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Packet Reservation Multiple Access Protocol for Digital Cellular Mobile Communications: A Performance Study

Mohammad J. Abedin
Department of Computer Engineering,
College of Computer and Information Sciences,
P.O. Box 51178, Riyadh 11543, Saudi Arabia.

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Abstract. Digital cellular mobile radio communications techniques is an area of wide public acceptance. This paper looks onto the performance of packet reservation multiple access (PRMA) protocol for the transmission of digital speech packets from mobile speech sources. Packets from active speech sources contend for access to a channel time slot. Computer simulations were done to study the performance of the protocol and to determine the capacity of channels having different bandwidth and frame size.

1. Introduction

Mobile radio telephone is a device which provides the user with the features of an ordinary telephone, connected to the telephone network, except that it does not have wires and can be used anywhere. Low power digital radio telephony could be an alternative for providing access to land-based public telephone networks due to its widespread portability. There is continuous effort throughout Europe, USA and Japan to develop different types of radio communication systems both for voice and data communications [1,2].

The most familiar means of wireless access to fixed networks is the cellular system of communication. In this system, transmissions are confined to small areas referred to as cells. The geographical area is divided into a large number of small cells and the user at any cell can initiate calls. He can move from one cell to another while speaking without any interruption of the call. Call set up and hand off are generally performed by a mobile telephone switching office (MTSO).

The high capacity of a cellular system is due to the fact that any wireless channel can be used simultaneously at different cells. The use of smaller cells provides the opportunity to reuse the frequency spectrum more often and hence increase the traffic density and spectrum efficiency.

First generation cellular radio transmission technology using analog devices have severe limitations due to lack of flexibility and vulnerability to interference. This happens due to poor network control systems, as the control signals are transmitted in the same channel. The second generation systems employing digital transmission techniques have dedicated channels for the exchange of network control information between the mobile and the base station. Several such wireless access systems are now in use [3,4].

It is expected that the future cellular radio communication systems shall be able to carry many types of information and be able to serve a very high user population. Voice-band data can be transmitted over cellular mobile systems and an integration of data and voice in digital radio access technology would be more appropriate and have similar trends such as ISDNs and B- ISDNs.

This paper has been organized as below. Section 2 describes about the operation of a cellular system. The frequency reuse mechanism is described in section 3. The packet reservation multiple access protocol (PRMA) is introduced in section 4. The performance of the protocol is explained in section 5, while the simulation is presented in section 6. Section 7 concludes the paper.

2. Operation of Cellular System

When a user in a cellular mobile unit activates the receiver of the mobile unit, the receiver senses a number of set up channels to select the strongest channel and locks on for a certain time. Since each site is assigned a different set up channel, locking onto the strongest set up channel usually means selecting the nearest cell site. The user then places the called number into an originating register in the mobile unit and presses the 'send' button. The cell site receives (base station) it, and selects the best antenna for the voice channel to use. The MTSO is called by the cell site and an appropriate voice channel is selected for the call. The MTSO also connects the wireline party through the telephone zone office. Conversely when a land-line dials a mobile unit number, the telephone zone office recognizes that the number as mobile and forwards the call to the MTSO. The MTSO sends a paging message to certain cell sites based on the mobile unit number and the search algorithm. Each cell site transmits the page on its own set up channel. The mobile unit recognizes its own identification on a strong set up channel, locks onto it, and responds to the base station. The mobile unit also follows the instructions from the cell site to tune to the assigned voice channel.

During the call, both parties are on a voice channel. When the mobile unit moves out of the cell boundary the reception becomes weak. The present cell site requests a hand off and the system switches the call to the new frequency channel in a new cell site without interrupting the call. The call continues as long as the user is talking. The user does not notice the hand off occurrences. When the mobile user turns off the transmitter, a particular signal transmits to the cell site and both sides free the voice channel. The mobile unit resumes monitoring pages through the strongest set up channel.

3. The Frequency Reuse Scheme

The efficient spectrum utilization in cellular mobile radio system depends on the efficient frequency reuse mechanism and the reduction of co-channel interference. Since the limitation in the system is the frequency resource, the challenge is to serve the largest number of customers with an acceptable performance quality. In a cellular radio system the same frequency is used repeatedly in a geographic area. There are many co-channel cells in the system. The total frequency spectrum allocation is divided into K frequency reuse patterns as illustrated in Fig. 1 for K = 4 and 7 [5].

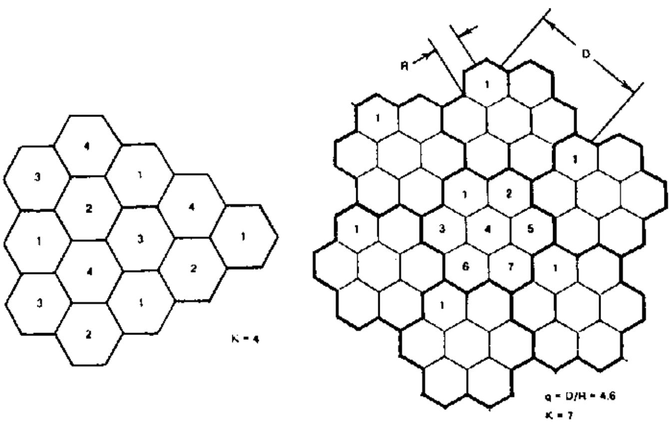

Fig. 1. Frequency reuse patterns in a cellular system for K = 4 and K = 7.

The minimum distance which allows the same frequency to be reused depends on many factors, such as the number of co-channel cells in the vicinity of the center cell, antenna height, transmitted power etc...

The frequency reuse distance D can be determined from [5] as

$$D = \sqrt{3K}\,R,$$

where K is size of the frequency reuse pattern and R is the cell radius (Fig. 1).

Thus $D = 3.46\,R$, for $K = 4$ and $D = 4.6\,R$ for $K = 7$.

Theoretically a large K is desired. When K is too large, the number of channels assigned to each of the K cells becomes small, thus decreasing the spectrum reuse efficiency. The selection of a smaller K, keeping the system performance requirements, involves estimating co-channel interference and selecting minimum frequency reuse distance D to reduce the interference. For a system with shift parameters i and j [6], K can be written as

$$K = i^2 + ij + j^2 \text{ (Fig. 1)}, \quad \text{for } i = 1 \text{ and } j = 2, K = 7.$$

The maximum number of frequency channels per cell, N, is related to the average calling time in the system. If Q_i be the maximum number of calls per hour per cell with T as the average holding time in minutes then the offered load A is given by

$$A = Q_i \, T/\, 60 \text{ Erlangs.} \tag{1}$$

For a blocking probability B, the number of channels N can be obtained from tables for Erlang B loss system [6]. For example if we let Q_i = 1000 calls/hour, T = 3 minutes, and B (the blocking prob.) as 2%, then from equation (1), the offered load A = 50 Erlangs. The entry in the Table [6] for A = 50 Er and B = 2 % is N = 60. For a seven-cell reuse system (K = 7) the total number of channels is N = 50 × 7 = 350 radios. The total number of subscribers per cell, M_i, is related to the percentage of car phones used in busy hour and the number of calls per hour.

In case the same frequency band is used many times, the capacity of the cellular system rises manifolds. The following example makes the point clear. Let us consider the city of Riyadh, Saudi Arabia, with an approximate area of 1600 Km^2. If the city is divided into cells of radius 2 Km each, the total number of cells become 127 (approx.). If the frequency reuse pattern K is taken as 7, the same frequency band can be used 127/7 = 18 times. If a total BW of 10 MHz is available for the mobile radio transmission and if each voice channel occupies a frequency band of 25 KHz then the total number of channels accommodated is 400. These 400 channels can be distributed into 7 cells and be reused 18 times in the cellular system. Thus there is an 18 fold increase in the total number of channels that can be used simultaneously i.e. 7200 channels. If the cellular system is not used the capacity remains equal to the number of channels available from the BW allocated i.e. 400. For a blocking probability of 2% these 400 channels can carry a traffic of 389.9 Erlangs [6]. This leads to a Q = 11577 calls/hour.

If this system is replaced by the cellular system as described above, the capacity increases to Q = 11577 × 18 = 208,386 calls/hour in total. If we take a reduction factor of 0.7 due to the vacant areas inside the city, the traffic capacity still remains very high i.e. Q = 146,000 calls/hour. In terms of the amount of traffic carried it becomes (from equation 1.) 4866 Erlangs. This is at least 12 times higher than the system where cellular system is not used. The above study shows that a properly designed cellular system can easily meet the demands of mobile telephone systems, in a large city like Riyadh, without causing any traffic congestion.

In a telephone conversation a speaker is active about 43% of the period of a call while the rest of the time the speaker remains silent [7]. In a completely analog environment the capacity of a channel can be doubled if the channel is assigned to another speaker in the silent period. The TASI (Talk Assignment Speech Interpolation) system [8] is utilizing this technique to increase the system capacity by assigning the channel at silent period to another talker.

It is also possible to improve system capacity by utilizing the silent period of a talker and assigning the channel to another talker currently in talk spurt, when a digital multiplexing of speech packets is done [8]. Several smart algorithms have been proposed in mobile telephony to improve the system capacity by utilizing the above technique. It is even possible to mix traffic of different types i.e. voice and data traffic to share the system capacity efficiently. In the next section we shall study the performance of such a protocol and see how it can be used to carry speech packets in a cellular mobile system.

4. PRMA Protocol

Packet contention techniques such as ALOHA, CSMA/CD [9,10] find extensive use in data communications. These techniques can serve a large number of terminals without any central co-ordination. They have an excellent throughput-delay performance at low to medium load, while the delay rises sharply at high offered load due to excessive packet collisions. PRMA is a random access technique closely related to reservation ALOHA protocol [11] and is suitable for short range radio channels.

The PRMA protocol [4] is organized around time frames with durations matched to the periodic rate of voice packets. Time is divided into equal length frames and each frame can accommodate a group of time slots. A voice terminal can reserve a time slot at a frame and can reuse it at successive time frames as a TDMA fashion. Once a time slot is reserved, a voice terminal transmits packets of equal length at each frame during the talk spurt and releases the slot at the end of the talk spurt. Speech activity detectors can be utilized to mark the start and end of a talk spurt.

The use of PRMA in the mobile radio environment can be explained as follows. Two different carriers are used to transmit the packet from base station to the mobile and mobile to base. The up channel is used to transmit speech packets from mobile to base using a slot in a frame. The down channel is used to transmit information and data frames from base to mobile which can also include acknowledgement packets. A one bit slot reservation field informs that a particular slot is reserved. When a talk spurt begins, the mobiles contend for an available slot. Once a slot has been acquired i.e. a packet has been successfully transmitted, the slot in successive frames is reserved and there are no subsequent collisions for that particular slot from other mobiles. At the end of the talk spurt the mobile releases the reserved slot by declaring the slot empty. At the end of each slot the base station broadcasts the feedback information that reports the result of the transmission. When two or more mobiles contend for the same time slot, the base station is unable to detect any packet from them. All colliding mobiles must retransmit in this case. An unsuccessful mobile continues to retransmit in subsequent slots with a permission probability, p, ($p < 1$) until the base station acknowledges the successful reception of the packet. The permission probability, p, is the probability of transmission of a packet from a mobile in the next available slot and is kept same for all mobiles [2].

5. PRMA Performance

As PRMA is a multiple access protocol closely related to R-ALOHA, the study of S. Lam [11] can be used directly for the performance study. In both the protocols time is divided into equal length frames, each frame being divided into M slots. If a particular slot m had a successful transmission by user X (say) in the previous frame, then X is the exclusive user of the current slot. Those slots which are unreserved in the last frame are available for contention in the current frame.

The channel utilization U for the case of single message users i.e. the Poisson source shuts itself off until all packets of the current message have been transmitted is given by

$$U = N\lambda\bar{h}T / (M\ (1+\lambda\bar{h}T)) \tag{2}$$

where N is the number of users, λ is the arrival rate, $\bar{h}$ is the mean number of packets in a message, T is the frame duration in seconds and M is the number of slots in a frame. Equation (2) above can be used for the calculation of the PRMA channel throughput in case of voice traffic.

Let us consider the case of N voice sources generating alternate talk spurt and silence intervals of exponential distributions. Each talk spurt is divided into speech packets of fixed length and transmitted at equal intervals. If $1/\alpha$ and $1/\beta$ be the lengths of mean talk spurt and mean silence intervals respectively, then the mean interarrival time $1/\lambda$ of the messages can be considered as $1/\alpha + 1/\beta$. This assumption is justified, as the expectation of the sum of two independent random variables is the sum of their expectations [12].

With N = no. of simultaneous talkers, λ the arrival rate, T the frame duration, $\bar{h}$ mean number of speech packets per talk spurt and M number of slots per frame equation (2) can be used for the calculation of voice throughput for the PRMA channel. For example, if we take $N = 22$, $M = 8$ slots, $T = 15$ ms from, $\lambda = 1/3$ messages/sec and $\bar{h} = 89$ packets/message ($1/\alpha T$) then from equation (2) The throughput U becomes 0.84 packets/slot. Here the message interarrival time $1/\lambda$ is taken as $1/\alpha + 1/\beta = 1.34 + 1.66 = 3$ seconds [7].

The average delay d is given in [11] by

$$\bar{d} = \bar{d}_a + (\bar{h} - 1)\ T \tag{3}$$

where $\bar{d}_a$ is the mean access delay for the first packet to transmit. The calculation of $\bar{d}_a$ depends on several factors such as the waiting time of a packet after arrival until the beginning of the next slot, the delay due to retransmission, the packet transmission time and the propagation delay. In Lam [11] this delay $\bar{d}_a$ is given by

$$\bar{d}_a = \left(1 + (1 - q) / pq\right) / (1 - U) \tag{4}$$

where p = 0.2 or 0.1, $q = e^{-S/q}$ and S is the slotted ALOHA throughput. Equation (3) can be modified for voice traffic and can be written as $\bar{d} = \bar{d}_a$ (same as equation 4). This is because once the slot is reserved by successfully transmitting the first packet in a talk spurt, the rest of the packets are transmitted periodically after the packetization delay, which is set as T ms, the frame duration. Thus equation (4) can be used to calculate the mean delay for voice packets in the PRMA protocol.

For the single message user case the slotted ALOHA throughput S is obtained from [11] as below.

$$S = \left(\lambda T (N - MU)\right) / \left((1 - U) M\right) \tag{5}$$

With U = 0.84 packets/slot, λ = 1/3 messages/sec, T = 15 ms, N = 22 callers and M = 8 slots, equation (5) gives S = 0.0597 which in turn provides $q = e^{-S/q} = 0.935$. Thus for p = 0.5, q = 0.935 and U = 0.84, the mean delay $\bar{d}_a$ becomes 7.119 slot times.

As the loss of a small percentage of speech packets does not degrade the quality of speech noticeably, we are interested in looking at the system performance with certain percentage of packets lost. Later on in our simulation study, we shall count packets which are waiting for a period more than the speech packetization time, as lost packets and remove them from the queue. This arrangement will limit the maximum packet access delay to the packetization time i.e. the frame duration. This means that the upper bound of d_a will be equal to T, the frame duration. It is to be noted that equation (4) can only be used in a lossless system i.e. at low voice load.

Our main interest of performance study is now to determine the PRMA channel capacity, in terms of number of simultaneous conversations the system can accommodate at different bit rates of the multiplexed channel, keeping in mind that we allow a small percentage of packets to be lost. This is to keep the delay within limit and hence to maintain the real time behavior of speech.

6. Simulation of PRMA Protocol

We studied the performance of PRMA protocol in case of voice traffic. Voice packets are generated from speech sources having alternate talk and silence intervals. The mean duration of talk spurt and silence intervals are taken as 1.34 sec and 1.66 sec respectively according to the measured distributions of P.T. Brady [7]. Voice packets are formed by collecting speech samples of duration 10, 15 and 30 ms.

Header bits are added at each voice packet before transmission. The length of each frame is set as equal to the voice packetizing time. Each voice source contends for a transmission with the first voice packet in to the medium. The permission probability, p, for transmission in the next available slot is chosen as 0.5 in our simulation. Once a slot is acquired successfully, that slot is reserved for successive voice packets to be transmitted in each frame until the end of the talk spurt. At the end of the talk spurt a silence interval is generated and the slot is released for other users. A voice sampling rate of 32 Kb/sec has been taken for our study.

As voice is a real-time traffic, it cannot tolerate long delays that may occur in a random access channel. Also, as the voice source generates packets at regular intervals, a long delay with a particular packet can cause an accumulated delay for successive packets. To minimize the packet delay, we considered the mechanism of packet dropping when packets are delayed more than the frame time. Once an old packet is dropped the source starts transmitting the newly generated packet until a slot is reserved successfully or the packet is again dropped due to excessive delay. This phenomenon occurs at the beginning of the talk spurt. Once a slot is reserved by a voice source, the slot is exclusively used by that source only.

A simulation program, based on the technique of continuous time advance mechanism, has been written for the study of performance of the PRMA protocol. The program advances the time, as it scans each slot in a frame, and records the activities within the slot. A simplified flow chart of the simulation model for the PRMA protocol is given in Fig. 2. Inputs to the program are bit rate of the channel, speech encoding rate, speech packet length, frame size, number of conversations, cell radius and the simulation time. During the simulation period each voice source alternates from talk spurt to silence and vice versa. The simulation was run for a sufficiently long period of time to include a large number of talk spurts and silence intervals from each talker. For each run of the program, the number of conversations in the systen was kept fixed while the variation of talk spurt and silence intervals were considered. This arrangement is justified as the variation of talk spurt and silence intervals is more frequent than the variation of number of calls in the system due to calls entering and leaving the system [8].

The output is the percentage of packets lost when the number of conversations is increased at steps to see the effect of loading the network by voice traffic. The performance of the PRMA protocol, with packet dropping due to excessive delays, causes voice packet delays bounded to an upper limit. We kept the percent packet loss below 1 % by not allowing new calls to initiate.

Voice carrying capacity of several digitally multiplexed channels were evaluated. The bit rates selected were 320 Kb/sec, 640 Kb/sec, 1Mb/sec and 5 Mb/sec. Different time frames were set and each frame was divided into a group of slots. The results are shown graphically in Figs. 3-6. The Figures show that with the increase in frame size, the voice carrying capacity of the channels also increase. Depending on the traffic intensity at each cell a specific multiplexed channel can be used. A high bit rate channel will eventually accommodate a large number of conver-

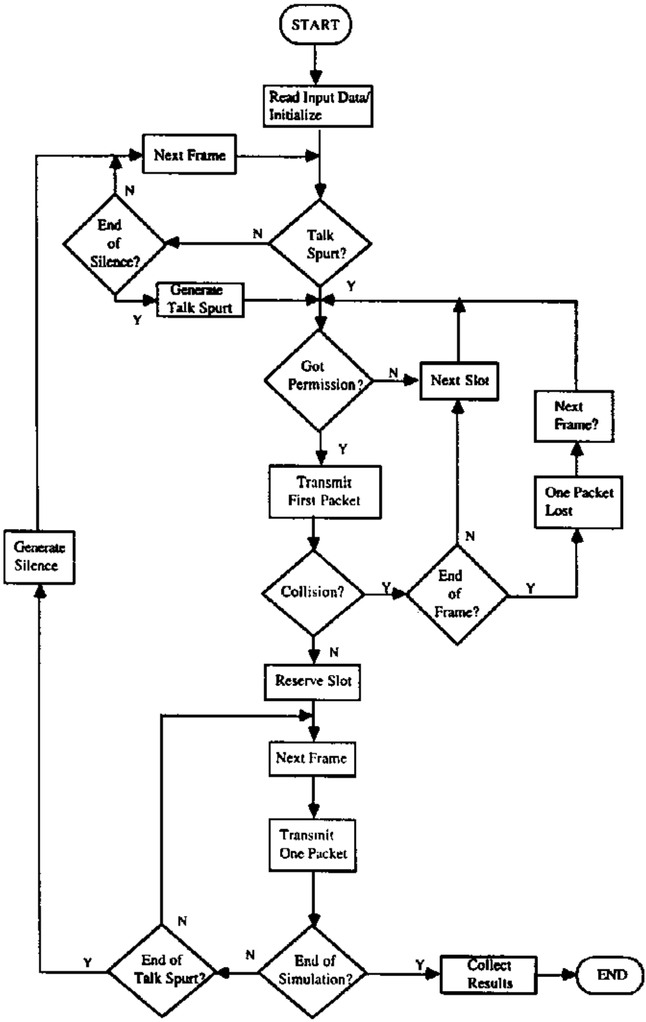

Fig. 2. A simplified flow chart of PRMA simulation model.

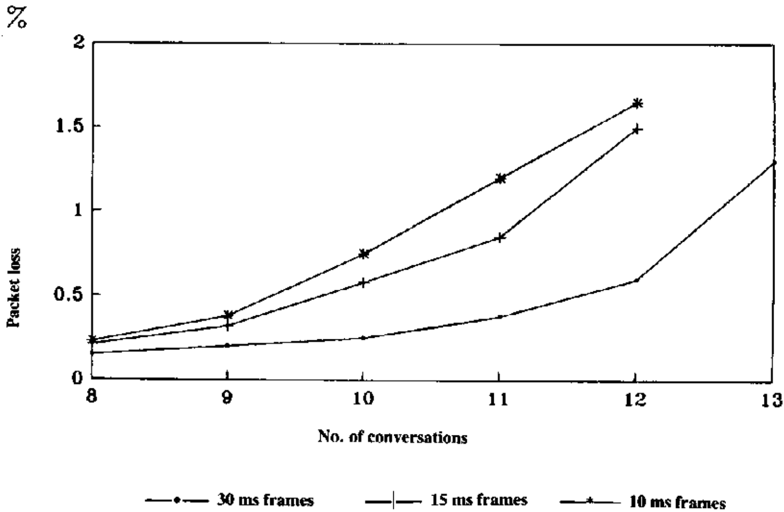

Fig. 3. Packet loss versus number of conversations for a bit rate of 320 KB/Sec.

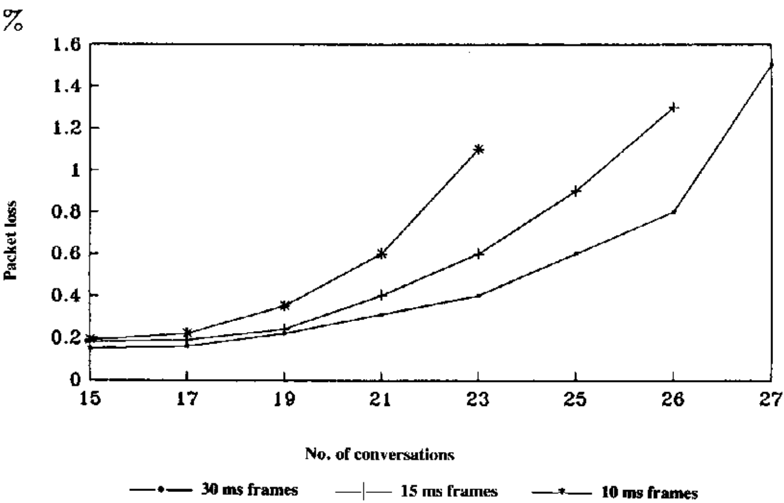

Fig. 4. Packet loss versus number of conversations for a bit rate of 640 KB/Sec.

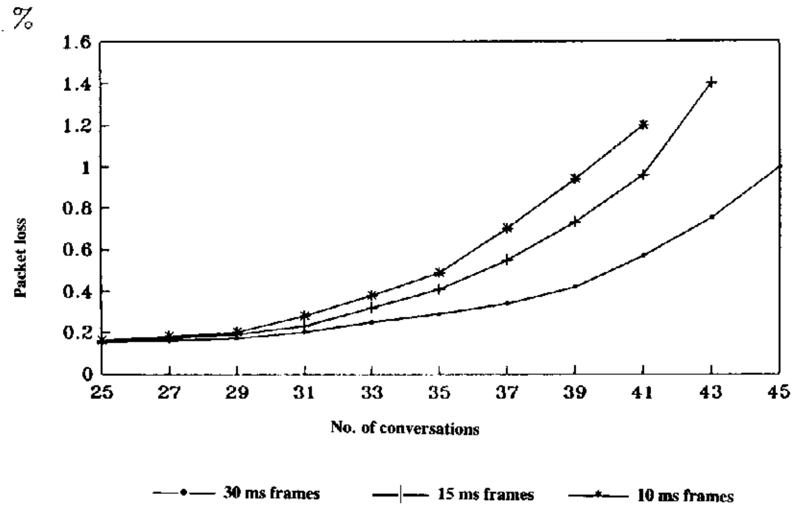

Fig. 5. Packet loss versus number of conversations for a bit rate of 1 MB/Sec.

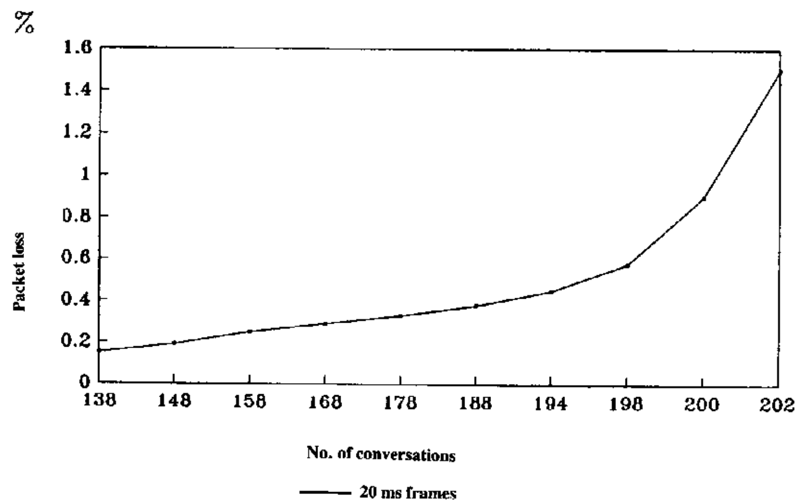

Fig. 6. Packet loss versus number of conversations for a bit rate of 5 MB/Sec.

sations per cell. For example, a 320 Kb/sec channel has a capacity of 11 conversations at a frame duration of 15 ms. A larger frame size can increase the capacity but at the expense of a longer packetization time. If a frame of 30 ms is selected, the 320 Kb/sec channel can accommodate 13 conversations. For increased traffic conditions, channels of higher capacities can be selected at appropriate frame sizes to accommodate larger number of simultaneous conversations. It is to be noted that only one multiplexed channel per cell is used at a time with the appropriate frame size.

In Table 1 the maximum capacity of each multiplexed channel, in terms of the number of simultaneous conversations the channel can hold, is shown with the packet loss kept below 1%. For example a channel of 1 Mb/sec can hold up to 45 simultaneous conversations for a frame length of 30 ms.

Table 1. Capacity of PRMA protocol in No. of conversations for different but rates of the multiplexed channel and frame sizes.

Frame size (ms)	Slots/frame	BW KB/sec	Capacity in no. of conv.
15	8	320	11
30	9	320	13
10	17	640	26
15	18	640	28
30	19	640	30
10	25	1000	39
15	26	1000	41
30	28	1000	45
20	138	5000	201

To compare with the traditional analog transmission system, let us consider a 1 Mb/sec digital channel. If the PRMA protocol is used, the channel can hold 45 simultaneous conversations for a frame size of 30 ms and having 28 slots per frame. In digital modulation a 1Mb/sec data rate can be supported by a frequency band of about 500 KHz if ASK or FSK modulation techniques are used [13]. This means an 11 KHz band is needed on average for each voice channel, while in traditional analog systems each voice channel occupies a 25 KHz band. Thus a two fold improvement of BW is possible in each cell. By reusing the same frequency at other cells a further improvement occurs. A cell supporting 45 simultaneous calls means carrying a traffic of 36.5 Erlangs at a blocking probability of 2%. This gives a calling rate Q = 1368 calls/hour/cell at a call holding time of 2 minutes [6].

If we use the 7 cell reuse pattern then $Q_T = 1368*7 = 9576$ calls/hour/7-cell pattern. If repeated 18 times (as in the earlier example of the city of Riyadh) then the total calling rate is Q = 172368. For 30% unutilized cells Q becomes 120,000, which uses a frequency band of $7 \times 0.5 = 3.5$ MHz. If we use a 10 MHz BW, the capacity becomes $Q = (10/3.5)*120,000 = 342,000$ calls/hour. Previously we saw in section 3 that the frequency reuse pattern, without using the PRMA protocol, gives an overall capacity of 146,000 calls/hour. So the capacity increases by a factor of 2.34 when the PRMA protocol is used. If necessary an even wider band can be used to support more traffic in each cell. For example, a 5Mb/sec multiplexed channel can hold up to 200 conversations; supported by providing 138 slots/20 ms frame with a loss rate due to packet dropping kept below 1 %.

7. Conclusion

Cellular mobile radio telephony is a new and attractive area of current interest. This paper provides a general idea of the cellular system with particular emphasis on the PRMA protocol for digital speech transmission. Our study shows that PRMA is a suitable candidate for the transmission of speech, as it increases the channel capacity by allowing more than one terminal to share a channel. This is done by multiplexing speech terminals in a TDMA fashion while keeping the implementation easier. This protocol is also suitable for integrated services and further study is in progress for the integration of voice and data with some priority of access mechanism for voice traffic.

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دراسـة لأداء نظام متعـدد المداخل لحجز حزم في دوائر الاتصالات السيارة الخلوية الرقمية

محمد زين العابدين

قسم علوم الحاسب، كلية علوم الحاسب والمعلومات، جامعة الملك سعود،
ص.ب. ٥١١٧٨، الرياض ١١٥٤٣، المملكة العربية السعودية

ملخص البحث. إن طرق الاتصالات اللاسلكية السيارة الخلوية الرقمية تدخل دائرة القبول لدى قطاع كبير من الناس. وهذا البحث يدور حول النظر في أداء نظام متعدد المداخل لحجز حزم في دوائر الاتصالات السيارة الخلوية الرقمية (PRMA) حين استخدامه في نقل الحزم الصوتية من مصدر سيار للكلام (الصوت).

وفي هذا الصدد فإن حزم الكلام (الصوت) من المصادر النشطة تتسابق للحصول على وقت مخصص للانتقال في قناة الإرسال. وقد تم باستخدام الحاسوب محاكاة نظام الحجز هذا لدراسة أدائه ولتقدير سعة القنوات التي لها حيز ذبذبي مختلف ومزمنن إطار مختلف.