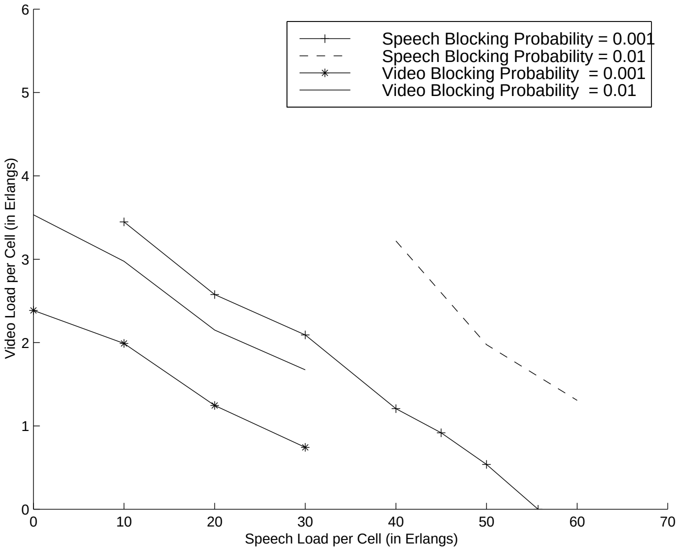

Fig. 4. Downlink capacity region for CBR traffic with buildings uniformly distributed.

suffer from a congested situation since the number of buildings attached to each cell site may vary significantly. It is clear that the cell with a larger number of building will operate in overload and will contribute a large number of blocked connections, thereby biasing negatively the overall performance. Fig. 3 shows the results for the scenario with a constant number of buildings per cell (namely equal to 10). It is seen that the overall performance is significantly improved.

Fig. 4 shows the downlink capacity curves for a scenario with buildings uniformly distributed all over the service area. As already mentioned, we assume no power control here. Precise modeling of the dynamic schemes used in practice seems

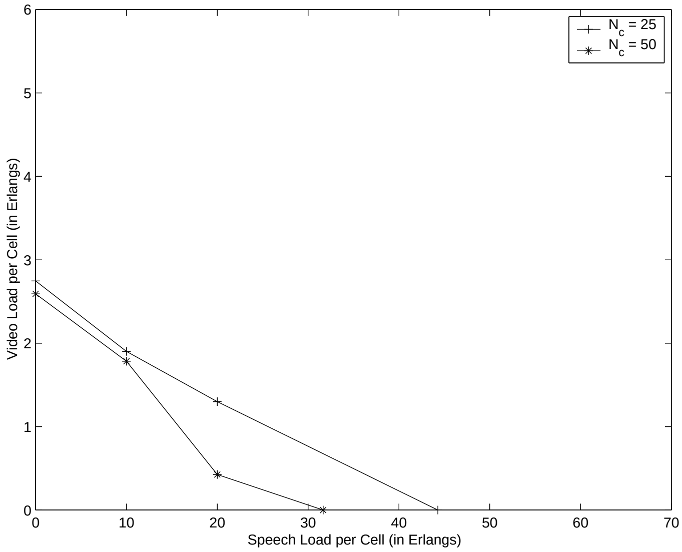

Fig. 5. Uplink capacity region for CBR traffic, buildings uniformly distributed, and spreading gain (N_c) equal to 25 and 50. (QoS metrics: connection blocking probability equal to 10^{-3} and 10^{-2} for voice and video traffic, respectively.)

to lead to exceedingly long simulations, due to the order of magnitude differences between the time scale involved in the power control algorithm and that of the arrival/departure process. The issue is currently under investigation.

As to the issue of code orthogonality, we note the following. In a perfectly flat channel, the symbol shape is unchanged during propagation, so that use of orthogonal codes on the downlink leads to zero intracell interference, thereby greatly improving the performance. On the other hand, if the channel is frequency selective, orthogonality is not preserved at the receiver, and residual interference is still present. However, the main paths of the uplink transmissions originated from the same building are in phase and therefore orthogonal; the same benefit holds for the downlink signals transmitted by the base station.

A.1 Varying the Values of the Parameters

Simulation results with a spreading gain equal to 50 are shown in Fig. 5 and compared to the case in which the spreading gain is equal to 25. Users are uniformly distributed in the service area and a coding rate equal to $1/2$ is used. As can be clearly seen, using a spreading gain equal to 50 a smaller capacity is achieved. The performance degradation is due to the fact that, doubling the spreading gain, the number of temporal slots in each frame is halved; thus, users are aggregated in 3 temporal slots and there is less room for segregating mutually interfering buildings. As expected, the results suggest that the slotted CDMA scheme outperforms pure CDMA.

Then, the system performance when the delay spread is taken equal to 1.0 μs and a constant number of users per cell is considered is investigated. Fig. 6 shows that performance signifi-

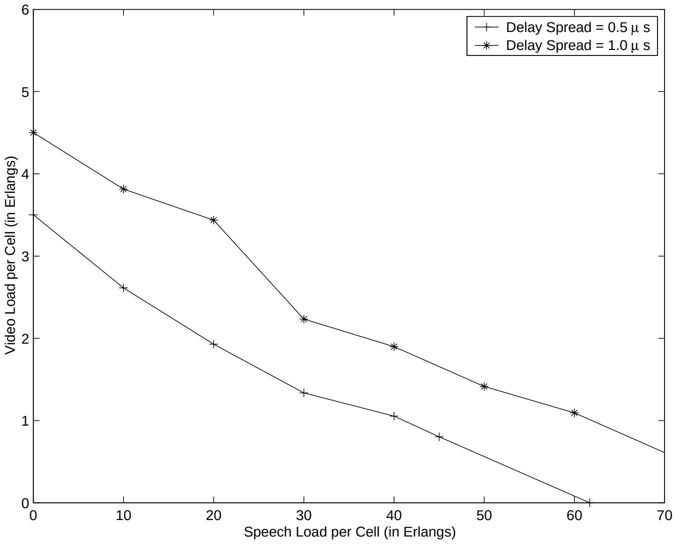

Fig. 6. Uplink capacity region for CBR traffic, a constant number of buildings per cell equal to 10, and delay spread equal to 0.5 and 1.0 μs. (QoS metrics: connection blocking probability equal to 10^{-3} and 10^{-2} for voice and video traffic, respectively.)

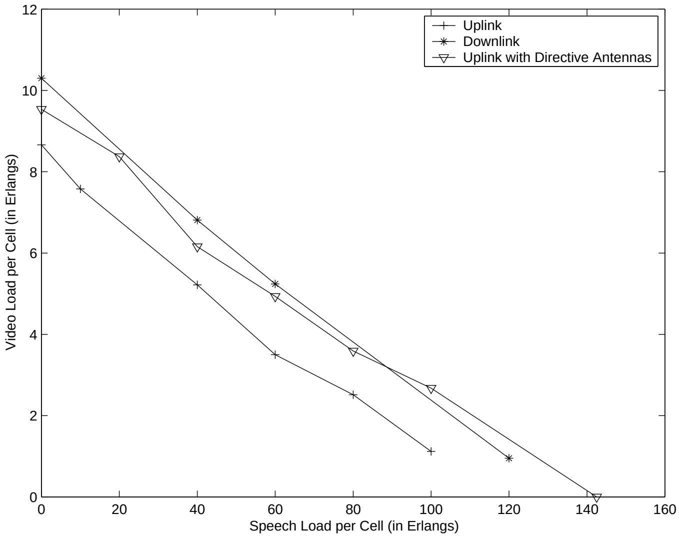

Fig. 7. Capacity region for CBR traffic, cell sectorization, and a constant number of buildings per sector equal to 5. The uplink is compared to the downlink capacity and to the uplink capacity obtained when directive antennas are used in addition to sectorization. (QoS metrics: connection blocking probability equal to 10^{-3} and 10^{-2} for voice and video traffic, respectively.)

cantly improves as the delay spread grows. Indeed, we assumed the use of a perfect RAKE receiver. The intended user signal at the base station becomes stronger as the number of rays arriving at the base station increases, whereas the interference grows by a smaller factor thanks to the orthogonality property of the codewords.

A.2 Cell Sectorization and Directive Antennas

Results were derived for cell sectorization, and a constant number of users per sector equal to 5, i.e., a number of users per cell equal to 15.

We assume that the reception profile of the base station's antenna (i.e., the antenna gain versus the direction of arrival) is given by

$$A(\theta) = \begin{cases} 1 & \text{if } \alpha_1 \leq \theta \leq \alpha_2 \\ 0 & \text{otherwise} \end{cases} . \qquad (1)$$

Details on the propagation model adopted for cell sectorization can be found in [11].

The results presented in Fig. 7 were obtained for $\alpha_1 = -60°$, $\alpha_2 = +60°$, and by considering the angles of arrival of the scattered signals ranging between $-15°$ to $+15°$ with respect to the users line of sight. The system is able to support an amount of traffic load that is slightly less than three times the load supported without sectorization. Indeed, a building located in a sector may affect also the neighboring sectors, so that the total interference a user experiences can not be scaled down by exactly a factor of 3.

Fig. 7 also shows the capacity region obtained for both cell sectorization and directive antennas at the remote terminals. The user antenna beamwidth is assumed equal to 30°. We consider that all the sectors that are located inside the area covered by the antenna beam of the user receive the whole interference power, while the sectors located outside receive a interference power scaled down by a factor equal to 8.

Although more extensive simulations should be run in this case for various values of the scaling factor, the derived results give an insight into the significant improvements that can be obtained by using directive antennas.

V. VBR Traffic Model

A. *Traffic Sources*

In the second part of this paper, we focus on VBR (Variable Bit Rate) traffic. Speech with silence activity detection and packet data sources are considered. ON-OFF models are used for both speech and data calls. As in the case of CBR traffic, the two traffic classes share the same bandwidth. We wish to determine the number of sources that may be admitted per cell so that certain Quality-of-Service (QoS) objectives are met.

In the CBR case, an admitted call was guaranteed error-free transmission for its duration and the QoS metric of interest was the blocking probability for a newly admitted call. In the VBR case, data and speech traffic sources generate packets during ON periods and are silent during OFF periods; thus, established calls are alternatively active or silent and acceptable interference levels at the receiver must be statistically satisfied. Further, data packets may be retransmitted in the case of a transmission failure. Therefore, it becomes important to evaluate the data message delay, where a message is defined as the data unit composed of the packets generated during the same ON period.

We selected the following QoS metrics at the network level:

i) the probability that a speech packet belonging to an admitted connection is lost due to excessive interference at the receiver; *ii)* the delay experienced by a data message due to delay in accessing the channel, backoff caused by transmission errors, and retransmissions; *iii)* the data packet loss probability due to queue overflow at the transmitter.

The admission policy is to allow a certain maximum number of speech and data calls to be admitted per cell. Thus, in a simulation run, a fixed number of speech and data connections, denoted by N_s and N_d respectively, are active per sector. The probability that a packet is in error, averaged over all connections in the speech traffic class, may then be found as a function of N_s and N_d. Similarly, we compute the delay statistics and the loss probability for data traffic. Based on the tolerable values of the QoS criteria for both speech and data, capacity regions are determined in terms of N_s and N_d. The plot of the system performance is a three dimensional surface. A capacity region is the intersection of this surface with the plane corresponding to a given target metric value. Similarly to the CBR case, the actual capacity region will be the intersection of several capacity regions corresponding to the different metrics.

In the next subsection we explain the traffic models used for speech and data connection. Later, the access scheme used for reserving and allocating transmission slots for both speech and data calls is explained.

B. Speech and Traffic Models

For this second study, speech and data sources are modeled as ON-OFF Markov sources. Speech traffic has an average information rate R_s=6.67 Kbps. A speech source is modeled as a discrete-time ON-OFF Markov process with time granularity equal to the duration of one frame (20 ms). As long as the source is in the ON state, one packet is generated every time frame, whereas no packets are generated during OFF periods. The average durations of ON and OFF periods are equal to $T_{on}^{(s)}$=1 s (50 time frames) and $T_{off}^{(s)}$=1.4 s (70 time frames), respectively. When active, a speech source generates information at a rate of 16 Kbps.

A similar discrete-time ON-OFF Markov process is used to model data traffic. A data source is characterized by an average information rate R_d (in these simulations, we chose $R_d =$ 153.6 Kbps), the ratio of the average information rate to the peak information rate denoted by α_d (two values were used: 0.1 and 0.2), and an average message length L_d (we chose $L_d =$ 1 Mbit). Thus, the associated ON-OFF Markov process has $T_{on}^{(d)}= (L_d{\cdot}\alpha_d)/R_d$ s and $T_{off}^{(d)}= [(1{-}\alpha_d){\cdot}L_d]/R_d$ s. Data packets generated during the same ON period form a *message*. If a user is not enabled to transmit, packets are queued in a buffer of size equal to *QueueSize*. Whenever a user's queue overflows, all the queued packets belonging to the current message are cleared from the queue; any further packets of the cleared message are discarded.

The target values for the QoS metrics are as follows: *i)* the packet and message loss probabilities should be less than or equal to 10^{-2} for both speech and data traffic; *ii)* delay specifications may change according to the average length of the data message; acceptable delays are usually of the order of 2 to 3 times the average message transmission duration. We consider acceptable average delays of 0.5 s and 1 s.

The traffic parameters that we used are listed in Table III.

TABLE III
TRAFFIC PARAMETERS USED FOR VBR TRAFFIC SOURCES

Parameter	Value
Frame Duration	20 ms
Spreading Gain	25
Time Slots Per Frame	12
Speech Model	ON-OFF Markov
R_s	6.67 Kbps
Rate in ON State	16 Kbps
$T_{on}^{(s)}$	1 s
$T_{off}^{(s)}$	1.4 s
Speech BER	10^{-3}
Data Model	ON-OFF Markov
R_d	153.6 Kbps
L_d	1 Mbit
α_d	0.1 or 0.2
$T_{on}^{(d)}$	$(L_d \cdot \alpha_d)/R_d$
$T_{off}^{(d)}$	$[(1-\alpha_d)\cdot L_d]/R_d$
Data BER	10^{-5}

C. Multiple Access for Integrated Traffic

When several data connections are simultaneously active, a MAC protocol should select which sources are allowed to transmit. Two important aspects of any MAC protocol are channel access requests through a signaling medium and scheduling [7], [13], [14]. We consider that the problem of access to the signaling channel is solved and we assume that the base station has knowledge of any transmission request from any data source in its cell. Furthermore, since our aim is to calculate the evolution of the capacity of the system when system parameters are changed and since all the metrics of interest are averages (mean message delay and mean packet loss), we disregard the problem of fairness in scheduling by increasing the size of the queues at each source and by doing round-robin polling of the transmitting sources.

A certain number of slots is available for transmission in each frame, a speech connection transmission requires one slot per frame and a data packet transmission requires a known number of codes during one frame in CDMA. Speech slots are allocated at the beginning of a simulation run and are not released throughout that run; the speech load is evenly distributed among the time slots of a slotted CDMA frame. The coding rate is equal to $1/2$, and for slotted CDMA the spreading gain is 25. Since the speech bit rate is 16 Kbps, the duration of a voice transmission slot is reduced by a factor of 2 with respect

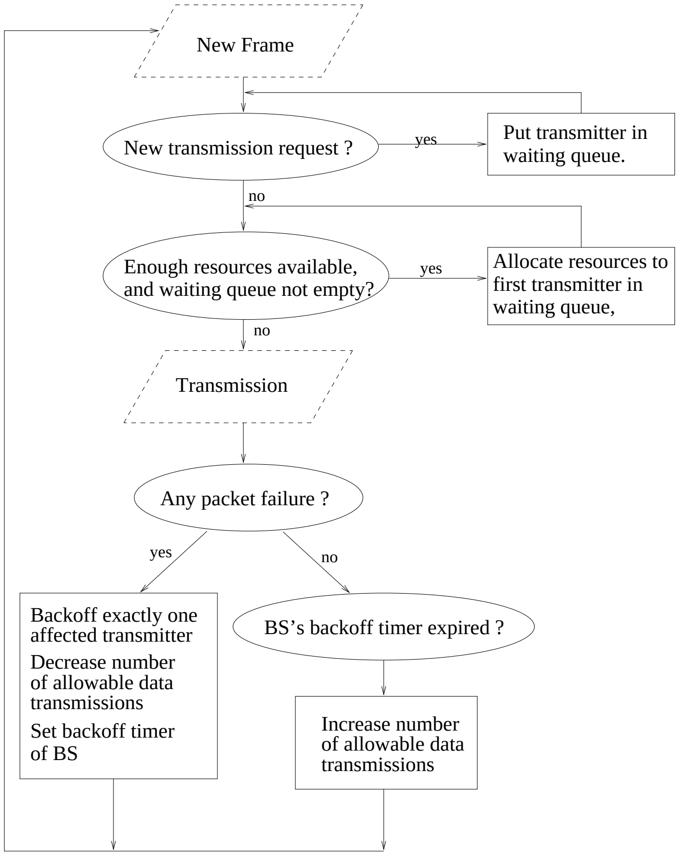

Fig. 8. MAC protocol flow chart for a base station.

to the voice model presented in section III. In slotted CDMA, a frame is 12 time slots long and up to C simultaneous transmissions using different codewords may be allocated at each time slot. This number C is not known a priori and varies according to interference conditions.

A MAC protocol is defined in order to share the remaining bandwidth between the active data connections (i.e., data sources with packets to transmit). A flow chart of the algorithm governing a base station is shown in Fig. 8.

We assume that the base station does not allow more than M_d simultaneous data packet transmissions per frame, where M_d is the maximum number of simultaneous data packets that may be successfully transmitted within a cell when no intercell interference is present. When a data connection becomes active, an unspecified ideal notification mechanism notifies the base station that the source has packets to transmit. At the base station, the transmission request is placed in a FIFO queue that keeps track of all users waiting to transmit. The base station updates a maximum number of allowable data transmission parameter; this parameter is always less than or equal to M_d. As soon as enough resources become available on the air interface, the first user in the queue is allowed to transmit. In our case, the peak transmission rate allows a data user to transmit one data packet per frame (recall that a user in ON state generates one packet per frame). Thus, the number of codes and time slots per frame required to transmit a data packet is determined by the peak user data generation rate. The user holds the reserved codewords and slots until either its queue becomes empty or it has to backoff due to a packet failure.

A backoff strategy is used to reduce packet losses and excessive interference. Whenever a data packet transmission fails, the base station chooses at random a data user among those that experienced a packet loss and forces it to backoff for a time interval equal to half $T_{on}^{(d)}$ (*BackoffPeriod*). At the same time, the base station reduces by one its current number of allowable simultaneous data transmissions and sets its backoff counter to *BackoffPeriod*. Once the user's backoff period ends, the data user decides with probability *RetransP* to request permission to transmit from the base station and with probability (1-*RetransP*) to back off again. Once the base station's *BackoffPeriod* expires, the base station increases by one the number of allowable data connections whenever no data failures occur, until the maximum number of data connections per frame, M_d, is reached.

Note that in general the target BER for data transmissions will be lower than that of speech transmissions. Accordingly, the target SIR for speech transmissions will be lower than that of data transmissions. Since our power control scheme insures the reception of all concurrent transmissions at the same SIR level on the uplink, data transmissions may fail while speech transmissions succeed. Thus, even when the media access algorithm is probing for additional bandwidth by increasing the number of data transmissions, speech calls will experience lower packet transmission failures than data calls.

The parameters used in the MAC protocol are the following: M_d equal to 2 for $\alpha_d = 0.1$ and 4 for $\alpha_d = 0.2$ (i.e., at most 16 codes may be used for data traffic); *BackoffPeriod*=$T_{on}^{(d)}/2$; *RetransP*=0.5.

VI. VBR SIMULATIONS

The simulation model adopted to study the system performance in the presence of VBR traffic is quite similar to the model used for voice and video traffic. However, in this case we always consider cell sectorization. A constant number of buildings per sector is obtained by following the algorithm described in section IV (5 buildings per sector in our simulations). The number of speech and data connections per sector are input parameters to the simulation.

Results were obtained for both a delay spread of 0.5 μs and 1 μs (L=5 and 10, respectively), and a peak to average ratio ($1/\alpha_d$) equal to 5 and 10. Thus four sets of results were obtained. The average data rate is constant at $R_d = 153.6$ Kbps and the average message length is always 1 Mbit. Both speech and data connections are initiated at the beginning of the simulation, and remain in existence for the all duration of the simulation run.

The aperture of the base station antenna is set to 120°, while the angles of arrival of the scattered signals range between

-15° to $+15^{\circ}$ relative to the user line of sight. The spreading gain is N_c=25, and a coding rate equal to $1/2$ is used. The target bit error probabilities are 10^{-3} for speech and 10^{-5} for data. Thus, according to [15] we have a coding gain equal to 3.8 dB and 5.1 dB for speech and data respectively, and the required SIR values are 0 dB for speech and 1.5 dB for data. The buffer size at the data sources is taken equal to 7.68 Mb; therefore, *QueueSize* is equal to 250 packets for $\alpha_d = 0.1$ and to 500 packets for $\alpha_d = 0.2$.

Speech connections were distributed uniformly among the time slots. The MAC protocol (introduced in section V-C) was devised to manage the allocation of channel resources to active data connections. Cell sectorization with 120° was assumed, with omnidirectional user antennas.

Our results assume that acceptable values for the quality of service parameters are: 10^{-2} for the average packet drop probability for speech, 0.5 s to 1 s for the average message delay and 10^{-2} for the average data packet loss probability. The capacity region is therefore bounded by several constraint curves.

For a peak rate of 1.536 Mbps ($\alpha_d = 0.1$) and a channel delay spread of 0.5 μ s, the behavior of the QoS metrics is illustrated in Fig. 9. The constraint curves that bound the capacity region in Fig. 10 are obtained by inspection from the plots in Fig. 9. For a target average *message delay* of 0.5 s, 7 data connections may be admitted when no speech connections are present. When only a small number of speech connections is admitted, data capacity is limited to about 4 data connections because of *speech packet losses*. When the number of speech connections is increased from 60 to 80, *data packet losses* decrease the acceptable number of data connections from 3 to 1. Finally, 190 speech connections may be accepted in the absence of any data traffic.

In Fig. 11 the two dimensional uplink capacity region is plotted for the four cases studied combining the peak rate values of 1.536 Mbps and 768 Kbps with the delay spread of 0.5 and 1 μs. In addition, the downlink capacity region for a peak rate equal to 1.536 and a delay spread equal to 1 μs is plotted.

In the case of delay spread equal to 0.5 μs, if we reduce the peak rate from 1.536 Mbps to 768 Kbps, fewer speech packet losses are encountered and capacity has a more linear shape: capacity decreases from 6 data connections when no speech connections are present, down to 1 data connection when 90 speech connections are present. The capacity curve is shaped by the constraint on *data packet losses*; when no data traffic is present, 190 speech connections may be accepted. A lower data peak rate makes the speech packet loss probability decrease since fewer channel access attempts occur. On the other hand, a higher peak rate means that the channel conditions are more variable and thus bad channel conditions are a shorter time phenomenon. Since large queues are present and the average message size is constant, at the same operating point (N_d, N_s), the data packet loss probability is higher for a lower peak rate.

For the capacity regions obtained with a delay spread equal to 1 μs, the same trends observed earlier for a delay spread of

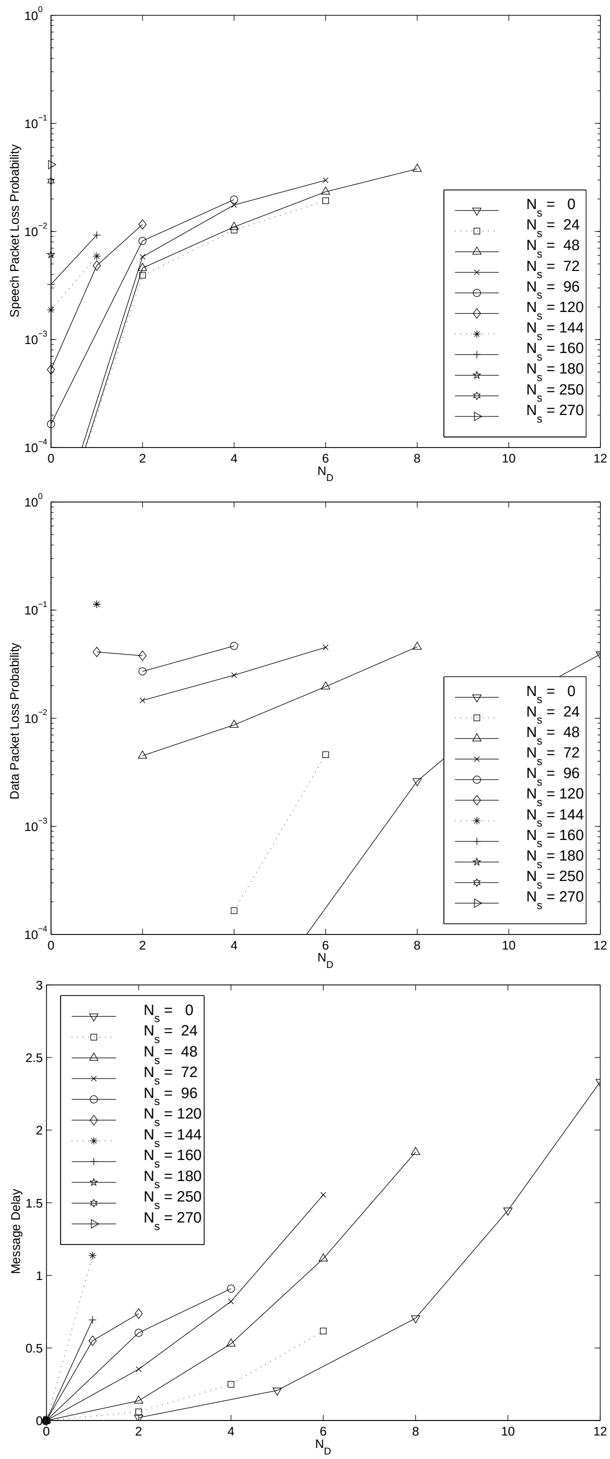

Fig. 9. Uplink QoS: Speech loss probability, data packet loss probability, and message delay. Delay spread equal to 0.5 μs and peak data rate equal to 1.536 Mbps.

0.5 μs are again apparent. The capacity in terms of the average message delay constraints is only slightly improved. For example, for a peak data rate of 1.536 Mbps, if 72 speech calls are

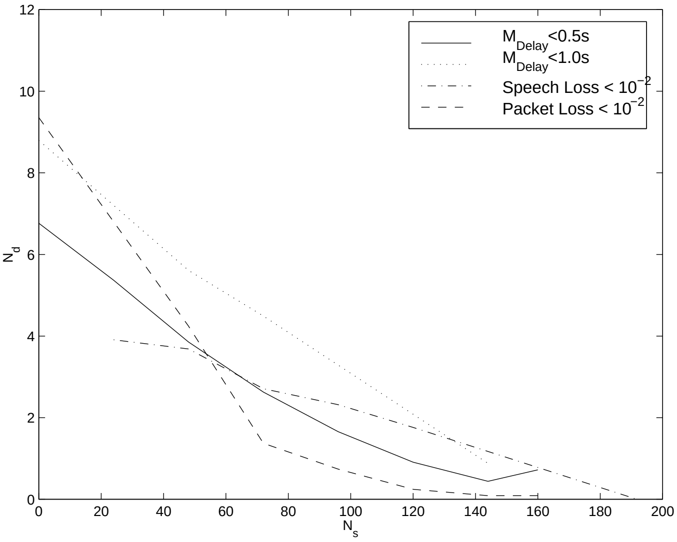

Fig. 10. Uplink capacity region for voice and data traffic. Delay spread equal to 0.5 μs and peak data rate equal to 1.536 Mbps.

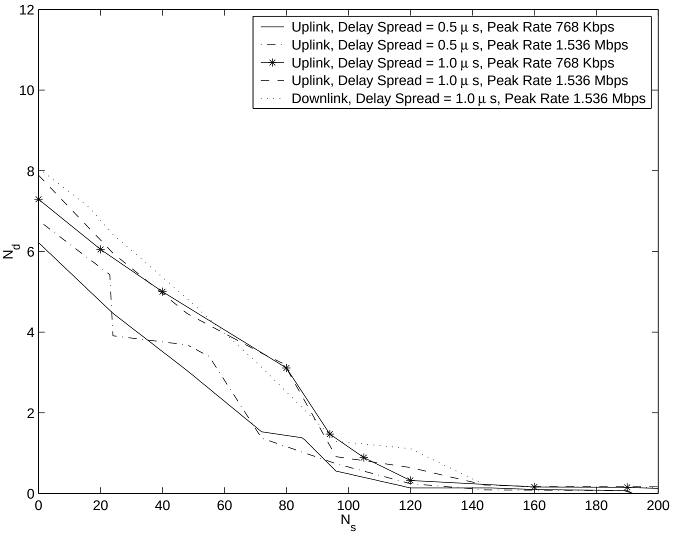

Fig. 11. Capacity region for voice and data traffic as the peak rate and the delay spread vary.

admitted: only 1 data call may be admitted when the channel time delay spread is equal to 0.5 μs, while 4 data calls may be admitted when the channel time delay spread is equal to 1 μs.

For the downlink capacity curve, we note that a capacity comparable to the corresponding uplink case is achieved.

To summarize, the results that we obtained are better for a longer delay spread because of the better diversity at the perfect RAKE receiver. A lower peak rate means that when active, the data connection remains active for a longer duration resulting in fewer access attempts but longer transmission time. Thus, the speech packet drop probability is smaller for smaller peak rates. On the other hand, for a higher peak rate data sources experience fewer packet losses, and therefore capacity increases. The average message delay is almost the same regardless of the peak rate.

VII. Conclusions

In this paper, a framework for simulating wireless systems was developed to evaluate the capacity of slotted CDMA WLL systems. Using this simulation tool, we studied the impact of various environment and system parameters on the network capacity. The major lessons learned from this effort can be summarized as follows: *i)* users in a bad location can greatly affect the average results, thus careful planning of the system is necessary; *ii)* the use of sectorization and directive antennas significantly improves system performance; *iii)* in slotted CDMA capacity increases when the spreading gain is reduced and the number of time slots per frame is increased; *iv)* a higher peak to average ratio in data sources causes better multiplexing on the uplink and therefore increases the capacity.

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