# A Packet Reservation Multiple Access Assisted Cordless Telecommunication Scheme

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***Abstract*— A packet reservation multiple access (PRMA) assisted microcellular cordless telecommunications (CT) system is investigated for office type environments in absence of cochannel interference. The objective speech quality is found as a function of channel signal-to-noise ratio (SNR) for transmissions over narrowband Rayleigh fading channels, parameterized with the number of PRMA users. A moderate complexity 16-ary CT scheme constituted by a 4-bit/symbol 32-kbps adaptive differential pulse code modulation (ADPCM) speech codec, Reed–Solomon forward error correction (FEC) codec, and diversity-assisted 16-level star quadrature amplitude modulation (16-StQAM) modem is proposed. The 264-kBd 20-slot PRMA scheme supports 36 users while maintaining virtually imperceptible speech degradation for channel SNR's in excess of about 25 dB, and for a mobile speed of 2 mi/h if the signal-to-interference ratio (SIR) is above 30 dB.**

## I. Introduction

With the ever-increasing traffic demands of high capacity personal communication networks (PCN's) it is of vital importance to exploit every means of improving spectral efficiency. The dense frequency reuse of microcells dramatically improves traffic density at the expense of increasing the base station infrastructure cost [1]. The typically high coherence bandwidth removes the need for channel equalization which reduces the portable station's (PS) weight, cost, and power consumption. Although further spectral efficiency gains can be attained by higher complexity low bit rate speech codecs, another solution is to provide sufficient signal-to-noise ratio (SNR) for multilevel modems by means of an appropriate increase in the power budget, if no significant cochannel interference is present. Then the speech codec's complexity and bit rate constraint can be relaxed at the expense of a slightly higher modem complexity, unless spectral efficiency is at an absolute premium regardless of the complexity ramifications [2].

Time division multiple access (TDMA) has been favored in most contemporary large cell digital mobile radio (DMR) systems, such as the Pan-European GSM system, the British DCS1800, the American IS-54, and the Japanese DMR schemes. The resulting higher TDMA transmission rate requires channel equalization in their typically larger cells, although the Japanese system dispenses with equalization, advocating second-order antenna diversity. However, in cordless communications environments 4-bit/symbol-diversity-assisted differentially encoded noncoherently detected 16-StQAM [3] can be employed, slashing the signaling rate by a factor of 4, allowing demodulation without equalization in these office microcells, if the cochannel interference is sufficiently low. A means of improving TDMA spectral efficiency without a significant increase in complexity is to exploit the bursty nature of human speech using a voice activity detector (VAD). The VAD flags speech segments as active or passive and those having an active flag are allowed to contend with other active speech users for a time slot in the TDMA frame. If no other user requires this specific time slot, the user reserves it for uncontended future use, as long as the VAD flag is active. If, however, there is a collision, neither contender gets a reservation and both keep contending either delayed by a random delay offset or using a less than unity permission probability to prevent consecutive collisions. In case a speech packet does not get a free TDMA slot within a tolerable delay window, it must be dropped, the probability of which is limited to about 1%. This procedure, originally proposed by Goodman, Nanda *et al.* [4], [5] is termed packet reservation multiple access (PRMA), and, assuming a packet dropping probability of 1%, PRMA can increase spectral efficiency significantly.

In what follows we highlight the architecture of our proposed prototype system, describe its system elements, and present simulation results characterizing the system's performance.

## II. Packet Reservation Multiple Access

PRMA is a statistical multiplexing method for conveying speech signals via TDMA systems. It is designed to efficiently organize the flow of information from geographically dispersed PS's to a central BS. In contrast to packet data transmission, where packet dropping is not aceptaable but higher delays are tolerable, in packet speech communications low delays, typically <30 ms, are required, but a packet dropping probability of about 1% is acceptable and hardly perceivable.

The operation of PRMA is based on the VAD [6] being able to reliably detect idle speech segments. Inactive users' TDMA time slots are allocated to other users, who become active. The users who are just becoming active have to contend for the available time slots with a certain permission probability, which is an important PRMA parameter. The reason for allowing previously colliding users to contend for the next available time slot only with a less than unity permission

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probability is to prevent them from consistently colliding in their further attempts to attain reservation. If more than one user is contending for a free slot, neither of them will be granted it. If, however, only one user requires the time slot, he can reserve it for future use until he becomes inactive. Under heavily loaded network conditions, when many users are contending for a reservation, a speech packet might have to contend for a number of consecutive slots. When the contention delay exceeds 32 ms, the speech packet of 20-ms duration must be dropped. The probability of packet dropping must be kept below 1%, a value inflicting minimal degradation in perceivable speech quality.

In practice collision between a PS at short range and another far from the BS might allow successful decoding of the packet having a higher signal level, particularly if forward error correction (FEC) coding is used, a phenomenon referred to as packet capture by Goodman [4], although this would not happen with accurate power control. System parameters are the speech source coding rate $R_s$, the channel rate $R_c$, and the number of header bits $H$ associated with each packet. The PRMA frame duration is $T$, and since each user's speech codec generates one packet per PRMA frame, this must be transmitted at the channel rate $R_c$ during a slot interval of $\tau = T/N$, where $N$ is the number of time slots per PRMA frame, given by $N = [R_cT/(RsT + H)]$, where $[\cdot]$ is the integer value of $\cdot$.

The operation of PRMA algorithm implies a few assumptions as regards to the control infrastructure of the system. The BS is supposed to continually broadcast the status of each time slot within the previous PRMA frame, indicating whether it is available or whom it is reserved for. The assumption that this broadcast message reaches the PS's almost instantaneously applies to the small office microcells with their very low propagation delays. If the BS receives no speech packet in a previously reserved slot, it assumes that the slot is now available for contention, and this message is fed back to the PS's via the broadcast channel.

## III. Speech Model and Voice Activity Detector

The efficiency of PRMA is fundamentally dependent on the speech talk and silence statistics, both of which have been shown to possess negative exponentially distributed durations [7]. This model has been used by Goodman *et al.* [4], [5] in their investigations. In our simulations one of the speech subscribers was using prerecorded real speech transmitted over a simulated communications link, while the others were simulated using the negative exponential packet generation model and their packets were not transmitted. Our simulated communications link consisted of a VAD, speech codec, FEC codec, and modem with transmissions over Rayleigh fading channels. Note that in many CT environments Rician channels apply, of which the Rayleigh is the worst case channel.

The VAD's operation can be characterized by a simple two-state Markov model, where the probability that a talk spurt with mean length of $t$ finishes within a time-slot duration of $\tau$ is given by $P_1 = 1 - \exp(-\tau/t_1)$. Similarly, the probability of a transition from the silence state to the talk state, i.e., that a silent interval of $t_2$ mean duration elapses is $P_2 = 1 - \exp(-\tau/t_2)$. Experimentally verified values are $t_1 = 1\text{s}$ and $t_2 = 1.35\text{s}$ [7].

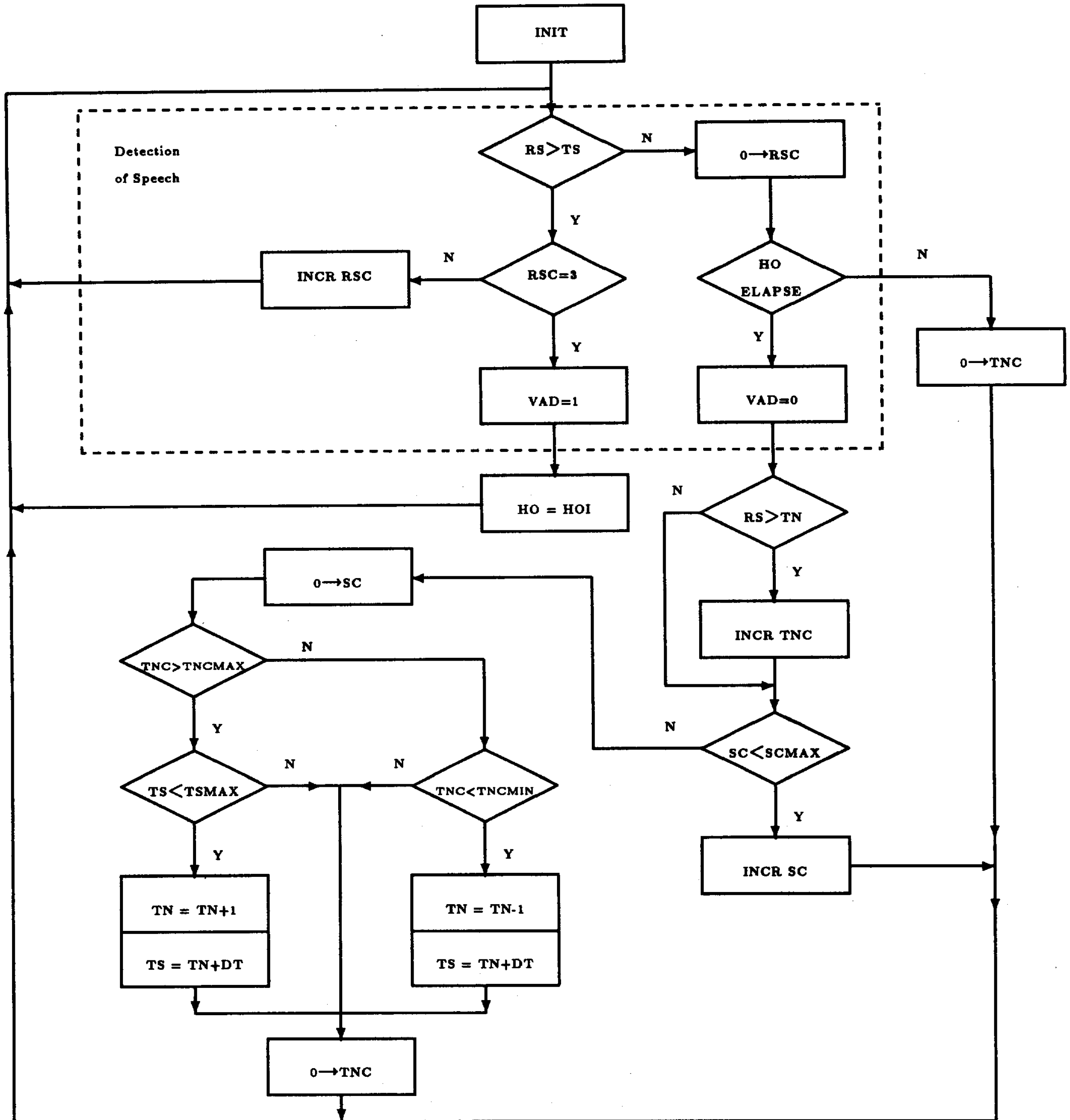
Any VAD must be designed to find a compromize among the following inherently contradictory requirements [8]: 1) minimizing false triggering on high-level noise; 2) transmission of low-level speech; 3) fast speech recognition to minimize initial talk spurt clipping or delay; 4) short switch-off delay (hangover or HO) to minimize unwanted activity, while preventing clipping of the low-energy end of talk spurt.

VAD's suggested in the literature exploit the differences between speech and noise properties in either the time-domain [6], [10] or in the frequency domain [11]. Our proposed VAD capitalizes on the different autocorrelation properties of speech and noise by exploiting that adjacent speech samples are highly correlated, as opposed to noise. Hence for speech several neighboring samples are more likely to be above an adaptively adjusted threshold than in case of uncorrelated noise. Assuming that the optimum speech detection threshold TS is known, the VAD decides that speech is at its input, if three consecutive input samples exceed TS. If, however this condition is not met, noise is assumed to be present. Once speech was detected, even if there is a short silent gap, the VAD flag is not set to zero, unless the so-called hangover (HO) period is elapsed. If during the HO period speech is detected at any time, the HO delay is reset to its initial value in order to prevent premature clipping of low-level speech spurt tails.

In order to render the VAD more sensitive to speech than noise we exploit that for equal rms valued speech and noise the speech has a long-tailed distribution, which reaches high amplitude values with higher probability than noise. From the complementary cumulative density function (CCDF) of speech and noise [6] we can infer, for example, that the (rms + 10) dB value is reached by speech 10 times more often than by noise. However, a simple threshold detector comparing samples with the above threshold value TS could still easily be deceived by high-level noise.

To circumvent this problem we exploit that, in contrast to noise, the long-term correlation function of speech suggests that speech is correlated over 300–400 $\mu$s, i.e., for 3–4 adjacent samples, if the speech sampling frequency is 8 kHz. On this basis speech is deemed to be present if three consecutive input samples are above the detection threshold. This condition is met by noise even in "covering" high-level noise very rarely. In fact this VAD requires some 10 dB higher noise power than the necessary speech power to activate it [6].

The VAD performance is further improved if adaptive threshold adjustment is used. The threshold optimization is carried out using a noise threshold (TN) and a speech threshold (TS). If noise is deemed to be present, the noise threshold (TN) is adjusted so that only a small percentage of noise samples, the noise peaks, shall be above the noise threshold TN. The speech threshold TS is then chosen to be above TN by a protection distance (DT), where DT = $10 \cdots 20$ dB, depending on the typical SNR. To compute the thresholds TN and TS the noise statistic is evaluated over an interval of approximately 1200 125-$\mu$s-spaced samples, which gives a reliable noise level assessment. Then the threshold TS is set so that more than 3%




INIT: Initialise

TS: Speech threshold
RS: Received signal sample
HO: Hangover
TN: Noise threshold
SC: Sample counter
TNCMAX: Maximum required value of TNC (60)
DT: Threshold step-size

TSMAX: Maximum value of TS
RSC: Received signal counter
HOI: Initial value of the HO delay
TNC: Noise threshold counter
SCMAX: Sample counter maximum
TNCMIN: Minimum required value of TNC (40)
INCR: Increment

Fig. 1. VAD flowchart.

but less than 5% of the noise samples, i.e., only the noise peaks, can exceed it.

The operation of the time-domain VAD proposed is summarized in the flowchart of Fig. 1, where the boxed section "Detection of Speech" is concerned with setting or resetting the VAD flag, according to whether speech is deemed to be present. After initialization (INIT), the current received signal sample (RS) is compared with the current speech threshold TS, and if RS is higher than TS, the received signal counter RSC is checked to see whether it has reached its maximum value of 3. If not, the counter RSC is incremented and the next RS sample is investigated. If RSC = 3, i.e., three

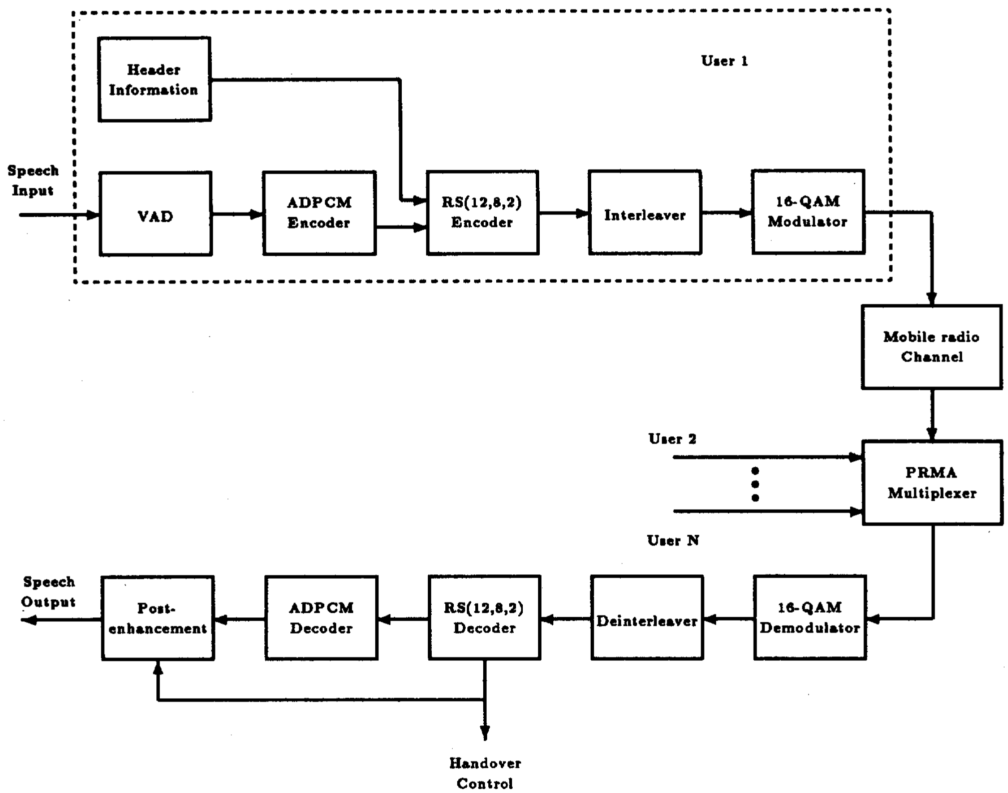



Fig. 2. System architecture.

consecutive samples exceeded the threshold TS, speech is deemed to be present, and the VAD flag is set to one. Also, VAD = 1 is maintained, if the RS samples are temporarily below the speech detection threshold TS, but the hangover HO has not elapsed. Again, the hangover has been introduced to prevent talk spurt clipping when low-power speech samples fall below TS. The HO delay was conveniently chosen to coincide with the speech frame length of 20 ms, i.e., 160 samples. Observe that if in the meantime three consecutive RS samples exceed TS, the hangover is reset to its initial value of HOI.

The rest of the flowchart outside the dashed line box is concerned with the adaptive threshold adjustment and it is activated only when noise is deemed to be present. In this state the noise statistics are computed for 1200 samples, i.e., 150 ms, and the noise threshold is adjusted so that it is exceeded only by the highest noise samples constituted by more than 3% but less than 5% of the 1200 samples. If the RS samples are above the noise threshold TN but below the speech threshold TS, noise is assumed to be present and hence the noise threshold counter TNC is incremented. The sample counter SC is always incremented in the threshold adjustment mode, unless it reaches its maximum of SCMAX = 1200 when it is reset to zero and a new noise averaging process is initiated. At this event the content of the noise threshold counter TNC is checked against its desired limits of TNCMAX = 60 ($0.05 \times 1200$) and TNCMIN = 40 ($0.03 \times 1200$). If $40 < \text{TNC} < 60$, i.e., the noise threshold TN is appropriately set to detect the noise peaks, then TNC is reset to zero, and the statistics computation resumes without updating the value of TN. Otherwise the noise threshold TN is appropriately increased or decreased by the step size of 1, and TS is updated according to TS = TN + DT. This process is repeated obeying the speech Markov model, where it is important to note that the computational complexity of the VAD is extremely low in the active speech transmission state and the threshold adjustment is carried out while no speech activity is detected.

## IV. System Architecture

The schematic diagram of our cordless telecommunication (CT) system is depicted in Fig. 2.

### A. Speech Codec

In the spirit of our introductory arguments for low complexity here we favor the 32-kbps CCITT G.721 ADPCM speech codec generating 4-bit symbols. Then, exploiting the relatively friendly propagation environment, we deploy a 4-bit/symbol 16-StQAM modem [3], [12], which compensates for the relatively high bit rate of the ADPCM codec and yields a Baud rate of 8 kBd.

### B. The 16-StQAM Modem

Although the maximum distance square 16-QAM constellation [12] is optimum in terms of noise protection distances for transmissions over Gaussian channels, its performance is degraded by carrier recovery false locking problems introduced by the fading environments that usually apply in CT's. Consequently, the differentially Gray-coded noncoherent 16-StQAM scheme proposed in [3], [14] is more appropriate for CT/PRMA. While 16-StQAM has a lower average distance among the constellation points than the maximum-minimum distance square 16-QAM, it is rotationally symmetric and hence neatly lends itself to differential coding. Further, differential coding can be advantageously combined with robust, low-complexity noncoherent detection, essentially facilitating tracking of the fading signal envelope and phase trajectory.

In our 16-StQAM modem the first bit of every 4-bit symbol is differentially coded onto two concentric rings of constellation points with radii of 1 and 3, respectively, yielding no ring-changing transition for logical 0, while requiring ring-swap for logical 1. The remaining 3 bits are differentially Gray-coded onto eight 45-degree-spaced phase positions, with 000 implying no phase angle change. Following the above-mentioned bit mapping oonto the phasor constellation the probability of any of the QAM bits being in error is nearly identical. The signal constellation points can be rearranged in any arbitrary manner to move some points closer to any of their bit-decision boundaries, thereby rendering some of the 4-bits more prone to errors, while increasing the integrity of others. This can be advantageously exploited, for example, in speech transmission schemes to provide bit sensitivity matched unequal error protection. However, this is only possible at the cost of increasing the overall BER for transmissions over fading channels.

Demodulation can be carried out by checking the magnitude ratio of two consecutive 16-StQAM phasors. If this ratio is lower than 2, logical zero is inferred, otherwise, logical 1. The remaining three bits are differentially Gray decoded from the phase rotation experienced.

### C. Error Correction Codec

To harmonize with this 4-bit/symbol system, we selected a low-complexity maximum minimum distance Reed–Solomon (RS) code, namely, the RS(12, 8, 2) code over the Galois field GF(16). This code encodes eight 16-ary GF(16) symbols into 12 GF(16) symbols and can correct two arbitrary symbol errors. Because RS codes have a maximum-minimum distance among codewords in the coding space, they have a high error-correcting power over channels with both bursty and random error statistics. Both soft-decision directed trellis decoding and hard-decision decoding are possible; however, the less complex hard-decision decoding yields similar performance. Therefore we favor hard-decision Berlekamp–Massey decoding. A further advantage of the RS codes is that their error detection capability can be exploited to control handovers [15] or speech postenhancement [12].

The operation of the RS codec is supported by a rectangular symbol interleaver, which has a memory of 22 RS codewords. In the case of nonbinary FEC codecs, symbol interleavers must be utilized, because bit interleavers disperse the bit errors into originally error-free RS coded symbols and thereby increase the symbol error rate. This is detrimental as regards the RS codec's performance, which operates on a symbol-by-symbol basis. The 22 RS codeword interleaving depth covers exactly one packet and it was chosen to ensure that no extra interleaving delay is introduced. At high PS speeds this interleaver provides sufficient randomising effect, but at walking speed it fails to improve the FEC performance. At a walking speed of 2 mi/h $\approx$0.9 m/s interleaving over an interval of 20 ms corresponds to about 1.8 cm, which is short in comparison to the wave length and hence becomes inefficient in dispersing the burst errors. However, at low fading rates, i.e., at low speed our differentially coded 16-StQAM modem's ability to track the fading envelope and phase trajectory improves, mitigating the effects of idling in deep fades. Furthermore, in the low-speed scenario the performance can be improved by deploying second-order diversity or frequency hopping, ensuring that every packet is transmitted at a different carrier frequency, which curtails the duration of long fades for pedestrian PS's.

### D. Speech Postenhancement

The principle of speech postenhancement is that whenever more than six consecutive RS-decoded codewords spanning approximately 6 ms speech are erroneous due to bad channel conditions, then the subjective speech quality can be improved if these erroneous speech segments are replaced by highly correlated error-free adjacent speech segments. The simplest solution is to substitute the previous segments if it was error free, but better results can be achieved by using simple correlative methods combined with time-domain raised-cosine smoothing at the segment edges [13], [12]. Should, however, less than six RS codewords be corrupted, the subjective effects are less objectionable, when no postprocessing is involved, because the channel errors are confined to a short (<6 ms) speech segment.

### E. Signaling Rate

When using the 16-ary transmission system shown in Fig. 2, the overall signaling rate is computed as follows. The 32-kbps ADPCM encoder generates 640 bits per 20-ms PRMA frame. These bits are transmitted in a 1-ms duration time slot, with the 64 header bits concatenated. The resulting 704 bits yield 176 4-bit symbols that are RS(12, 8, 2) coded to 264 GF(16) Galois field symbols, which in turn constitute 22 RS(12, 8, 2) codewords that represent the 20-ms speech segment. An interleaving depth of 22 RS codewords is used to "overbridge" deep channel fades and to disperse bursty errors upon deinterleaving at the receiver in case of higher PS speeds. Transmitting 264 4-bit symbols per 1 ms slot is equivalent to a signaling rate of 264 kBd. When using root raised cosine Nyquist transmitter and receiver filtering with a rolloff factor of 2/3, the bandwidth requirement of our 20-slot 16-StQAM system becomes 440 kHz, implying a modem bandwidth efficiency of 2.4 b/s/Hz.

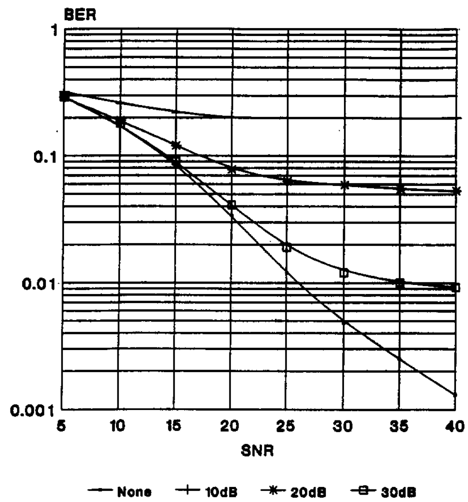



Fig. 3. 16-StQAM BER and SIR performance versus channel SNR.

## V. Simulation Studies

All the system elements of Fig. 2 were simulated and tested separately. We used a Rayleigh-fading channel having a propagation frequency of 1.9 GHz, that of the emerging PCN, a signaling rate of 264 kBd and PS speed of 2 mi/h.

### A. 16-StQAM Performance

Our 16-StQAM modem has a high bandwidth efficiency of 2.4 bit/s/Hz, but it requires relatively high SNR and signal-to-interference (SIR) values. This reduces its overall spectral efficiency in cellular type systems, where high cochannel interference prevails, since larger cluster sizes must be used in order to keep cochannel interferers further apart. An explicit formula for the overall spectral efficiency was derived by Gejji [17], which is based on Lee's formula [18] and Shannon's channel capacity equation, under the assumption of inverse fourth power wave propagation law and six closest interfering cells.

In this paper we characterize our 16-StQAM modem in terms of its channel bit error rate (BER) versus channel SNR performance, parameterized by the SIR values of 10 dB, 20 dB, and 30 dB as well as $\infty$ corresponding to no cochannel interference, as seen in Fig. 3. In this experiment a single Rayleigh-faded interferer was used, which corresponds to the worst case scenario. Observe that with no cochannel interference there was no residual BER at the assumed low PS speed, since our differentially coded modem was able to effectively trace the Rayleigh fading envelope and phase. However, with SIR values of 10 dB and 20 dB the residual BER became excessively high for the RS code and the speech quality was impaired. An SIR of 30 dB was required for channel SNR's in excess of about 25 dB for error-free speech quality.

For these reasons it should be noted that the number of channels supported by higher level modulation schemes, such as 16-StQAM would be reduced in a cellular system with frequency reuse. A study concerned with the efficiency of a range of modulation schemes when both the number of modulation levels and the required cluster sizes were varied was given in [19] for cellular systems. This showed that the efficiency is dependent on the required BER, but in cellular systems multilevel schemes can become less efficient than binary or quarternary constellations. Therefore the relative gains of the 16-StQAM scheme proposed in our CT system must be appropriately moderated, when deployed in a cellular system with frequency reuse. Furthermore, adaptive timing advance control must be used in order to ensure that the PS has instantaneous slot status information from the BS, as proposed for the GSM system.

However, we concern ourselves with CT systems in office type environments, where mild interference can be maintained by ensuring that cochannel BS's are on every other floor, and spaced widely apart on the same floor. In this scenario the cochannel interference does not significantly effect the BER performance, since each BS operates like a telepoint node, providing overlapping coverage with its neighboring BS's, which use a different set of channels. In this situation multilevel modulation will result in more users being supported per BS than in case of binary modems, as we will show.

TABLE I
PRMA Parameters for Initial Scenarios

| PRMA Parameters | Scenario 1 | Scenario 2 |
|---|---|---|
| Channel rate ($R_c$) | 720 kbit/s | 224 kbit/s |
| Source rate ($R_s$) | 32 kbit/s | 8 kbit/s |
| Slot duration ($\tau$) | 1 ms | 1 ms |
| Number of slots | 20 | 20 |
| Header length | 64 bits | 64 bits |
| Maximum speech delay | 32 ms | 32 ms |
| Permission probability ($P_{\mathrm{perm}}$) | variable | variable |

### B. PRMA Studies

*1) Initial PRMA Experiments:* For our PRMA scheme we considered two initial scenarios identical to those analyzed in [4], [16], the parameters of which are listed in Table I. Upon stipulating a 1% packet dropping probability and varying the permission probability we evaluated the number of users supported and acquired very similar curves to those in [4], [16]. These experiments using 20 slots suggested that at $P_{\mathrm{drop}} = 1\%$, the permission probability of $P_{\mathrm{perm}} = 0.3$ allowed us to support the highest possible number of users in both scenarios, namely, 37. At higher permission probabilities previously colliding users have to contend too frequently with resulting secondary collisions. When too low a permission probability was used, PS's becoming active might have to wait too long before they were allowed to contend, thereby reducing the overall throughput. According to [5], for different numbers of slots $P_{\mathrm{perm}} = 0.3$ must be inverse proportionally scaled by the number of slots to maintain an approximately constant number of users contending for any slot. For example, when doubling the number of slots to 40, $P_{\mathrm{perm}} \simeq 0.15$ yielded the highest number of users supported, while for 10 slots $P_{\mathrm{perm}} \simeq 0.6$ was preferable.

TABLE II
OVERALL PRMA SYSTEM COMPARISON

| System | Modem | Speech Rate (kbit/s) | PRMA Source Rate (kBd) | TDMA User Bandwidth (kHz) | Number of TDMA Users/Carrier | Number of PRMA Users/Carrier | Number of PRMA Users/Slot | PRMA User Bandwidth (kHz) |
|---|---|---|---|---|---|---|---|---|
| A | GMSK | 32 | 52.8 | 40 | 11 | 20 | 1.82 | 22 |
| B | $\frac{\pi}{4}$-DQPSK | 32 | 26.4 | 33.8 | 13 | 24 | 1.85 | 18.3 |
| C | 16-StQAM | 32 | 13.2 | 22 | 20 | 38 | 1.9 | 11.6 |
| D | GMSK | 16 | 28.8 | 22 | 20 | 38 | 1.9 | 11.6 |
| E | GMSK | 8 | 16.8 | 12.6 | 35 | 67 | 1.91 | 6.6 |
| F | 16-StQAM | 16 | 7.2 | 12 | 36 | 69 | 1.92 | 6.4 |
| G | 16-StQAM | 8 | 4.2 | 7 | 62 | 122 | 1.86 | 3.6 |

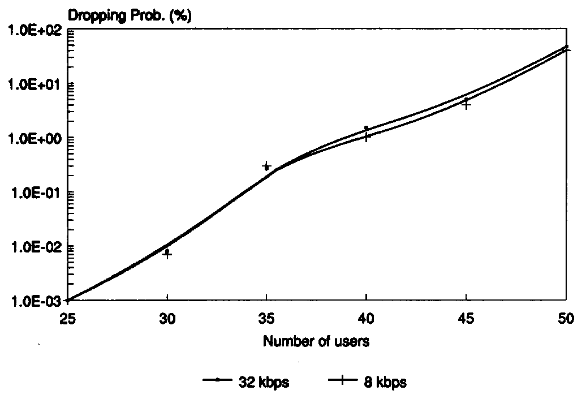



Fig. 4. PRMA packet dropping probability versus number of PRMA users for Scenario 1 and Scenario 2.

We then fixed $P_{\text{perm}} = 0.3$ for our 20-slot systems and evaluated the packet dropping probability $P_{\text{drop}}$ for both the 32-kbit/s and 8 kbit/s scenarios of Table I. The results are shown in Fig. 4, suggesting again that for $P_{\text{drop}} = 1\%$ about 37–38 users can be supported. Observe that these curves indicate identical packet dropping probabilities for the same number of users in both scenarios, since $P_{\text{drop}}$ depends essentially only on the number of users and time slots provided, which was 20 in both cases. The curve of the relative slot occupancy, defined as the proportion of actively used time slots versus number of PRMA users for these systems in Fig. 5 suggests that in the case of about 37–38 users, assuming $P_{\text{drop}} = 1\%$ the system operates with a relative slot occupancy of about 0.78. This value was further increased to a maximum of about 0.88, when supporting 50 users, but only at the cost of an unacceptable packet dropping probability of some 20%, as seen in Fig. 4. Beyond this peak the slot occupancy curve rapidly decays due to frequent collisions and "slot hogging."

*2) Nonbinary PRMA:* In our following experiment we embarked upon studying the effects of using 1, 2, and 4-bit/symbol binary and nonbinary signaling on PRMA efficiency, while fixing the source rate at 32 kbit/s and the system bandwidth at 440 kHz, as required by our proposed CT system. We refer to these sehemes as System A, B, and C, respectively, as seen in Table II. The bit error rate (BER) versus channel SNR performance and bandwidth efficiency ramifications of using 2-bit/symbol $\pi/4$-shifted differential quadrature phase shift keying ($\pi/4$-DQPSK) as in System B and those of a 4-bit/symbol 16-StQAM scheme as in System C were documented in [21] along with objective and subjective speech performance, robustness, and complexity issues. Here we only concentrate on the PRMA-related problems of these tradeoffs, emphasizing that these modem schemes accommodate different signaling rates or Baud rates within a fixed bandwidth. We will show that this fact has important ramifications as regards PRMA efficiency.

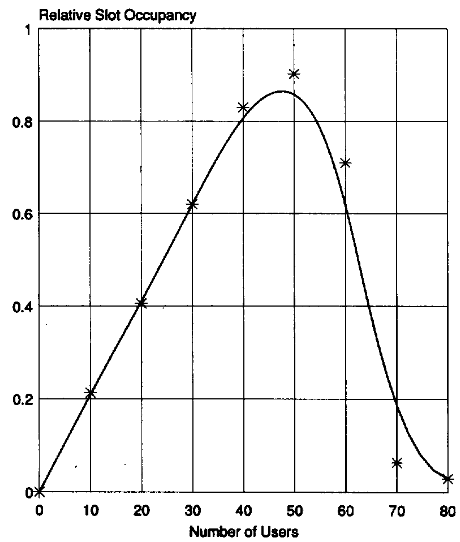



Fig. 5. Relative slot occupancy versus number of PRMA users of Scenario 1 and Scenario 2.

For example, using our bandwidth estimates from [21] for a 440-kHz channel slot, the 1-bit/symbol partial response Gaussian minimum shift keying (GMSK) scheme of System A with a typical bandwidth efficiency of 1.35 bit/s/Hz can accommodate a signaling rate of 594 kBd. As regards the 2-bit/symbol 264 kBd $\pi/4$-DQPSK modem of System B, where similarly to the American IS-54 system root raised cosine Nyquist filtering with a rolloff factor of 0.35 was used, the maximum signaling rate was approximately 440 kHz $\times$ 1.62 bit/s/Hz = 712.8 kbit/s = 356.4 kBd. Last, as stated before, the signaling rate of our 16-StQAM modem used in System C was 264 kBd.

Systems D–F are considered later in our deliberations; hence let us focus our attention on Systems A–C, where the speech source rates are 32 kbit/s, but the FEC coded source rates

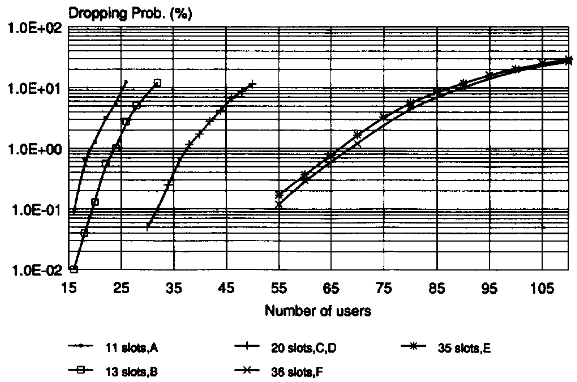



Fig. 6. PRMA packet dropping probability versus number of users for various systems.

including the packet header are different, as seen in Table II, column 4. For example, in case of binary transmissions at 594 kBd, as in System A, the 704 bits constituted by the 640 speech bits and 64 header bits yielded a source-rate of 35.2 kBd. After $R = 2/3$-rate channel coding, this amounted to a source rate of 52.8 kBd, allowing for 11 time slots to be created, as seen in columns 4 and 6 of Table II. With 2 bit/symbol $\pi/4$-DQPSK modem schemes, the 26.4 kBd signaling rate yielded 13 slots, while the 13.2-kBd 16-StQAM rate supported 20 time slots.

The number of users supported in these three scenarios at $P_{\text{drop}} = 1\%$ is seen from Fig. 6 to be about 20, 24, and 38, respectively. These figures are listed in Table II, column 7. The equivalent number of users per time slot supported was 20/11 = 1.82, 24/13 = 1.85, and 38/20 = 1.9, respectively, which are tabulated in Table II, column 8. When using multilevel transmissions, the Baud rate was reduced, allowing for more time slots to be created, which in turn decreased the packet dropping probability when PRMA was used.

The overall bandwidth efficiency figures of the 1, 2, and 4 bit/symbol schemes were computed as follows. The 11-slot binary System A supported 20 users at $P_{\text{drop}} = 1\%$ within 440 kHz, requiring 440 kHz/20 users = 22 kHz/users, as shown in the last column of Table II. For the 13-slot $\pi/4$-DQPSK System B the equivalent user bandwidth was 440 kHz/24 users $\simeq$18.3 kHz/user, while the 20-slot 16-StQAM System C required an average of 440 kHz/38 users $\simeq$ 11.6 kHz/user, which are listed in the last column of Table II. Observe that the QAM modem had the best PRMA efficiency in terms of number of users per time slot, followed by the $\pi/4$-DQPSK scheme, although the differences are not dramatic. However, the number of PRMA users per 440 kHz channel slot is nearly doubled for the 16-StQAM scheme in comparison with the GMSK modem, and therefore the required user bandwidth is nearly halved. As mentioned, all attributes of Systems A, B, and C are summarized for ease of comparison in Table II.

*3) Reduced Source-Rate PRMA:* Similarly to using multilevel modems, the number of time slots created can also be increased, when using half-rate or quarter-rate speech codecs. For example, when using a 16- or 8-kbps speech codec in a 594-kBd binary scheme, the following figures apply. A 16-kbps codec generates 320 bits per 20 ms PRMA frame, which after adding 64 header bits yields a bit rate of 384 bit/20 ms = 19.2 kbit/s. Again, assuming the same $R = 2/3$-rate error correction coding the source-rate becomes 28.8 kbit/s, creating 594/28.8 $\simeq$ 20 time slots for PRMA users, as shown for System D in Table II. An 8-kbit/s speech codec produces 160 + 64 = 224 bits per 20 ms, giving an FEC-coded PRMA source rate of 16.8 bit/s, which allows 35 slots to be supported, when using System D. These characteristics of Systems D and E are tabulated along with other system features in Table II, columns 2–5. The packet dropping probability versus number of users curves for Systems D and E were plotted for comparison with those of the nonbinary schemes in Fig. 6.

Observe that at $P_{\text{drop}} = 1\%$ both Systems C and D supported 38 users, but the 16-StQAM scheme required higher SNR and SIR values. In return for this, System C had a lower overall complexity, since it incorporated a lower complexity speech codec and did not require an equalizer. Similar arguments are valid for Systems E and F, which supported 67 and 69 users, respectively. It is also interesting to note that doubling the number of bits per modulation symbol, as in case of the $\pi/4$-DQPSK modem, was not sufficient to support the same number of users, as in the case of halving the speech source rate. This was due to the fact that the relative dominance of the packet header in the case of the shorter speech packets of low rate codecs was less detrimental than the comparatively low bandwidth efficiency gain achieved by the 2 bit/symbol $\pi/4$-DQPSK modem, which improved the bandwidth efficiency from 1.35 bit/s/Hz to 1.62 bit/s/Hz only. The number of users supported within a bandwidth of 440 kHz was nearly doubled in case of the 16-StQAM modem when compared to the binary GMSK scheme, if no cochannel interferers were present, when using identical speech codecs.

The equivalent number of users per time slot supported by the binary Systems D and E in case of the 16 and 8 bit/s speech codecs at $P_{\text{drop}} = 1\%$ is seen from Fig. 6 to be 38/20 = 1.9 and 67/35 = 1.91. In comparison, these figures are 1.92 and 1.97 in the case of the 4-bit/symbol 16-StQAM modem. These values are tabulated in the last but one column of Table II. The relative user bandwidth requirements for the binary GMSK Systems D and E are 440/38 = 11.6 kHz and 440/67 = 6.6 kHz, respectively, as demonstrated by the last column of Table II. The corresponding values for the 16-StQAM modem are 440/69 = 6.4 kHz and 440.122 = 3.6 kHz. The most users were accommodated by System G, supporting 122 users, each requiring a bandwidth of 3.6 kHz only.

All in all, the 4-bit/symbol 16-StQAM scheme of our choice was preferable in efficiently using the bandwidth allocated to each CT BS. Furthermore, better PRMA efficiency was achieved in a fixed bandwidth by halving the speech source rate than by doubling the number of bits per symbol. Indeed, the number of bits per symbol needed to be quadrupled in order to match the bandwidth efficiency gain of a half-rate speech codec. When bandwidth efficiency is at absolute premium, quarter-rate speech codecs further improve the PRMA- and overall bandwidth efficiency, as seen for Systems E and G in Table II. Lastly, when using higher slot numbers guaranteed by multilevel transmission schemes the packet dropping probability versus user number curves in Fig. 6 become less steep, implying a more graceful grade-of-service degradation characteristic.

### C. Objective Speech Quality Degradation with PRMA via Ideal Channels

With the PRMA system parameters settled, we then focused our attention on the objective degradation of ADPCM coded speech as a function of both the number of users ($N$) and packet dropping probability ($P_{\text{drop}}$). Different objective speech quality measures quantify different types of degradations. In order to ensure confident speech quality assessment, we used both the segmental signal-to-noise ratio (SEGSNR) as a waveform fidelity measure, as well as the cepstral distance (CD) [20] measure quantifying the speech spectral envelope distortion. The actual SEGSNR and CD values measured between the original and decoded speech are characteristic of the speech impairments due to ADPCM encoding as well as decoding and their value is typically speaker dependent. In our deliberations we were more interested in the PRMA- and channel-induced impairments and hence we used the speech SEGSNR degradation (SEGSNR-DEG) and the cepstral distance degradation (CD-DEG) due to PRMA packet dropping and channel errors, related to their values under perfect channel conditions when using no PRMA. A further advantage of using SEGSNR-DEG and CD-DEG instead of SEGSNR and CD was that the results became speaker independent.

The SEGSNR objective speech quality measure is defined as

$$\text{SEGSNR}^{\text{dB}} = \frac{1}{M}\sum_{m=1}^{M} 10\log_{10}\frac{\sum_{n=1}^{N} s_{\text{in}}^2(n)}{\sum_{n=1}^{N}[s_{\text{out}}(n)-s_{\text{in}}(n)]^2} \tag{1}$$

where $N = 160$ is the number of speech samples within a segment of 20 ms at a sampling rate of 8 kHz, while $M$ is the number of 20-ms segments, over which $\text{SEGSNR}^{\text{dB}}$ is evaluated. While the SEGSNR is a time-domain metric, the so-called cepstral distance measure represents the logarithmic spectral envelope distortion and is computed as follows

$$\text{CD} = \sqrt{[C_0^{\text{in}} - C_0^{\text{out}}]^2 + 2\sum_{j=1}^{3p}[C_j^{\text{in}} - C_j^{\text{out}}]^2} \tag{2}$$

where $C_j^{\text{in}}$ and $C_j^{\text{out}}, j = 0\cdots 3p$ are the cepstral coefficients of the input and output speech, respectively, and $p \approx$ 8–10 is the order of the linear predictive filter used in its evaluation.

Our findings for scenario 1 of Table I can be viewed in Figs. 7 and 8 in terms of the number of users supported and packet dropping probability, respectively. Both the SEGSNR-DEG and CD-DEG objective degradations became more steep for $P_{\text{drop}} > 1\%$ and $N > 38$, while below these values only minor degradations were observed, which is particularly true for $N \leq 35$. These objective SEGSNR and CD degradation figures became more meaningful in our later experiments for transmissions over fading channels, where the channel impairments caused greater degradations than those inflicted by the occasional packet dropping.

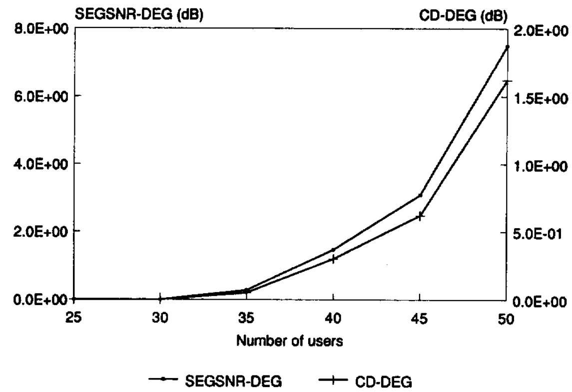



Fig. 7. Objective speech quality degradation with PRMA versus number of users.

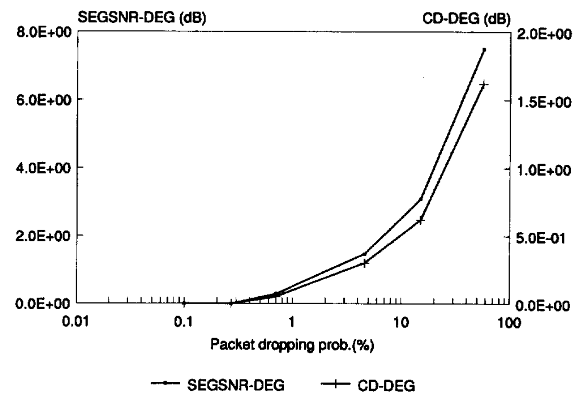



Fig. 8. Objective speech quality degradation with PRMA versus packet dropping probability.

### D. Transmission via Rayleigh Fading Channels

The problems of losing slot reservations over a fading PRMA uplink, when the BS failed to decode the packet header or when other PS's cannot error-freely decode the downlink slot-status information and hence collide with an active user were studied in [22]. In our experiments we assumed that the above problems did not occur. We note, however, that these problems can be mitigated by including a flag bit in the header, indicating for the BS in every uplink packet, whether there are more packets to come. The reservation would then be surrendered only if several consecutive packet headers were corrupted.

In our simulation studies we considered a Rayleigh channel, a propagation frequency of 1.9 GHz, a pedestrian speed of 2 mi/h and a signaling rate of 264 kBd. The bit error rate (BER) versus channel SNR performance of our 16-StQAM modem is depicted both with and without diversity in Fig. 9. Due to the randomizing effect of the channel interleaver, the most severe error bursts were broken up, providing a near-memoriless channel. The FEC codec removed all residual errors without diversity for channel SNR's above 35 dB, and with diversity for SNR's exceeding 25 dB. Diversity gave a 10 dB SNR advantage for our 16-StQAM modem.

In Figs. 10 and 11 we restricted the number of PRMA users to $N = 20$ in order to maintain a packet dropping

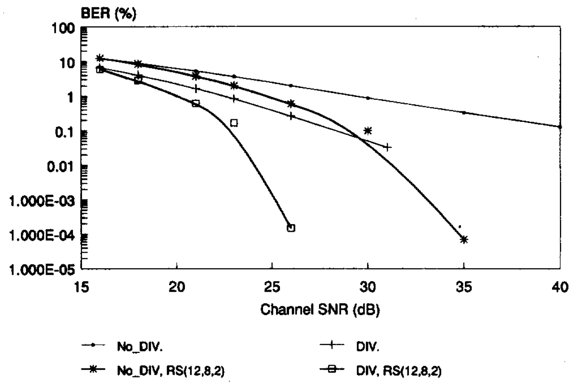



Fig. 9. 16-StQAM BER versus channel SNR via Rayleigh fading channels.

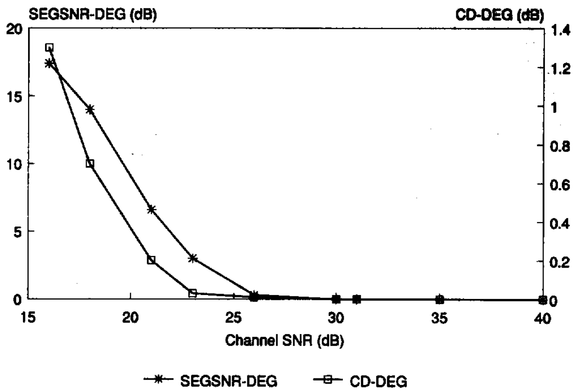



Fig. 10. Objective speech quality degradation versus channel SNR using diversity.

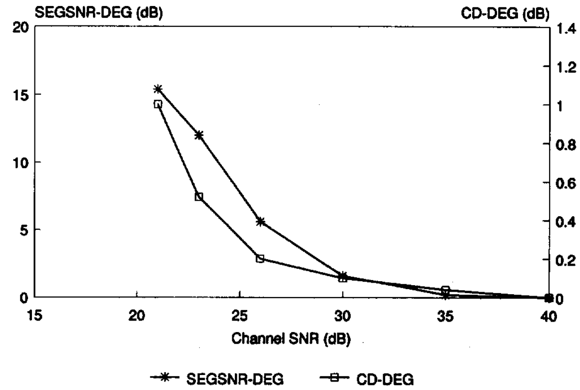



Fig. 11. Objective speech quality degradation versus channel SNR without diversity.

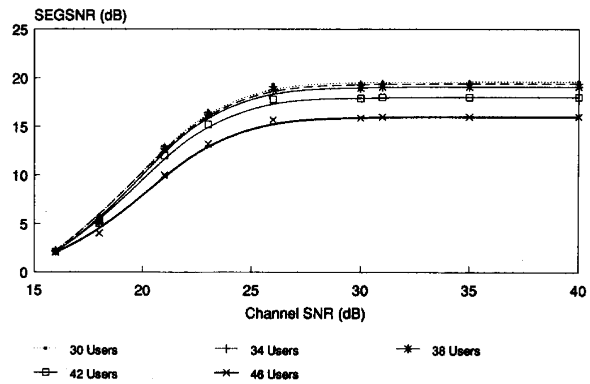



Fig. 12. Objective speech quality degradation versus channel SNR with different number of PRMA users.

probability of $P_{\text{drop}} = 0$, and investigated the SEGSNR-DEG and CD-DEG objective speech degradations as a function of the channel SNR with and without diversity. As expected from the 16-StQAM BER performances in Fig. 9, for SNR's in excess of 25 dB and 35 dB with and without diversity, respectively, virtually no objective speech degradations were inflicted.

Due to its superior performance and moderate complexity in our final scheme System C we favored the diversity assisted 16-StQAM modem. The overall combined system performance is characterized with the aid of Fig. 12, where the objective SEGSNR versus channel SNR curves are plotted parameterized with the number of users supported. Observe first of all that up to 38 users practically no objective SEGSNR degradation was inflicted. It is only for $N > 42$, where the speech degradation due to PRMA packet dropping became significant and comparable to channel impairments. The SEGSNR degradation (SEGSNR-DEG) due to supporting 42 users instead 38 users is roughly comparable in objective terms to that due to reducing the channel SNR from 25 dB to 23 dB.

## VI. Summary and Conclusion

In summary, using our ADPCM/RS(12, 8, 2)/16-StQAM/DIV/PRMA system operating in a CT office type environment supported 38 users via 20 TDMA time slots. We achieved virtually unimpaired speech quality over the microcellular CT mobile channels for a pedestrian speed of 2 mi/hr corresponding to a Doppler frequency of 5.6 Hz, when the channel SNR was in excess of 23–25 dB and the SIR was above 30 dB. The speech quality degradation introduced by PRMA packet dropping was more modest than that due to channel impairments. For higher Doppler frequencies, although these results are not included in this treatise, we found that the modem's performance was slightly degraded due to the differential codec's inability to accurately trace the fading trajectory. This was compensated by the RS codec interleaver complex's improved performance, and hence a similar overall system performance was maintained. An improvement by a factor of 1.8 in spectral efficiency measured in terms of bits per hertz was achieved by deploying PRMA.

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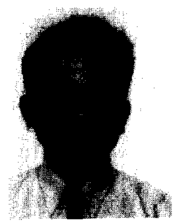

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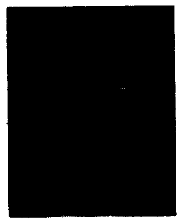

**Raymond Steele** (SM'80) received the B.Sc. degree in electrical engineering from Durham University, England, in 1959 and the Ph.D. and D.Sc. degrees from Loughborough University of Technology, England, in 1975 and 1983, respectively.

Before receiving the B.Sc. he was an indentured Apprentice Radio Engineer. After research and development posts with E. K. Cole, Cossor Radar and Electronics, and Marconi, he joined the lecturing staff at the Royal Naval College, London. He moved to Loughborough University in 1968 where he lectured and directed a research group in digital encoding of speech and picture signals. During the summers of 1975, 1977, and 1978 he was a consultant to the Acoustics Research Department, Bell Laboratories, and in 1979 he joined Bell's Communications Methods Research Department, Crawford Hill Laboratory, NJ. He returned to England in 1983 to become Professor of Communications in the Department of Electronics and Computer Science at the University of Southampton, a post he retains. From 1983 to 1986 he was a nonexecutive Director of Plessey Research and Technology and from 1983 to 1989 a Consultant to British Telecom Research Laboratories. In 1986 he formed Multiple Access Communications Ltd., a company concerned with digital mobile radio systems. He is the author of the book *Delta Modulation Systems* (Halstead, NY, 1975), editor of the book *Mobile Radio Communications* (Pentech Press, London, 1992) and the editor of a series of books on digital mobile communications. He and his coauthors were awarded the Marconi Premium in 1979 and in 1989, and the *Bell System Technical Journal's* Best Mathematics, Communications, Techniques, Computing and Software and Social Sciences Paper in 1981.

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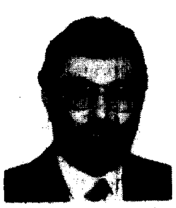

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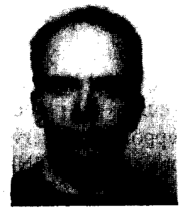

**William T. Webb** graduated from Southampton University with a first class honors degree in 1989. During his time at University he was sponsored by Ferranti Computer Systems Ltd. While at Southampton he was awarded the top year prize for each of his three years study. He received the Ph.D. degree in 1992, also from Southampton University. He is currently studying part-time towards the MBA degree.

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