A NEW PROTOCOL TO DESIGN CELLULAR SYSTEMS WITH VARIABLE SPREADING FACTORS

Dr. Hayam A. Al-Yasiri
Electrical and Electronic Eng. Dept.
University of Technology, B.O. Box 35010, Baghdad, Iraq.

ABSTRACT
The emerging M-commerce (mobile +e-commerce) needs more support in operating systems in order to be successful over a wireless environment. The system needs to support a seamless integration to support voice, audio and conventional data. It should also support many users with guaranteed quality. In this paper, a new effective design protocol for Code Division Multiple Access (CDMA) mobile systems with Variable Spreading Factor (VSF) is presented, such that various Quality of Service (QoS) based on different traffic types can be provided. Increasing spreading factors can benefit the system because it will increase the desired signal strength linearly. The measured Bit Error Rate (BER) can be reduced 75 times with a long spreading factor. By taking advantage of this benefit, switching the spreading factors dynamically is proposed, based on the current load with multimedia traffic.
These proposals are implemented in mobile and base stations, and the system performance is measured. The results indicated that the proposed system can always maintain a desired quality for all voice connections.

الخلاصة
أن تزايد تجارة (النقال – الإلكترونية) يحتاج دعم اكثر للأنظمة المشغلة لكي تكون ناجحة في الأوساط ألالاسلكيه.النظام يحتاج لدعم التكامل من اجل دعم البيانات الصوتية المسموعة والعادية. كذلك يجب أن يدعم عدد كبير من المستخدمين بنوعيه مضمونه. في هذا البحث، بروتوكول تصميم جديد وفعال لأنظمة تعدد الوصوليه بتقسيم الجفرة / عريضة الحزمة، ذوات عوامل مد متغيرة تم تقديمه، بحيث ان مختلف نوعيات الخدمة المعتمدة على أنواع الإشارات يمكن التعامل بها . زيادة عوامل المد يمكن أن يكسب المنظومة لانها سوف تزيد الإشارة المرغوبة بشكل خطي. معدل الخطأ المقاس يمكن تقليله إلى 75 مرة كلما يزداد طول عامل المد . بأخذ هذا المكسب ، تم تغير عامل المد بشكل ديناميكي، اعتمادا على الحمل الحالي ضمن الحركة في الأوساط المتعددة
هذه المقترحات نفذت في محطات متنقلة وثابتة واداء المنظومة تم قياسه .النتائج أشرت أن المنظومة المقترحة دائما تبقي النوعية المرغوبة لكل الاتصالات الصوتية.

KEY WORDS
Mobile communications, Cellular systems, CDMA, Spreading factors, Design protocols.

INTRODUCTION

Wireless networks make it possible for people to shop with online anytime and anywhere. Recent technology advances are increasing multimedia capabilities in mobile devices. In a near future, most of the multimedia applications will be able to run on mobile devices.

The ideal solutions of M-commerce should also support many users with guaranteed quality. These solutions should also transmit e-mails and file transfer protocol (ftp) without connection re-establishment even in the middle of voice conversation. This application switching should be transparent. To avoid 'connection' setup overhead, the user should stay connected no matter what application is running. The aim of this work was to make these functionalities feasible.

Traditionally, voice-only-CDMA system has been designed based on the assumption of identical user channel capability. In this system, all the traffic channels are of the same data rate, and have the same BER. However, this assumption may not result a good performance when multimedia communication is taken into account. Therefore, it is worthy to investigate schemes that can provide various QoS based on different traffic types. With such schemes, the overall system performance should be improved.

One specific reason that the existing voice-only-CDMA systems are not flexible enough is because they are based on a predefined fixed spreading factor. Different researches are done in this area, some of these literature was related to multi-service CDMA development based on either interference estimation [Frenger/1999] or multi-code assignment [Choi/1997]. However, a packet scheduling protocol for slotted CDMA is proposed in [Akyildiz/1999]. Also the system performance of a Direct Sequence-CDMA when setting to a different spreading gain is studied in [Oh/1999], and evaluation of the BER has been studied extensively in [Fukumasa/1994, Lops/1998].

Accordingly, a new middle-ware protocol is needed to integrate different services efficiently (i.e. voice, data, video), which is the aim of this paper. The reasons why variable spreading factors are needed to achieve this objective is presented in the following section. The mathematical analysis of BER, necessary to assess the performance of the new protocol, is described in section 3. The a detailed description for the proposed protocol, both in the Mobile Station (MS) and the Base Station (BS), and the related design tradeoffs are presented in section 4. The tests and the obtained results are discussed in section 5, finally the work is concluded in section 6.

THE NEEDS FOR VARIABLE SPREADING FACTORS

The spreading factor of CDMA is the key variable in determining user data and associated BER. In theory, spreading factors make CDMA possible by repeating user's data signals such that they can be re-constructed at the receiving mobile stations. However, spreading factor is also responsible to produce interference called Multi-Access Interference (MAI) between different users' signals. Spreading factors can be as short as 4, or can be as long as 160 [Adachi/1997].

Increasing spreading factor can benefit BER because it will increase the desired signal strength linearly. The mean MAI caused by other users will decrease accordingly, and will approach to zero when the spreading factor approaches to infinity. However, in order to support a variety of BERs, many hardware manufacturers are considering to bring in the dynamic capability of changing spreading factor in mobile and base stations. Nevertheless, hardware capability only provides the possibility that different spreading factors can be jointly adopted. Hardware capability alone does not accomplish the desired goals. A novel middle-ware solutions are required, such that monitoring of the network is done, then switching the spreading factors dynamically based on the current load from the multimedia traffic.

In this work, the admission control protocol design is particularly addressed based on the VSF principle. All protocol designs are based on IS-95 channel architecture for backward compatibility. Because of the possible adaptation of spreading factors, a novel admission protocols are proposed in mobile and base stations. With the new protocol, it is possible that a mobile user is notified to change the spreading factor. This usually happens when a new mobile presents an OPEN request. The detailed description is given in a next section.

ANALYSIS OF BER WITH VSF

Previous work of BER analysis on CDMA systems assumed a fixed Spread Factor for all users [Fukumasa/1994, Lops/1998]. When a variable spreading factor assignment is taken into account, a new BER model for the uplink channel is desired.
Consider a single cell environment, if p^k is the spreading sequence (long PN sequence) of k^{th} user, then the spreading signal can be represented as [Tech Group/1999] :

$$p_k(t) = \sum_{n=-\infty}^{\infty} p^k(n)\delta_\tau(t - n\tau) \tag{1}$$

where $\delta(x)$ is the unit rectangle pulse with duration τ.
If u^k is the data sequence of the k^{th} user, then the modulated sequence with spreading factor of S_k is:

$$u_k(t) = \sum_{n=-\infty}^{\infty} u^k(n)\delta_{T^k}(t - nT^k) \tag{2}$$

where $T^k = S_k * \tau$ is the duration of a data bit after modulation.
Then the k^{th} transmission signal will become :

$$b_k(t) = p_k(t).u_k(t) \tag{3}$$

Assuming an Additive White Gaussian Noise (AWGN) with zero mean n(t), perfect power control, and no multipath fading environment, the received mixed signal should be:

$$r(t) = \sum_{i=1}^{K} b_i(t) + n(t) \tag{4}$$

For an interested user k, the correlator output is represented by :

$$d_k(n) = \int_{(n-1)T^k}^{nT^k} r(t)p_k(t)dt \tag{5}$$

A mixed likelihood decision rule based on $d_k(n)$ is used at the receiver, thus the decoded n^{th} bit of user k is:

$$\hat{u}_k(n) = \begin{cases} 1 & \text{if} d_k(n)>0 \\ -1 & \text{else} \end{cases} \tag{6}$$

It is clear that the bit error probability is determined by the strength of desired signal, the MAI, and the channel noise. When the signal is strong enough (large signal-to-noise ratio), it is well known that MAI is the dominant factor of decoding error. Therefore, it will be focused on signal strength and MAI in the analyses followed.

THE PROPOSED PROTOCOL

The ultimate goal of admission control protocol is to support as many users as possible while still satisfying BER requirements for all existing connections. It is clear that there are many factors that can influence the performance of the call depending on the duration of the call. It is even possible that during a long call, the system will accept and terminate a number of calls, thus the number of users will alter very often. Those alterations will affect BER at any given time. Thus the admission control protocol needs to be adaptive for these changes. As the system admits new calls (thus increasing the number of active users), system will dynamically change the spreading factors, if necessary, for certain traffic.

Both MS and BS need to participate in the call admission process. MS's are the ones that make the requests and/or update their parameters to adjust themselves to changing environments. The environment change includes increase/decrease in the number of users and alterations in traffic types.

The Proposal in the Mobile Station

Fig. (1) depicts the state diagram of the proposed protocol for the mobile stations.

In traditional CDMA systems (IS-95) [Stavroulakis/2000], MS's can have three types of request: OPEN a new connection, ALTER the traffic type, and CLOSE the connection. The first two request types follow the similar steps except the fact that altering the traffic type does not change the number of users. The application in MS sends an OPEN request to BS on the access channel. Along with the request the type of the traffic (and the desired minimum data rate, if required) is sent. The proposed protocol defines four traffic types (VOICE, AUDIO, VIDEO, and DATA) and multiple data rate options for each traffic type. IS-95 does not allow mobile to specify the desired traffic type. The detailed proposal is illustrated in **Fig. (1).**

The Proposal in the Base Station

Fig. (2) depicts the state diagram of the proposed protocol for the base station. The major contribution to the new protocol design is covered with a dashed line. If it determined that the new connection can be satisfied, the BS moves further to check if existing connections can be satisfied once the new connection is up. This facility is not available in IS-95 CDMA. However, it is added to the new protocol to ensure quality of all users. For each existing connection, the system calculates the expected average BER corresponding to the increased number of users. If the expected average BER is high, BS tries to increase the spreading factor and check the maximum data rate in order to see if data rate requirement can also be satisfied with increased spreading factor. This step is repeated till all existing connections are reviewed.

Design Tradeoffs

The added and/or improved functionalities bring new challenges to the implementation of the protocol, as well as better performance of data communication. Two design tradeoffs were the consequences of applying the new protocol; these are: the long processing time required at the base station, and the contention time required for all mobile stations to acknowledge the accomplishments of changing spreading factors. These two timing factors result in longer end-to-end connection setup time.

The end-to end delay is defined as the total time between a user making a connection request and the user becoming ready to transmit data. It can be formulated as:

$$T_{ete} = T_m + T_p + T_d + T_{ud} + T_a + T_s \tag{7}$$

where

T_m = Time to send request from MS to BS using access channel.

T_p = BS processing time, includes finding SF and checking existing connections.

T_d = Notifying destination and receiving ACK from destination.

T_{ud} = Broadcasting UPDATE message, and receiving ACKs.

T_a = Sending ACK to requesting MS using paging channel.

T_s = MS updates channel parameters and becomes ready to transmit.

On the first design tradeoff, the BS has a high computation complexity (T_p). While the performance of all connections is guaranteed, the BS processing time increases proportionally to the traffic load of the system. The reason is that the BS checks every existing connection one by one. It also has to do some computation, like finding appropriate SF, calculating the data rate, and doing comparisons. It has to repeat these operations for all connections if an update is necessary.

In general, each change of SF will result in a higher computational time than the previous one because more existing connections need to be examined and informed. Fortunately, the process time in the BS increases with a slight angle for most of the time. The reason is that, if the current SF for existing connections is large enough, there is no need to decide new SF for those connections.

Reduction the Processing Time

Throughout the test performed on the protocol, it realized that the connections belonging to the same traffic type usually change to the same SF. This observation shows that the average BER is not affected by the type of the traffic. All external signals from other types of traffic are treated as noise no matter what type of traffic they carry. Thus, connections with the same BER requirement (same traffic type) are recommended to use same SF.

Originally the proposed protocol calculates SF connection by connection in case of an update. There are two operations performed during each review, finding SF and checking data rate satisfaction. The protocol can be further improved by calculating only one SF for each traffic type before starting the whole review. This improvement saves an extra operation (without affecting the protocol correctness) during the review of each connection.

Reduction the Contention Time

The second tradeoff is the T_{ud} component. The dominating factor of T_{ud} is the possible UPDATE broadcasts. This tradeoff is affected when existing contentions require a change of SF (thus an update for overall system parameter). The system design goal is to provide good BER performance for all existing connections at any time. Therefore, before accepting the new connection, all active MS should switch to the new SF so that the BER never increases above the acceptable point. Originally, system design enforces that MS's should send acknowledgments back to BS after they change their SF's. Since access channel of a CDMA system has only one common channel, this common channel is shared by all MS's. Thus, there will be extra delays due to possible collisions, if more than one MS need to access the common channel.

The proposed protocol design in the BS will wait until it receives all acknowledgments from the MS's. Therefore, the duration time that after all collisions are resolved, plays a very important role in the end-to-end time T_{ete}. In general, T_{ete} seems to be determined by the combination of two dominating time components, T_p and T_{ud}. T_p is treated; however T_{ud} does not occur all the time. For instance, T_{ud} is 0 when no update is needed. Since the BS does not broadcast UPDATE message,

MS's thus do not required sending back any acknowledgments. Therefore, the total end-to-end delay grows slowly when no updates are needed.
After analyzing the overall system behavior, it is discovered that the UPDATE delays can be reduced by decreasing the contention period. One of the approaches that can potentially decrease the contention period is increasing the number of available access channels for MS's. Since more than one access channels are available, all MS's need to be assigned to a specific access channel when they are admitted by the BS. BS should attribute access channels among all users equally, thus the overall T_{ud} can be reduced evenly across all MS's. A hash function implemented in the BS should be sufficient for this purpose. By taking a MS's serial number or PN, the number of mobiles using each access channel can be balanced. The advantage is that more access channels means smaller contention slots, thus shorter delays.

RESULTS AND DISCUSSION

The system necessary to assess the proposed protocol performance has been simulated by MATLAB. The Gold PN sequence from IS-95 is adopted. The tests were performed such that a single cell is considered. Interference from neighboring cells is neglected. Multipath effects and thermal noise were not taken into account since emphasize is considered on the effect of spreading factor only.
First the how spreading factors and number of active users affect the BER has been investigated. **Fig. (3)** depicts the BER under different number of active users and spreading factors. It is clear that the increase of spreading factor can effectively decreases the BER for a given number of users. For example, with 10 users, increasing the spreading factor from 64 to 96 will reduce BER of about 62.5%.
To assess the performance of the proposed protocol, BER was measured in every MS, and the average BER among all voice streams was calculated and illustrated in **Fig. (4).** As depicted by the curve labeled as "SF32", fixed SF schemes did not guarantee the over all performance among all the users. When the connections are less than 5, the average BER was acceptable (lees than 10^{-2}). However, when the number of connections exceeded 5, every connection (including existing and the newly accepted connections) suffered with the BER quality above 10^{-2}.
On the other hand, by using the proposed admission protocol, the average BER was always maintained below 10^{-2} threshold for voice even as the number of users increased. The curve labeled with a sequence of (SF64, SF96, SF128, SF160) in **Fig. (4)** stated the time instants that the proposed CDMA system re-acted to the increasing demand of users, and new spreading factors have been adopted. The system never exceeded the upper limit for the voice traffic (10^{-2}).
The design tradeoff represented by the long processing time (T_p) at the BS, and the improvement made by reduction this time, are investigated through some tests, the obtained results are shown in **Fig. (5).**
For the case of 5 connections, the processing time has been reduced from 22 msec to 17 msec (i.e. 22.7%). The improved degree can be more significant for the cases of 9 and 14 connections, the reduction reaches to 26.5% and 27.1% respectively. The achieved improvement in T_p can be considered close to the optimum that a system can reach. It is necessary that the system reviews each connection since each of them may have different data rate requirement. Nevertheless, T_p time is reduced during the review to a minimum, which is now only checking the data rate satisfaction.
The second tradeoff (reduction the contention time) is treated by increasing the number of access channels. **Fig. (6)** illustrates these effects. The preliminary results demonstrate very promising results on reduction the T_{ud} component. By using 2 access channels, the average T_{ud} of 5 connections have been reduced from 221 msec to 112 msec (i.e. 49.3 %) while it reduced from 612 msec to 308 msec (i.e. 49.%) when 14 connection are supported. Accordingly, it is clear that using 2 access channels T_{ud} can be reduced 49% stably independent of the number of existing connections.

However, with 4 access channels, T_{ud} can be further reduced. By increasing the access channels from 2 to 4, a reduction about 48% to 58% of T_{ud} is accomplished.

CONCLUSIONS

The presented protocol admits new calls only if the new connections' as well as all the existing connections' quality can be guaranteed. It uses variable spreading factor scheme. In order to satisfy the BER requirement, the protocol tries to dynamically change the spreading factor used by connections that may experience a high BER. The connections may alter their SF several times depending on the changes in the number of the users in the system.

The BER performance and admission time of the protocol, have been evaluated under an increasing number of voice connections. Also, the proposed protocol is compared to the regular CDMA protocol that uses a fixed spreading factor. The results show that the VSF protocol provides significant improvement in BER satisfaction compared to CDMA with fixed SF. Even though the number of users increases in the system, the average BER of all connections are maintained below the upper limit for their traffic type.

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LIST OF SYMBOLS AND ABBRIVATIONS:

AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BS	Base Station
b(t)	Transmitted signal

CDMA	Code Division Multiple Access
d(n)	Correlator Output
ftp	File transfer protocol
MAI	Multi-Access Interference
MS	Mobile Station
n(t)	noise signal
PN	Pseudo-random Noise
P(t)	Spreading Signal
QoS	Quality of Service
r(t)	Received signal
SF	Spread Factor
Sk	Spread factor of user k
T	Duration of data bit after modulation
U(l)	Data sequence
VSF	Variable Spreading Factor
τ	Time duration
δ	Unit rectangle pulse

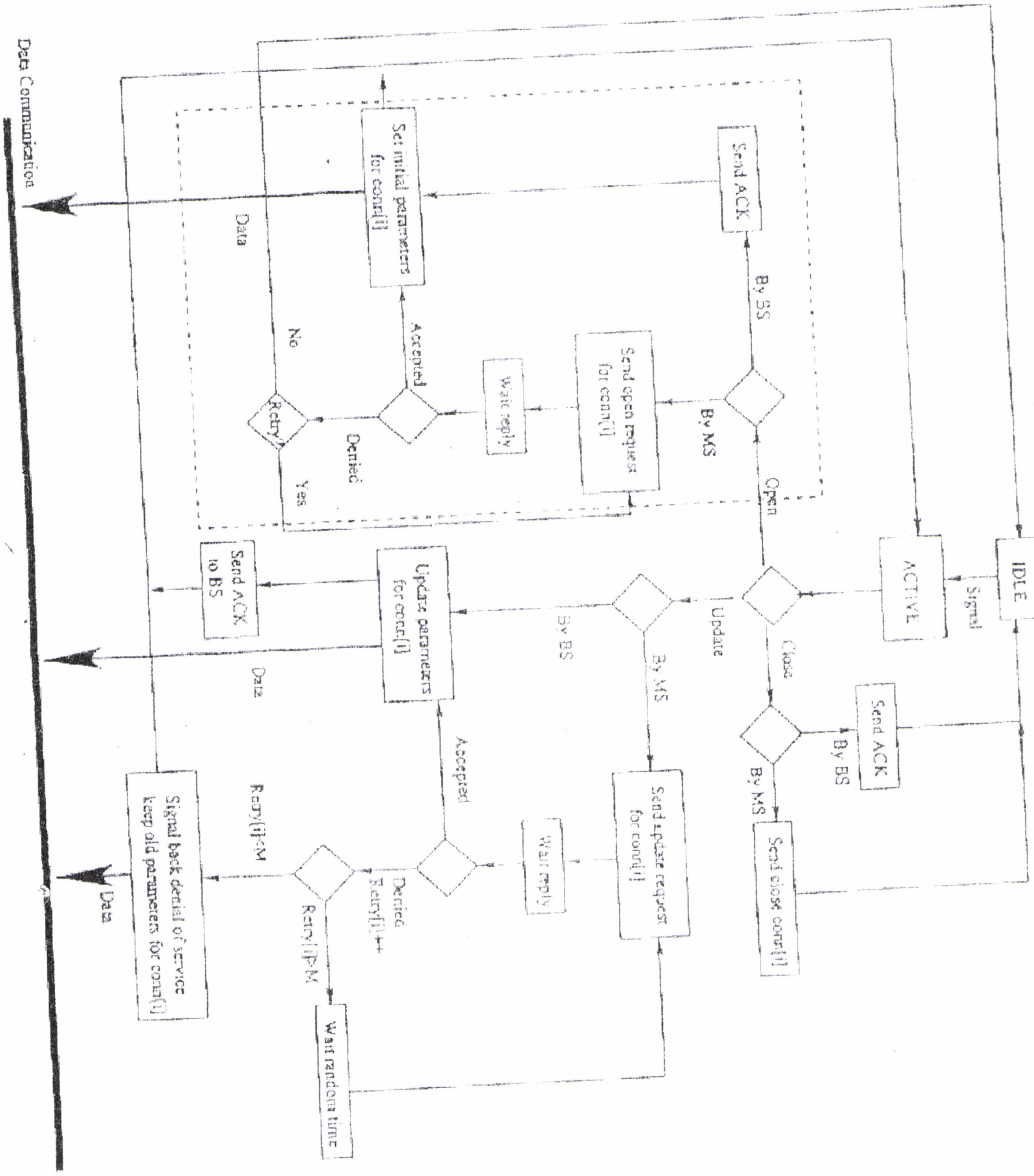

Figure 1: The state diagram and protocol of a mobile station.

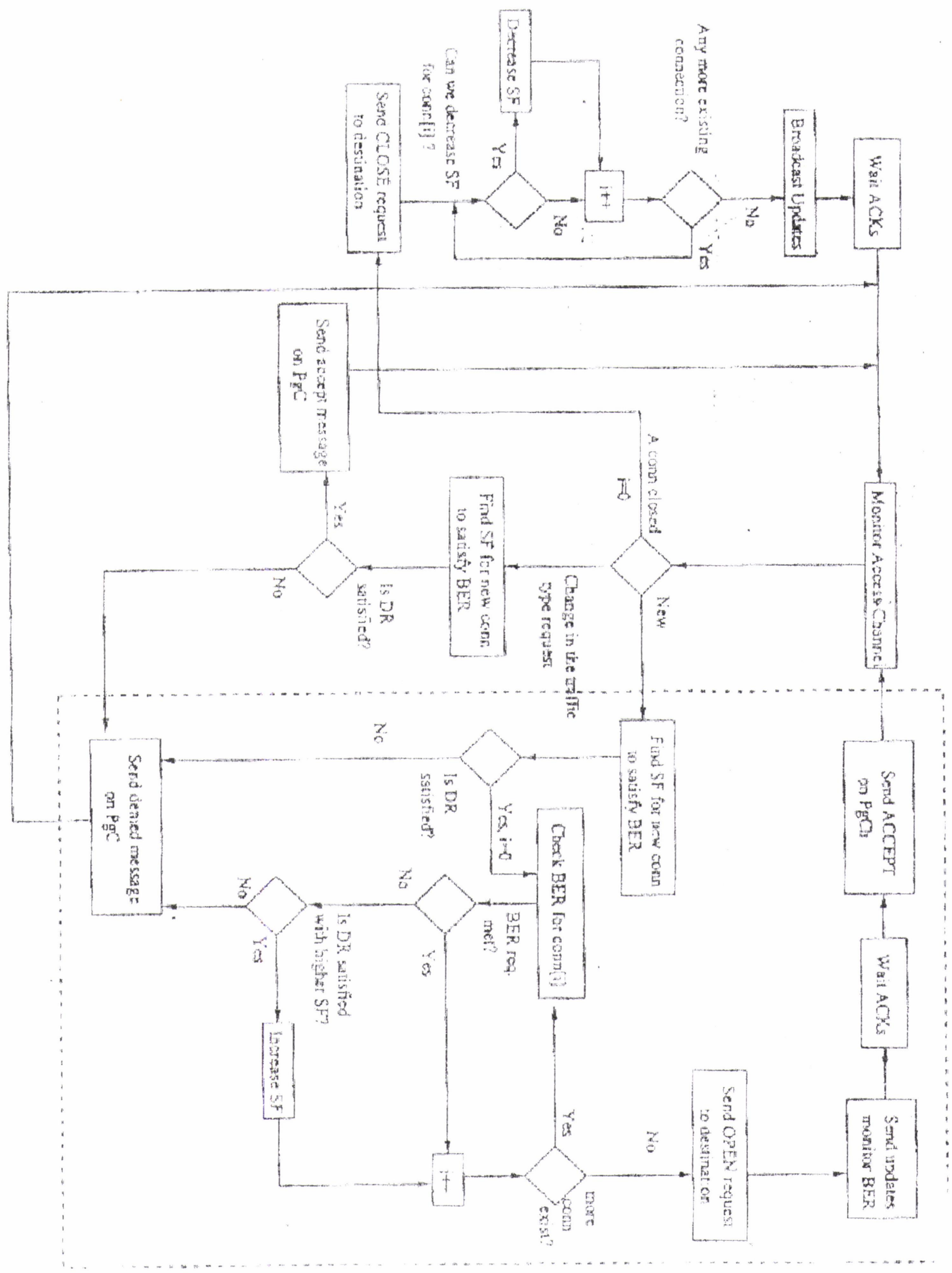

Figure 2: The state diagram and protocol of a base station.

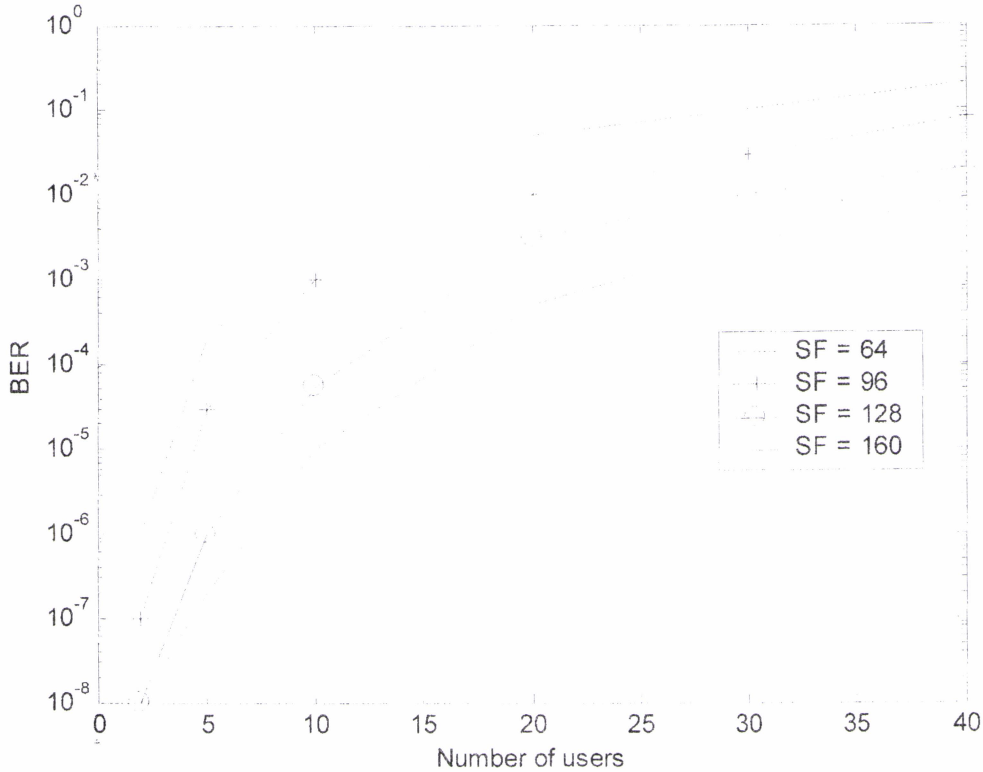

Fig. (3) BER performance versus users number for different spreading factors.

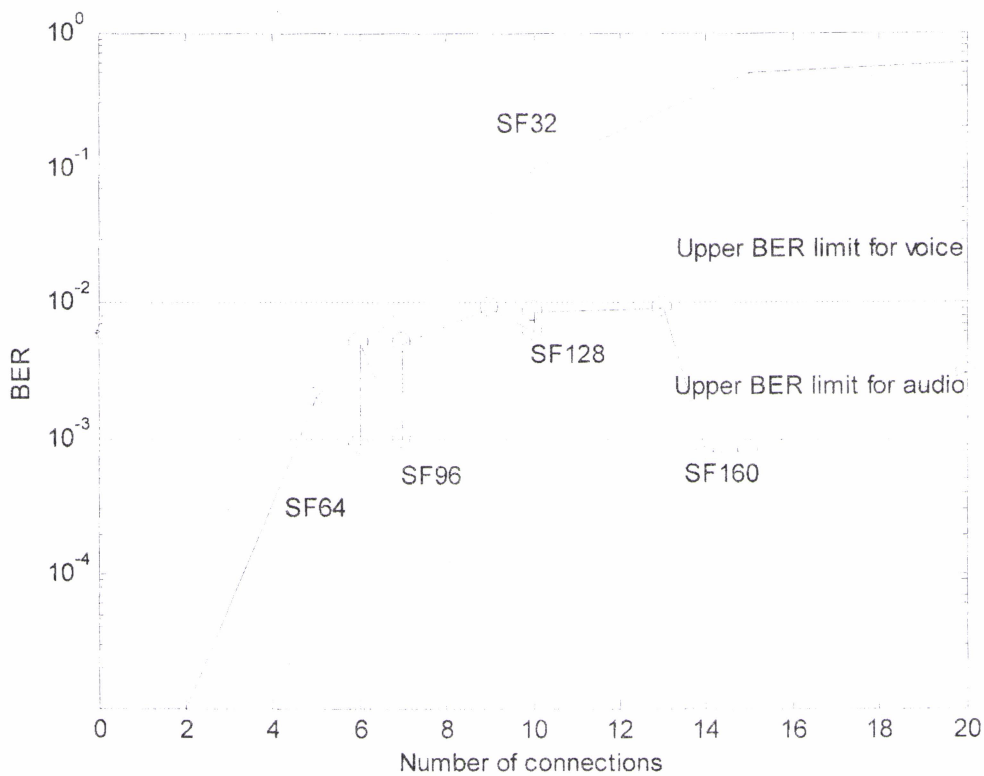

Fig. (4) Performance guarantee with the proposed protocol.

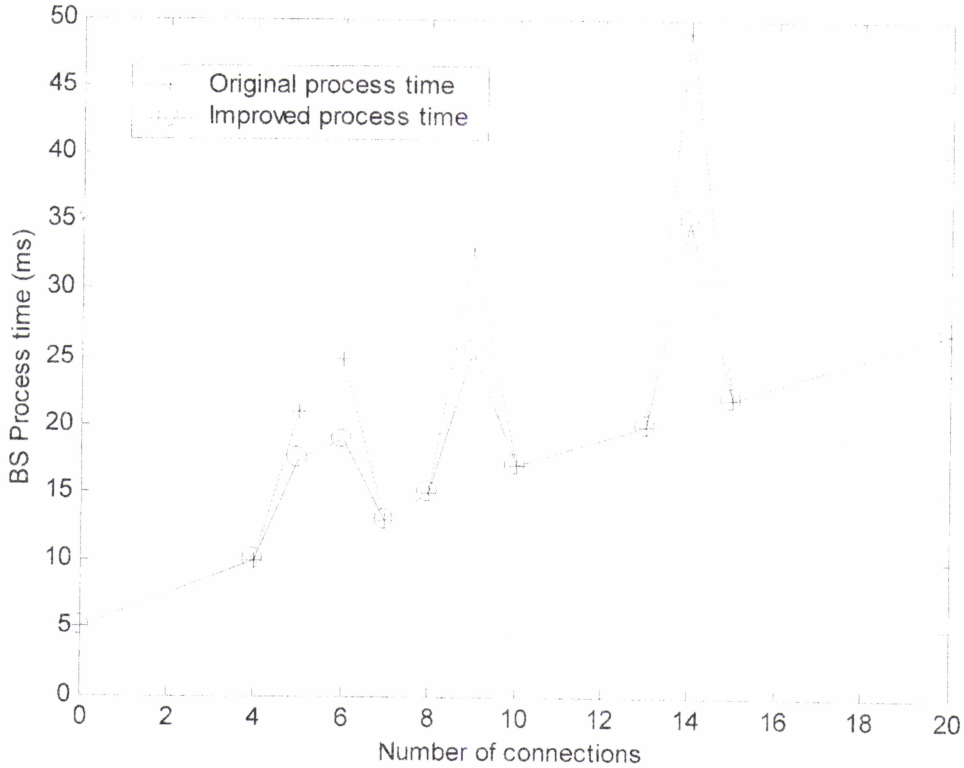

Fig. (5) Improved process time.

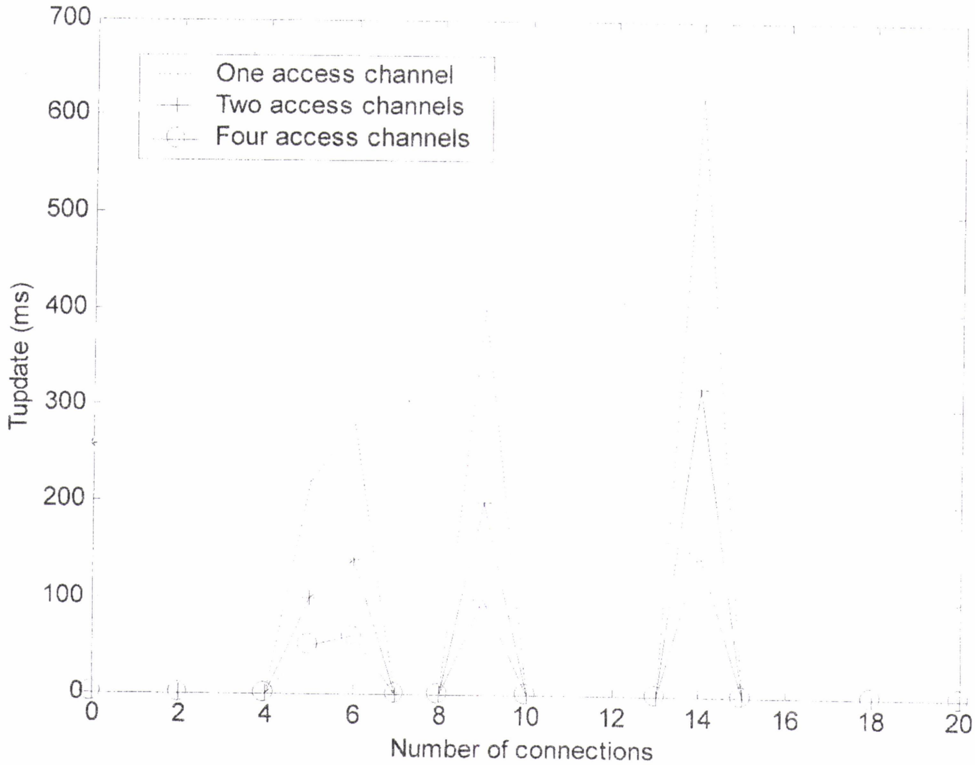

Fig. (6) Improved T_{update} by using multiple access channels.