OUTAGE EFFECTS ON THE TCP-WIRELESS INTEGRATION FOR DATA/VOICE SERVICES IN CDMA SYSTEMS USING MULTIPLE ACCESS

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Abstract – We look into the issues of integrating voice and TCP data in a CDMA network using Reservation Code Multiple Access (RCMA). We dimension the buffers of the voice/data stations and consider packet delay and packet loss as quality of service measures. Channel outage is introduced in the model depending on the total number of active users. The capacity of the system is obtained under the conditions described.

I. INTRODUCTION

Broadband wireless access schemes that support integrated voice/data services are increasingly becoming a consumer expectation, fuelled among others by the rapid growth of the Internet. The important advantages of CDMA (Code Division Multiple Access) systems in this regard - robustness/graceful degradation vis-a-vis multiple access interference (MAI) and multipath fading, soft handoff capability and enhanced system capacity - have been well-documented. Presently, most wireless services are voice oriented, thereby presenting a design challenge for incorporation of data services due to the different nature of voice and data traffic. In this work, we analyze a multiple-access protocol for voice and data services in a CDMA system and its impact on TCP/IP performance at the network level.

Voice and data packets are treated separately due to their different characteristics with voice packet loss probability and data packet delay being the primary figures of merit, respectively. The analysis in [1] does not account for the effects of multiple access interference and finite buffer size for the data terminals since an infinite buffer is considered. In [3], the model of [1] is extended to consider the data subsystem with finite buffers in the stations and the effects of interference by using RCMA. Thus our submission focusses on the issues of code utilization, buffer dimensioning and management techniques with a view to establishing the trade-offs between capacity and buffer size.

Finally, we assume TCP/IP is used at the network level for wireless data packets, and we seek to investigate it's performance with the proposed CDMA/RCMA multiple access scheme under finite buffering and lossy channel models. We provide results on the dimensioning and buffer management applying packet discard methods, the trade-offs of outage and system capacity, as well as those of buffer size and capacity by integrating voice and data services as in [3]. TCP/IP packets are assumed to have a length of 576 bytes, [2], to establish the optimal buffer length and a threshold of admittance in order to minimize the number of lost packets and thus achieve optimal performance.

II. MODEL DESCRIPTION

The scheme used here, as in [1], is for the use in both short-range local wireless communications and microcell mobile communications, where a central station (base station) serves a number of voice and data terminals which can be either fixed or mobile (portable). The scenario considered is the uplink on a single microcell of a DS-CDMA system, where there are both voice and data terminals who are potential transmitters. Mobility is not considered in this model, we are considering a channel affected by outage due to interference. All of these terminals share a group of N spreading codes which are assigned according to the RCMA (Reservation Code Multiple Access) protocol. Let M_v and M_d denote the total number of voice and data terminals, respectively (from these, the set of data terminals is considered to have a source of fixed length TCP/IP packets to transmit). Among them, b data terminals and c voice terminals are in contention. An indispensable operation is to identify the status of the spreading codes. Using RCMA, each code

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is identified as *reserved* or *available* based on the feedback information (acknowledgement messages) from the base station at the end of each slot. The feedback information is carried on one or more specific spreading codes on the down-links from the base to terminals. When any of the terminals needs to transmit a packet, it will need to make first a reservation of the spreading code that will be used. A *reserved* code will exclusively serve the terminal that has been granted a reservation in the subsequent slots; an *available* code can be likely used by any terminal with packets in its buffer but no reserved code in possession.

The RCMA protocol treats voice packets and data packets in a different way because of the different features of these services. One factor considered in the RCMA protocol design is the fact that voice traffic requires real time service while data does not, but data need lower BER than voice. Once a voice packet succeeds in contending for an *available* spreading code, the protocol grants the terminal a reservation for the exclusive use of this code in subsequent slots until all packets in that talkspurt are transmitted. On the other hand, when a voice packet fails to obtain a spreading code and stays in a contending state for time longer than a preset threshold D_{max} it has to be dropped. The threshold is determined by the delay constraints on speech communication and it is a design parameter of the RCMA system, [1]. In the numerical results section we found that a D_{max}=0.25 time units is appropriate for the case. Data terminals have to compete for a spreading code to transmit each of its packets. The data packets failed to be transmitted will be held in the buffer without time limitation, however, if more packets are received in the buffer than its total capacity K, the entire batch is dropped. This because it would be a delivery of an incomplete TCP packet to the next layer and a retransmission would be requested anyway.

All the control is implemented through three operations. First, the base station synchronizes all the terminals by broadcasting the system timing whereby the terminals can adjust their timing clock. Since the timing error among terminals in a short-range area is small, it is feasible to use the slotted (synchronous) spread spectrum signaling in the system. Second, the base broadcasts traffic information including the status of transmitted packets and the codes that will be available in the next time slot. A short time delay in a small coverage makes it possible for the terminal with a transmitted packet to receive feedback information from the base station in the same slot as the packet was sent. And third, by assigning different priority to voice and data, the RCMA system provides a powerful mechanism to integrate the two services. The common point of the two services is that a packet from either voice or data terminal has the same size and is spreaded by a spreading code which is chosen by the RCMA protocol.

The wireless channel is an important point for our purposes, it is the main restriction in a transmission when the number of users increases because the channel goes into the outage state. An analysis on how to model this is done in [?]. Here, the channel model is taken to represent the state of outage in the wireless channel. When the signal power is below a given threshold, the channel is in the bad state, otherwise it is in the good state. If the channel is in the good state at the beginning of the time slot, and it remains in the good state throughout the duration of the time slot, then the probability of successful transmission (departure) is P_d. In all other cases the probability of successful transmission is assumed to be zero. If the transmission is successful, the CDMA-data-packet departs, otherwise it will be retransmitted repeatedly in the following slots until a successful transmission occurs. If the transmission of a voice packet is failed, it is retransmitted for a time equal to D_{max} as maximum, if after this time no succesful transmission is done, then the complete talkspurt is dropped.

III. SUBSYSTEMS

Voicesubsystem.– The speech activity for a voice terminal can be modeled as a two-state model in which conversational speech is characterized by periods of activity and periods of silence, [1]. In activity, the bit stream is packetized, the group of packets generated in the talking state are called a talkspurt, and the length of a talkspurt is a random variable L. A voice terminal can be in the *silence*, *contention* or *reservation* state. After a code has been reserved, the terminal keeps transmitting the complete talkspurt, when the talkspurt is transmitted, the spreading code is released. The state evolution can be described by a model shown in [1] The maximum allowable delay for voice packets is a presetted value D_{max}.

Datasubsystem.– A data terminal has a bit rate different from a voice terminal, here, data terminals are considered to generate TCP/IP fixed size packets equivalent to n CDMA packets in each arrival. The data packets are packetized into smaller packets the same way as the information stream of a voice terminal. A batch of packets will be rejected complete if the buffer of length K can not afford the complete batch. A data terminal is not permitted to reserve a spreading code, it can only stay either in an *idle* state or in a *contention* state. The terminal remains in the contention state as long as it has at least one packet in its buffer (backlogged terminal). A backlogged data terminal has to contend for a code to transmit for each packet, the data terminal releases the code after the transmission of one single packet and gets back to contention. Evolution of buffer in each terminal can be described as a two-dimensional process, depending on the state of the channel due to outage as described in [4]. In both, voice and data terminals, a transmission can be unsuccessful due to the state of the channel which is considered to get into an outage state due to the interference as the number of terminals in the system increases.

IV. NUMERICAL RESULTS

We illustrate the results obtained with a simulation done for the RCMA protocol working over a CDMA system adding a characterization of the channel as described before. TCP is assumed to use constant length packets of 576 bytes, [2], and to have a Poisson rate of arrival. In voice terminals, the rate to pass from silent state to contention state is $\sigma_d = 1$ arrival/time unit and the service rate is $\gamma = 200$ packets/time unit for the voice subsystem model and for the data model these are $\sigma = 200$ packets/time unit and $\gamma = 1$ packet/time unit. The length of voice talkspurts has a geometric distribution with parameter $(1 - \gamma) = 0.09$. The outage function is taken as proposed in [?]. Perfect power control is considered at each terminal. The outage probability given the number of users present in the system (both voice and data terminals) is calculated by means of a cumulative Gaussian function whose mean and variance are 83 and 13, respectively.

BufferSize.– One parameter to analyze is the buffer size because it is important to have an optimal buffer length to avoid losses and unnecessary buffer size in each terminal. In Figure 1 we see simulation results with 64 users and 64 codes, it can be observed that the packet loss is reduced and becomes stable approximately at 75 CDMA packets of capacity. From here we consider the buffer size as 78 CDMA packets, which is the equivalent to 3 TCP packets. Packet loss becomes zero as the buffer size increases, while voice packets maintain certain level of loss almost fixed, this level is directly governed by the maximum delay permitted (preset) D_{max}, in Figure 1, this D_{max} has a value of 0.25 time units.

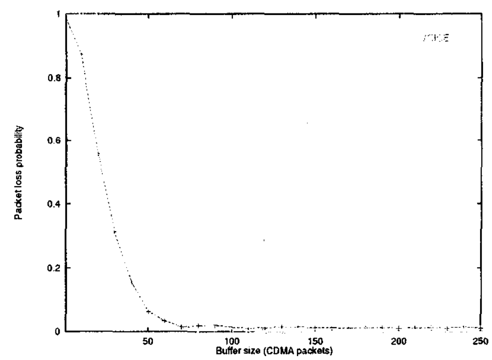

Figure 1: Packet loss probability, D_{max}=0.25 time units, N=64 users, 64 codes, No outage

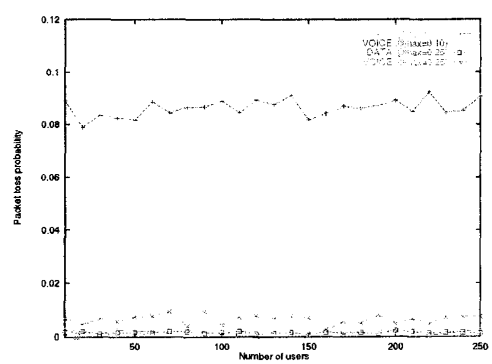

Figure 2: Packet loss probability (voice and data), buffer size=78, 64 users. No outage.

Capacity.– To obtain the capacity of the system a simulation was done with the conditions of 64 codes and a buffer size of 78 packets, varying the number of users from $N = 0$ to $N = 250$ in two cases: with and without the channel affected by outage. As shown in figures 2 and 3, we can interpret these two cases. In the no-outage case, Figure 3, it can be seen that the packet loss in the system remains constant eventhough the number of active users increases. The voice packet loss is mostly caused by the preset value of D_{max}, which is set to satisfy QoS required for the service. So in the no-outage case, the loss of voice packets is directly influenced only by the value preset for D_{max}.

In Figure 3 the factor of outage is taken into account and as it can be seen, depends directly of the number of users in the system. It is possible to realize that outage is the first constraint for the capacity of the system. Figure 3, shows how voice packets are lost as a function of both the outage probability and the value of D_{max}. When there are a few users in the system the loss depends on D_{max}, and as the number of users increases the packet loss increases due to the high outage probability and D_{max}.

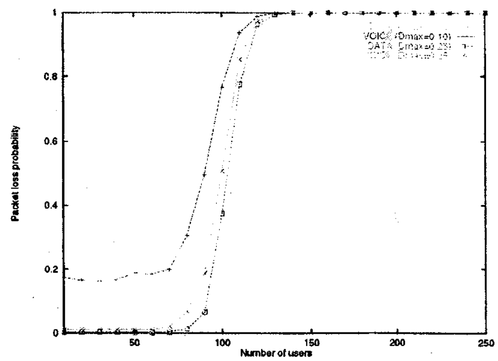

Figure 3: Packet loss (Voice and Data) increasing number of users, buffer size=78, 64 codes, Outage.

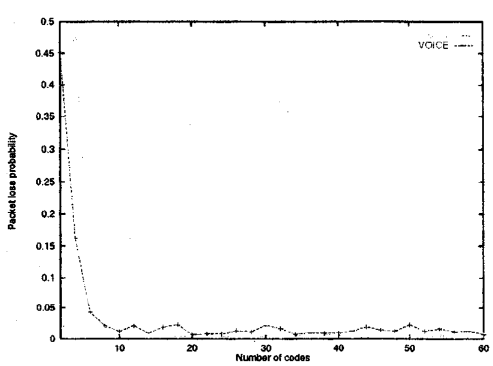

Figure 4: Packet loss, 64 terminals, buffer size=78. Outage

If we vary the number of codes, Figure 4, the losses become stable after 10 spreading codes in the system. In Figure 4 we can see that the number of spreading codes N does not need to be high to get more capacity on the system, we already showed how outage probability does not allow higher capacity. So we have that 10 codes are sufficient to maintain the same good performance for 64 users within the system.

Until now we have seen the packet loss in the system while varying the number of users, this is, increasing the same number of voice and data terminals at a time. In figures 5 and 6, we plot the packet loss while varying separately voice terminals and data terminals. It can be seen that the number of users, regardless of the type of these, produces certain outage probability which causes the channel to be in the bad state (outage state), and this is reflected in more losses in the system. It can be seen that the number of 64 users approximately is good for the system performance no matter if they are voice or data type.

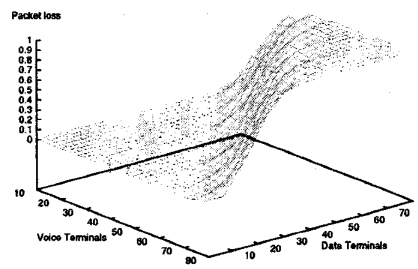

Figure 5: Packet loss, D_{max}=0.25 time units, buffer size=78. Data.

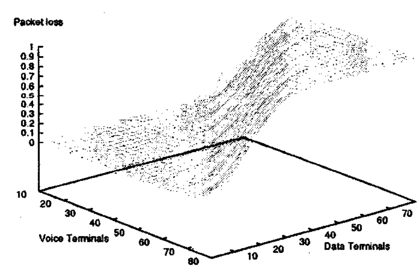

Figure 6: Packet loss, D_{max}=0.25 time units, buffer size=78. Voice.

The plot of the surface of Figures 6 and 5 shows the same results. According to data and voice losses surfaces, we can confirm that there is no difference if the number of terminals is not balanced to either data or voice. The important factor here is the total number of terminals which produces the outage probability and causes packet loss.

Delay.—Another important parameter in the system is the delay of the packets that are transmitted for each terminal (voice or data). This delay is considered as the time elapsed between the arrival of a packet to the terminal buffer and the time in which the packet is transmitted successfully. Although the data packets are allowed to have a more flexible delay, it is important to have as a reference of performance of the system the delay of the data packets before they get transmitted. Next, we analyze the voice and data packets delay when the number of users in the system increases.

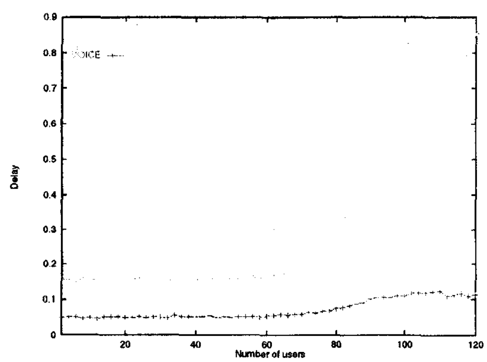

Figure 7: Voice and data packets delay. Buffer size=78, Dmax=0.25 time units

The delay of data packets is more affected by the number of users in the system, as we can see in Figure ??, as outage probability increases because of the amount of users in the system, delay of data and voice packets increases too.

Fixing the values for the number of users, buffer size and number of codes, the delay found on the time elapsed between the arrival time and the departure time after a successful transmission of a packet is approximately constant of 0.17 time units while the number of users do not pass the maximum allowed of 64 users. This time is less than the value fixed for D_{max}.

V. CONCLUSION

We found for the parameters established earlier that the minimum buffer size for each terminal for which the system finds good performance is 75 CDMA packets, but we propose 78 because this is equivalent to 3 TCP packets. In this way, if two TCP packets arrive there exists still the possibility to allocate a third one. With respect to capacity, the system would support a maximum of 64 users before experience significant losses due to outage because of the high number of active users in the system. If this happens the number of users will cause to surpass the limits of BER allowed by each type of terminal.

As seen in Figure 4, only 10 spreading codes are enough to support the quantity of users pointed earlier. This result is proposed by [3], channel impairments were not taken into account. For the delay we can conclude that for the same number of users, voice packets keep an average delay limited by D_{max}. With respect to delay of data packets, it can be observed that it maintains a low delay which results in a better QoS.

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