ISMA-DS/CDMA MAC Protocol for Mobile Packet Radio Networks

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Abstract **-** In this paper an ISMA-DS/CDMA MAC protocol for a packet transmission network is presented. The main feature of this protocol is its ability to retain the inherent flexibility of random access protocols while at the same time reducing to some extent the randomness in the access in order to increase the system capacity. In this framework, the protocol is adapted to a frame structure similar to that specified in the UTRA ETSI proposal for third generation mobile communication systems. Additionally, some adaptive mechanisms are proposed that improve protocol performance by means of varying the transmission bit rate according to the channel load that is broadcast by the base station. As a result, an adaptive bit rate algorithm is presented that reaches a throughput value close to the optimum.

I. Introduction

Research into efficient MAC (Medium Access Control) protocols in a wireless environment gains added interest in light of the fact that third generation mobile communication systems will be focused on providing wireless access to different kinds of multimedia services, thus needing to handle traffic sources of very different statistical nature with similar capabilities to those of the wired backbone network. In this context, DS/CDMA (Direct Sequence Code Division Multiple Access) packet transmission networks play an important role thanks to their inherent statistical multiplexing capability.

The main objective of medium access protocols is to guarantee efficient access to a set of shared resources. These resources strongly depend on the multiple access technique being used. This paper will focus on the DS/CDMA technique, as it is emerging as the predominant multiple access scheme to be used in third generation mobile communications systems, particularly in proposals such as UTRA (UMTS Terrestrial Radio Access) or cdma2000. According to this scheme, in which all the users share the same bandwidth and time slots, the resources to manage become the number of available codes and also the power levels that can be applied in order to limit the overall interference level.

For DS/CDMA networks, the most appropriate assignment strategy to manage a set of code sequences will depend on the service to be provided. For example, in applications such as voice, where users transmit continuous flows of information, the assignment of codes on a per user basis after an initial setup process can be an efficient strategy. However, this fixed assignment strategy lacks flexibility when dealing with packet based applications that generate bursty traffic. In such cases, other random access policies become more suitable.

One of the most commonly used random access protocols is S-ALOHA. In a DS/CDMA environment, according to this protocol when a given user needs to transmit a packet, it randomly selects a code sequence from the available set of existing codes in the base station and performs the transmission. This protocol is very flexible for light traffic loads, but as the load increases, the system becomes limited not only by multi-user interference but also by a higher number of collisions in the access. This limitation arises mainly from the fact that users do not have knowledge as to whether or not there are other users transmitting with the selected code sequence.

It is obvious that this type of protocol can be improved by reducing to some extent the randomness in the access. According to this criterion, the ISMA (Inhibit Sense Multiple Access) protocol achieves this reduction by broadcasting information about the uplink channel in the downlink. Specifically, the base station indicates the status (*busy* or *free*) of the various available code sequences. Thus, users have additional information before attempting to access the system. This protocol was originally designed on a TDMA basis as an adaptation of CSMA (Carrier Sense Multiple Access) designed to overcome the drawbacks of the radio environment, such as the hidden terminal problem [1][2]. Some adaptations of this protocol to CDMA are presented in [3]-[5].

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The basic behavior of the ISMA-DS/CDMA protocol is shown in Figure 1 according to the model presented in [4]. Users remain in an *idle* state while they have no information to be transmitted. Whenever the application levels of a given user require a message to be sent, it is split into fixed length L-bit packets. Then, the broadcast channel is listened to in order to determine the available code sequences (i.e., those which are not *busy*). After this process, one of these codes is randomly selected and the first packet is transmitted. If a collision occurs (i.e., another user has selected the same code sequence in the same time slot) or if the packet is degraded by multi-user interference, the user will change to the *backlogged* state and attempt to gain access again in the next time slot by applying a certain probability p_r. However, if the packet is successfully received and the message contains other packets to be sent, the user will change to the *transmission* state, the base station will broadcast the selected code as *busy* and it will be assigned to the user until the end of its transmission. Then, no other users will be allowed to select that code, so access to it is said to be *inhibited*. However, having acquired a code does not guarantee success in subsequent packet transmissions, mainly due to multi-user interference. In this case, the user will retain the code, but it will change to the *retransmission* state and attempt to transmit the corrupted packet again in the next slot by applying a certain probability p_b.

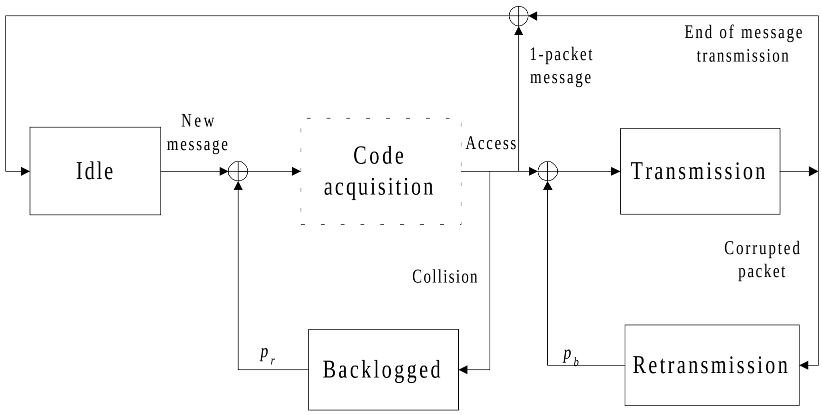

Figure 1 ISMA-DS/CDMA model

According to this behavior, in [4] a Markov modeling is presented to analyze the performance of the protocol in terms of throughput and message delay, and is validated against simulations. However, this modeling is too complex to obtain results for high values of the number of available codes or the number of registered users. Furthermore, it fails to take into account a number of issues regarding practical implementation. Particularly, it assumes that at the beginning of each time slot all the users can know the status of the available codes and whether the next packet to be transmitted corresponds to a new packet or to a retransmission. This paper works from this basis, and in an attempt to cope with these issues, an ISMA-DS/CDMA MAC protocol is proposed and adapted to a realistic frame structure such as that defined in the UTRA (UMTS Terrestrial Radio Access) ETSI proposal [6]-[8].

The knowledge that ISMA-DS/CDMA provides about the system load can be further applied to improve protocol performance. Particularly, in this paper some strategies are presented regarding the use of adaptive access probabilities and adaptive transmission bit rate algorithms depending on the number of *busy* code sequences. In this framework, two adaptive algorithms are compared: the MS algorithm, presented in [9] for an S-ALOHA-DS/CDMA system, and the proposed ThMS algorithm, which overcomes some of the deficiencies of MS by taking advantage of the inherent characteristics of ISMA-DS/CDMA. Specifically, ThMS is shown to present a better reaction capability in the face of changes in the overall system load.

The rest of the paper is organized as follows. In Section II, the channel organization and frame structure of the UTRA proposal are briefly described in order to present the proposed ISMA-DS/CDMA protocol. In Section III, the adaptive transmission bit rate algorithms MS and ThMS are presented and compared, resulting in a better behavior for the latter. Finally, in Section IV the conclusions are summarized.

II. Adaptation of ISMA to the frame structure defined in UTRA

A. Channels and frame structure defined in the UTRA proposal

With the aim of defining a realistic framework for the ISMA-DS/CDMA protocol, the frame structure defined in UTRA will be considered. In the following, a brief system description of UTRA is given in order to better understand how the protocol can be adapted to it.

UTRA is based on FDD-WCDMA (Wide-band Direct-Sequence CDMA) technology with a basic chip rate of 4.096 Mcps. The following physical channels are defined:

- DPDCH and DPCCH (Dedicated Physical Data Channel and Dedicated Physical Control Channel): The first carries information from layer 2 and above, while the second carries layer 1 control information such as pilot signals and power control indications. In the uplink, these channels are mapped into the I and Q components of the transmitted signal, while in the downlink they are time multiplexed. The frame structure for the uplink is shown in Figure 2. As can be observed, the time axis is subdivided into 10 ms frames which in turn are divided into 16 slots per frame, each of them corresponding to a power control period.

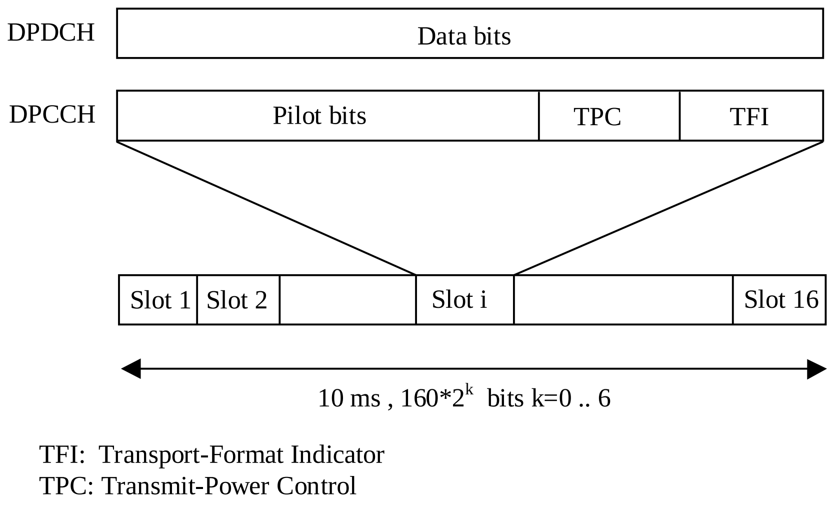

Figure 2 Frame structure of DPCCH/DPDCH in the uplink

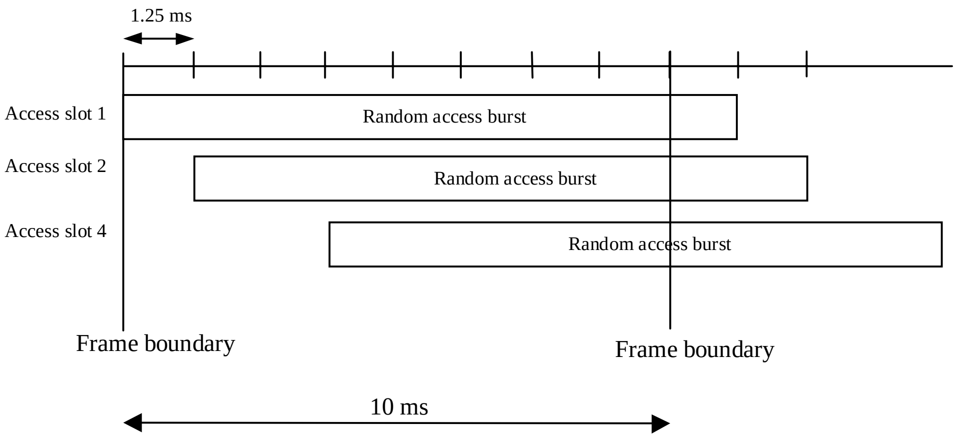

Figure 3 Frame structure in the PRACH channel

- Primary and Secondary CCPCH (Common Control Physical Channel): The first carries the broadcast control channel while the second carries other common control channels such as the FACH (Forward Access Channel) and the PCH (Paging Channel).

- Primary and Secondary SCH (Synchronization Channel): These allow the mobile stations to acquire slot and frame synchronization with the target base station.

- Physical Random Access Channel (PRACH): This channel is intended to transmit random access bursts and short packets in the uplink. The frame structure is shown in Figure 3. A 10 ms frame is divided into 8 access slots (also called offsets), and each random access burst has a total length of 10 ms preceded by a 1.25 ms preamble.

The random access procedure currently defined in UTRA is based on S-ALOHA-DS/CDMA. Whenever a mobile terminal wishes either to access the system or to transmit a short packet, it randomly selects an access slot in one of the existing random access codes and starts the transmission. It then waits for confirmation through the FACH channel. A collision may occur if two or more users select the same code and the same access slot. However, if the same code is selected by two users in different access slots, both packets will be correctly received thanks to the auto-correlation properties of the spreading sequences [6]-[8].

B. Adaptation of ISMA-DS/CDMA to the UTRA frame structure

When attempting to adapt the ISMA-DS/CDMA model defined in Section I to the UTRA frame structure the following practical considerations must be taken into account:

The base station should be able to broadcast the status of all the available codes (*busy* or *free*) just before the beginning of each packet transmission (i.e., just before the beginning of each 10 ms frame). Consequently, it should also be able to determine which initial packets have been correctly received and which users have transmitted their last packet and as a result will release their codes in the next frame.

The first objective can be accomplished by making use of the frames and slots structure proposed in UTRA. Particularly, new users should only be allowed to access the system during the first 1.25 ms of each frame (i.e., during a PRACH channel access slot). Access would consist in selecting a *free* code and transmitting a preamble A on it (see Figure 4). During the following 1.25 ms period a response R from the base station will be expected in the corresponding downlink of the selected code. This response should indicate whether or not the code acquisition has been correctly performed. According to this procedure, after the first 1.25 ms period the base station will know exactly which codes will be *busy* during the next frame and will broadcast this information for the remaining 8.75 ms frame period, for example by making use of a Secondary CCPCH.

In order to indicate when a given code will be released, each transmitted packet should include a bit indicating whether or not there are other packets waiting for transmission. Regarding the acknowledgement delay, it can be assumed that a packet transmitted during frame n will be acknowledged in the downlink during frame $n+1$.

Finally, power control must be taken into account. It is obvious that at the beginning of a transmission, a closed-loop power control strategy cannot be applied due to the lack of a downlink dedicated control channel. This means that the first transmission will be performed by applying an open-loop power control technique. However, after a code has been acquired, the existence of a downlink control channel can be assumed and then it can be considered that subsequent transmissions will be performed with an ideal closed-loop power control thanks to the 16 power control periods per packet.

Therefore, the protocol behavior is described as follows (see Figure 4):

1) All the registered users are synchronized with the frame structure of the target base station. They generate messages that are divided into packets of a fixed length depending on the spreading factor to be used and put into the transmitter buffer. The spreading factors defined in UTRA are $S_f=256/2^k$, k=0..6, corresponding to packet lengths of $L=160*2^k$ bits respectively.

2) Whenever a message is to be transmitted, the terminal listens to the broadcast channel and determines those codes that are not being used at present. Then, it randomly selects one of them and transmits an initial 1.25 ms preamble A by applying an open-loop power control. In addition to other information, a random number is transmitted in this preamble. A certain CRC is also added to the transmitted information in order to detect the correct reception of the preamble. In the event of the packet being correctly received, the base station acknowledges it in the corresponding downlink of the selected code during the next 1.25 ms period and replies with the detected random number together with power control indications. The purpose of this random number is to avoid a situation in which two or more users have selected the same code but, due to capture effects, one of the packets has been correctly received. In such cases, only the user whose number corresponds to the number replied by the base station will have acquired the code.

   Before transmitting the preamble in the selected code, a regulation method is applied, that consists on applying a certain probability p_r depending on the number of *busy* codes broadcast by the base station. This allows to reduce the number of random accesses whenever the number of active users in the system approaches to a given maximum K_{max}, whose value is discussed later. This probability is applied not only by backlogged users but also by idle users, thus avoiding unfairness between them.

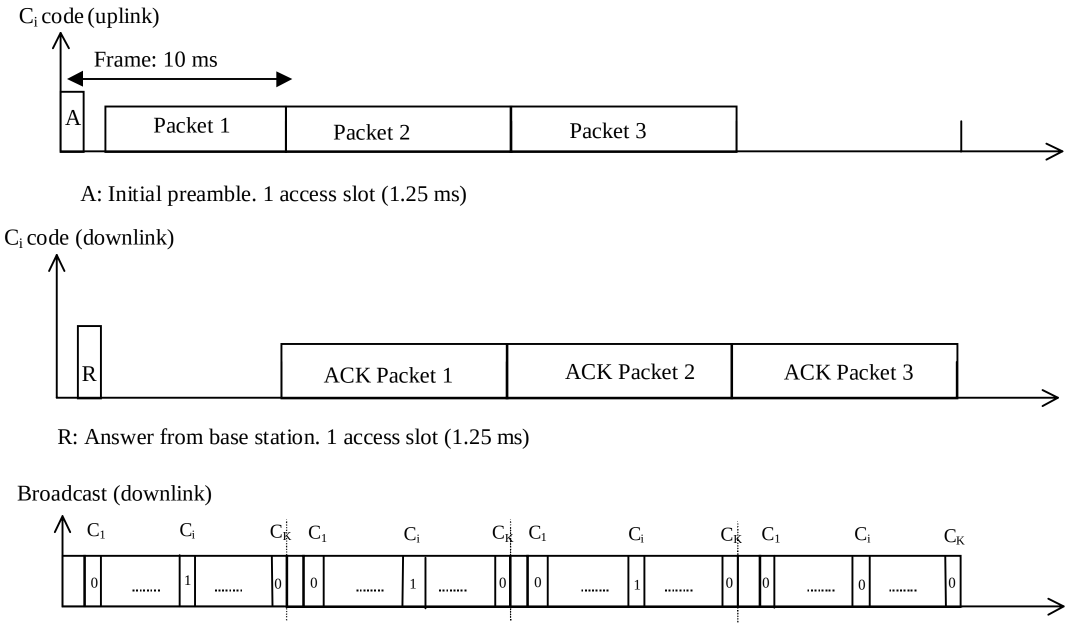

Figure 4 Frame structure of the ISMA-DS/CDMA protocol

Particularly, it should be noted that low values of p_r will tend to reduce the number of simultaneous users in the system, thus decreasing the collision probability but increasing the delay, while high values will tend to facilitate the access, thus reducing delay but also reducing the throughput. Accordingly, when there are few simultaneous users, high values of p_r may be suitable, whereas when the number of simultaneous users increases, it is advisable to reduce or even inhibit new accesses by means of p_r values close to zero. As a result, p_r should decrease from 1 to 0 when the number of simultaneous users (i.e., *busy* codes) increases from 0 to K_{max}.

A possible function to specify the variation of probability p_r in terms of the number of *busy* codes x may be, among others, a linear variation, this is:

$$p_r(x) = 1 - \frac{x}{K_{max}} \qquad (1)$$

This linear variation is assumed instead of other possibilities, such as the quadratic one, as not significant differences have been observed when comparing the results for different cases.

3) After having transmitted the preamble, the mobile terminal listens to the downlink of the selected code and waits for the base station reply R. If this reply indicates that the code has been correctly acquired, it continues the transmission for the rest of the frame and the successive frames, while the base station broadcasts the status of the code as *busy*. If no reply is received, or if the random number does not correspond to the transmitted one, the mobile terminal changes to the *backlogged* state, and will reattempt access, with probability p_r, in the subsequent frames.

4) During the successive frames the remaining packets are transmitted, and ideal closed-loop power control will be assumed thanks to the dedicated downlink control channel and the 16 power control periods per packet. The base station acknowledges a packet during the next frame. If a packet is not correctly received due to multi-user interference it will be retransmitted in the subsequent frames with probability p_b, but meanwhile the code will remain *busy*. A Selective Repeat ARQ (Automatic Repeat reQuest) technique is assumed.

A linear backoff strategy can be used to control the retransmission probability p_b [10] in order to decrease the interference level whenever the number of users increases. Assuming this strategy, p_b decreases as a function of the number of retransmissions of a given packet in the following way:

$$p_b(\text{i} - \text{th} \quad \text{retransmission}) = \frac{1}{i} \qquad (2)$$

5) A bit indicates in each packet whether or not there are other packets waiting for transmission. Then the base station knows when to broadcast the code status as *free* again.

The number of maximum simultaneous users in the system K_{max} depends on system parameters such as the spreading factor S_f, the packet length L, the correcting capability t, the power control technique being used and other criteria such as the maximum block error rate allowed. For example, let us assume a system with S_f=64, L=640 bits corresponding to one of the specified bit rates in UTRA. Additionally, let us consider that no channel code is being used (t=0) and that an ideal closed-loop power technique is applied in a single cell scenario. Under this ideal situation, the Gaussian approximation can be applied to model interference and to determine the bit error probability [11]. Then, the maximum throughput that could be reached can be calculated by maximizing the following expression:

$$S = xP\left[\text{correct packet} \middle| x \text{ simultaneous transmitted packets}\right] = xp_{ok}(x) = x\left(1 - \frac{1}{2}erfc\left(\sqrt{\frac{3S_f}{2(x-1)}}\right)\right)^L \quad (3)$$

The maximum of (3) is found to be S_{max}=14.3 packets/slot, which corresponds to a number of simultaneous transmissions x_{max}=16. This means that, under these conditions, a possible criterion would be to set K_{max}=16 in order that no new accesses are allowed whenever the number of simultaneous users equals 16, as more users would cause the throughput to decrease. Other possible criteria would be to fix K_{max} depending on the maximum allowable block error rate that could be related to a particular quality of service to be provided.

Note that function (3) does not consider any specific MAC protocol and simply takes into account the number of simultaneous transmissions, without considering how users have entered the system. As a result, the value of 14.3 corresponds to the maximum throughput value that any MAC protocol could reach under the same assumptions regarding S_f and L. Hence, it can be used as an upper bound to measure the performance of a given protocol such as ISMA-DS/CDMA.

C. Comparison between ISMA-DS/CDMA and S-ALOHA-DS/CDMA

In order to show the good behavior of the ISMA-DS/CDMA protocol in a packet transmission environment, the comparison between this protocol and the S-ALOHA-DS/CDMA protocol defined in UTRA for the transmission of short packets in the PRACH channel is presented in this subsection.

In the S-ALOHA protocol, access is allowed in each of the eight access slots of a frame, so several users can share the same code by making use of different time offsets. However, in ISMA-DS/CDMA new users are only allowed to select a code during the first access slot of each frame, because at the end of this period the base station must know all the correctly acquired codes. Consequently, in this protocol, a code sequence cannot be shared by different users.

Figure 5 presents the comparison between the two protocols for N=100 users sharing K=30 code sequences, average message length=10 packets, spreading factor S_f=64 and packet length L=640 bits. It can be observed that ISMA outperforms S-ALOHA in terms of both throughput and message delay. This is due to the fact that ISMA provides users with knowledge about the system status and avoids collisions once a code has been acquired.

In Figure 6, the effect of different message lengths is shown. Particularly, it should be pointed out that a decrease in the message length corresponds to a decrease in the differences between the two protocols. The reason for this is that when transmitting short messages, users take less advantage of the code acquisition process as they maintain the code for a shorter period of time. In the extreme case when messages only contain a single packet, S-ALOHA behaves better than ISMA since it allows several users to share the same code with different time offsets and also because in this case the information in the broadcast channel of ISMA is of no use, as no user maintains the selected code after the first packet transmission. The main point here is that the S-ALOHA scheme defined in UTRA is specially designed for short packet transmissions, while the proposed ISMA protocol is also able to handle greater message lengths thanks to its ability to maintain the acquired codes.

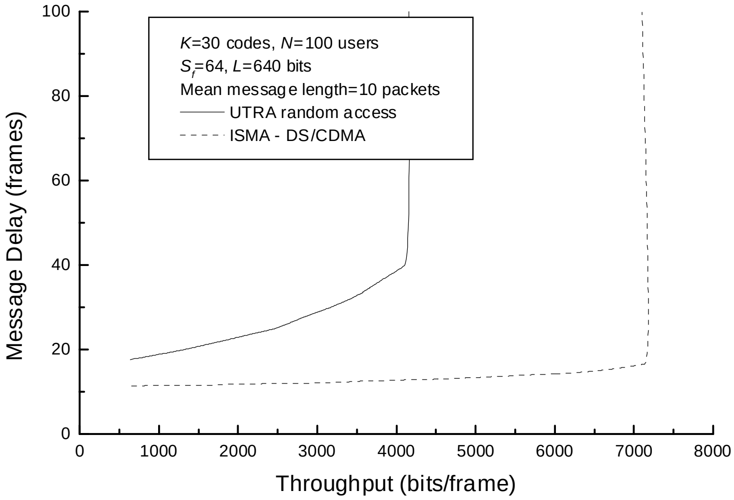

Figure 5 Comparison between ISMA and S-ALOHA in terms of the response throughput-message delay

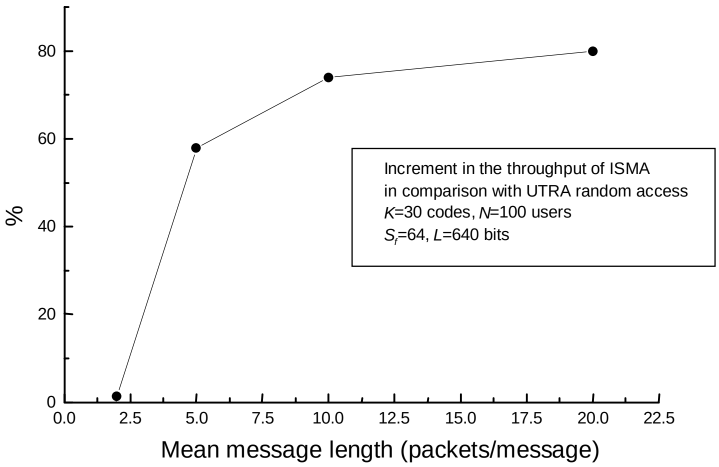

Figure 6 Increment in the maximum throughput between ISMA and S-ALOHA for different message lengths

III. Adaptive transmission bit rate algorithms in ISMA-DS/CDMA

One of the keys for the flexibility of DS/CDMA is the fact that the transmission bit rate can be varied together with the spreading factor (i.e., the protection against interference) without modifying the total bandwidth. This means that a further improvement can be achieved in a MAC DS/CDMA based random access protocol when applying mechanisms to adapt the transmission rate to channel characteristics. Accordingly, under low load conditions, users can make use of reduced spreading factors to obtain high bit rates and reduced delays. On the other hand, when the offered load increases, it is better to reduce the bit rate and thus obtain a higher spreading factor that allows more capacity at the expense of tolerating a longer transmission delay.

In the context of S-ALOHA-DS/CDMA, different algorithms have been proposed that deal with the bit rate adaptation according to the number of erroneous and successful transmissions perceived by a given user, which allows an estimation of the channel load [9]. At this point, it should be stated that a benefit of the ISMA-DS/CDMA centralized operation mode is the intrinsic knowledge that it provides about the channel load by means of the broadcast channel. Hence, this knowledge can be combined with other measurements, such as the number of erroneous and successful transmissions to adaptively vary the transmission bit rate.

In this section, two different adaptive bit rate algorithms for the ISMA-DS/CDMA protocol are addressed. First, the Mobile-Controlled algorithm (MS algorithm) proposed in [9] is considered. This algorithm shows some deficiencies, mainly due to the fact that it was developed for a purely decentralized random access protocol such as S-ALOHA. In order to overcome these deficiencies, a threshold based strategy, the ThMS algorithm, is proposed that takes advantage of the information regarding the number of *busy* codes broadcast by the base station.

Without loss of generality, five of the rates defined in UTRA, which are specified in Table I in terms of spreading factors and packet lengths, are considered. Additionally, the number of simultaneous transmissions x_{max} that maximizes expression (3) is given for each bit rate together with the corresponding maximum S_{max}. A closed-loop power control is assumed except for the transmission of the initial message preamble A (see Figure 4), when open-loop power control is employed. The target received power follows the criterion of having the same E_b for all the users. Hence, the received power for a user transmitting at αv b/s should be α times the received power for a user transmitting at the slowest rate v b/s.

Bit rate	S_f	L (bits)	x_{max}(users)	S_{max}(packets/frame)
v	256	160	85	69.6
$2v$	128	320	37	31.0
$4v$	64	640	16	14.3
$8v$	32	1280	8	6.9
$16v$	16	2560	4	3.7

Table I Bit rates considered

A. MS algorithm

The MS algorithm, proposed in [9] for its use together with S-ALOHA-DS/CDMA, is based on varying the bit rate according to the number of consecutive successful or erroneous transmissions for a given packet. The main parameters of this algorithm are *min_suc* and *max_tr*. Specifically, after *min_suc* consecutive successful transmissions, the rate is increased as the interference level is assumed to be low enough. On the other hand, after *max_tr* consecutive erroneous transmissions of a given packet, a high interference level is assumed to be in the system and thus the bit rate is reduced in order to obtain the higher protection provided by the higher spreading factor. This scheme is directly applicable to the proposed ISMA-DS/CDMA protocol simply by performing rate adaptations both when the code has been acquired and when the acquisition is being attempted.

The main problem with this algorithm is that, since it was initially designed for a purely random access protocol, it does not take advantage of the broadcast information to determine the most appropriate rate to be used depending on the number of *busy* codes. As a result, whenever a user starts a message transmission, it will make use of the last bit rate that was applied in its last transmitted message, or, in the case of the first access into the system, the lowest bit rate will be applied, with a cautious approach. Therefore, the initial channel estimation is performed through the previously applied bit rate, which can lead to erroneous estimations if message inter arrival time is high or if the overall offered load varies with time.

In order to study the behavior of this algorithm in an ISMA-DS/CDMA framework, results have been obtained through simulation when considering N=200 registered users sharing a total number of K=90 code sequences in a single cell scenario. Messages are generated according to a Poisson statistic and their length is exponentially distributed with a mean 4000 bits. They are divided into 160 bit packets, corresponding to the lowest bit rate, and zeros are added in order to complete the last packet. The bit rate is adapted by the MS algorithm, but the number of 160 bit packets that remain in the buffer is also taken into account to determine the final bit rate: for example, if the algorithm allows transmission at *4v* but there are only two 160 bit packets, then the bit rate *2v* will be applied instead.

According to Table I the number of simultaneous users in the system that maximizes the number of successful transmissions is 85, which is reached when all the users apply the lowest bit rate. Consequently, the regulation access method described by equation (1) is applied with parameter K_{max}=85, in order to inhibit new accesses whenever there are 85 simultaneous users.

Figure 7 presents the throughput-delay response of the MS algorithm with parameters *max_tr*=1, *min_suc*=7, which are the most suitable ones according to [9]. The corresponding curves for the different fixed bit rates are also presented. For high and medium loads, the delay response follows approximately the enveloping of the fixed bit rate responses, as is supposed to happen with an adaptive bit rate algorithm. However, for low loads, the delay is slightly longer than would be obtained with the fastest bit rate. This is mainly due to the rough estimation in the initial bit rate, which causes the algorithm to evolve until it reaches the correct bit rate. Hence, there are transmissions at slower rates that contribute to increase the message delay.

Furthermore, the main problem with the MS algorithm is its slow reaction capability in the face of sudden variations in the overall traffic load. In such cases, the initial bit rate estimation rarely reflects the real channel traffic load, as is based on measurements from an old message transmission. Consequently, in most cases the algorithm will require an evolution towards the appropriate bit rate.

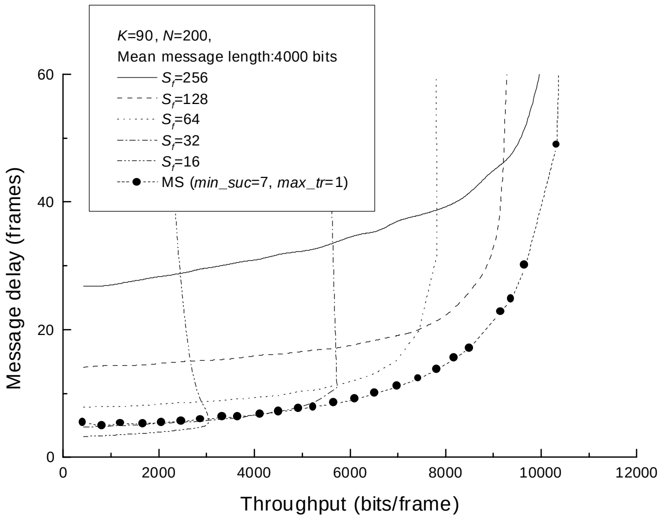

Figure 7 Throughput-delay for the MS algorithm

B. Threshold based MS algorithm (ThMS algorithm)

For the purpose of overcoming the disadvantages of the MS algorithm, the information provided by the base station can be exploited in order to decide the most appropriate bit rate to start a message transmission. This decision could be taken depending on the number of *busy* codes, *x*, according to some specific thresholds. Then, the bit rate is chosen as follows:

```
if (x<threshold_4)
        bit rate=16v;
else if (x< threshold _3)
        bit rate =8v;
else if (x< threshold_2)
        bit rate =4v;
else if (x< threshold_1)
        bit rate =2v;
else    bit rate =v;
```

Thresholds must satisfy the inequality:

$$threshold\_4 < threshold\_3 < threshold\_2 < threshold\_1 \quad (4)$$

In order to specify these thresholds, the maximization of equation (3) can be applied, which gives the maximum number of allowed users for each bit rate, presented in Table I. Taking these values would mean that, whenever there are more simultaneous users than the maximum allowed for a given bit rate, a reduction in the initial bit rate would be performed. Accordingly, given the values in Table I, the thresholds are specified as follows:

$$threshold\_4=4,\ threshold\_3=8,\ threshold\_2=16,\ threshold\_1=37 \quad (5)$$

These thresholds will provide the users with a suitable estimation of the initial bit rate to be applied at the beginning of a message transmission. Once the transmission has started, users will dynamically change their bit rate according to the MS algorithm with parameters *min_suc* and *max_tr* depending on the number of consecutive successful and erroneous packet transmissions in the subsequent frames. This mechanism improves the MS algorithm by providing it with a good initial estimation and thus it avoids the slow starting process to reach the correct bit rate. Consequently, it allows a better reaction capability in front of changes in the overall offered load.

Figure 8 and Figure 9 present the throughput and delay responses of the ThMS algorithm. A good performance in terms of delay and throughput for both high and low loads can be appreciated. This is due to the good estimation of the initial bit rate provided by the thresholds and also to the adaptation capability provided by the MS algorithm. Thus, the response of the ThMS algorithm approximately follows the enveloping of the response for the different fixed bit rates along the whole margin of offered loads considered, as it corresponds to an ideal adaptive bit rate algorithm. Additionally, note in Figure 9 that the maximum throughput obtained by this algorithm is quite close to the optimum value of 11136 bits/frame corresponding to 69.6 packets/frame (see Table I) at the slowest bit rate, where 1 packet =160 bits.

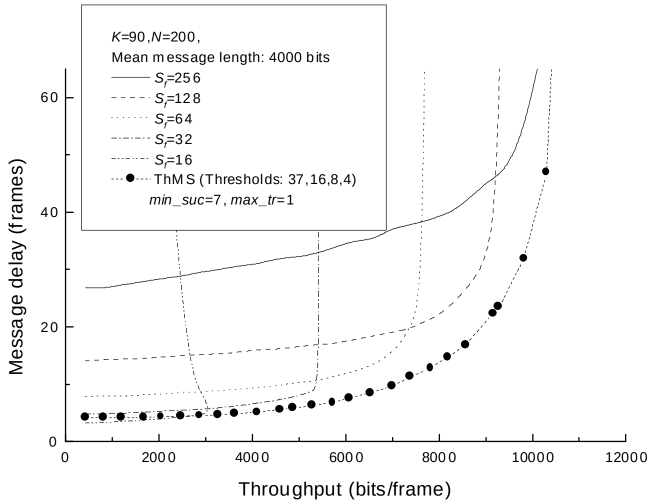

Figure 8 Throughput-delay of the ThMS algorithm

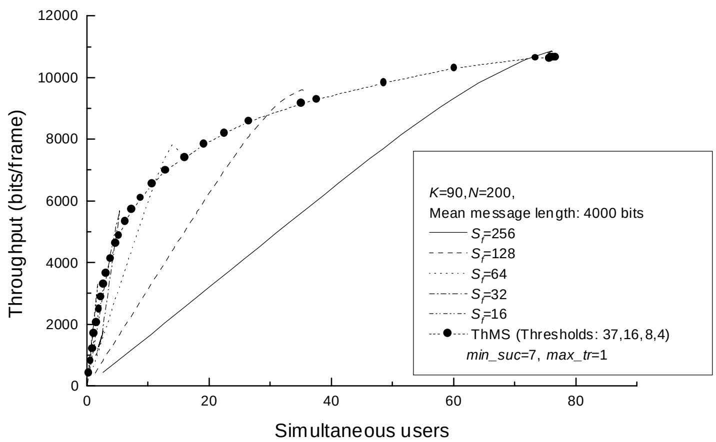

Figure 9 Throughput of the ThMS algorithm

Furthermore, Figure 10 shows the mean and the standard deviation of the bit rate applied by each of the algorithms as the number of simultaneous users increases. As was stated above, the ThMS algorithm outperforms the MS algorithm for low loads, with a higher mean bit rate that leads to a shorter message delay. But in addition to this, the standard deviation of the MS algorithm is higher, which means that, in practice, there may be many users not gaining any real benefit from the algorithm.

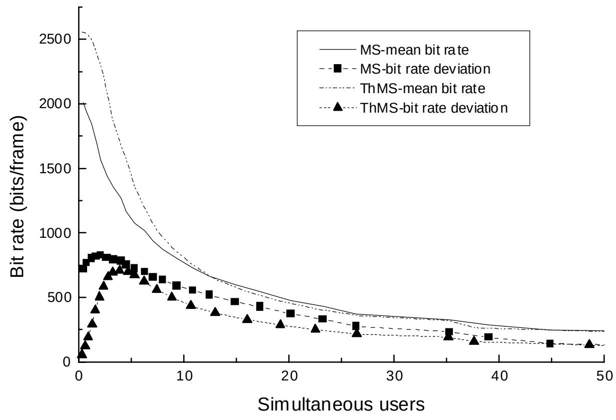

Figure 10 Mean and standard deviation of the bit rate for the MS and ThMS algorithms

IV. Conclusions

In this paper an ISMA-DS/CDMA protocol adapted to a frame structure similar to that specified in the UTRA proposal has been developed, and it has been compared with the S-ALOHA-DS/CDMA random access protocol. A noticeable improvement in terms of throughput and delay has been obtained thanks to the knowledge about the system status provided by ISMA-DS/CDMA. Some strategies designed to improve protocol performance have also been described. First, an access regulation scheme depending on the number of *busy* code sequences has been proposed in order to not let any user enter the system whenever a maximum number of simultaneous users is reached. This strategy allows an increase in system capacity closer to the theoretical upper bound. Next, two possible adaptive transmission bit rate algorithms dependent on system status have been analyzed on an ISMA-DS/CDMA basis. Particularly, the MS algorithm that was originally proposed for S-ALOHA-DS/CDMA has been compared with the proposed ThMS algorithm, which outperforms MS by applying a threshold based strategy to decide the initial bit rate for a message transmission. This mechanism allows a higher reaction capacity in the face of sudden variations in the system load and also reflects a lower standard deviation of the bit rate.

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