Performance of PRMA Schemes via Fading Channels

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Abstract

A packet reservation multiple access (PRMA) assisted microcellular cordless telecommunications (CT) system is investigated for office type environments. The objective speech quality is found as a function of channel signal to noise ratio (SNR) for transmissions over narrowband Rayleigh fading channels, parametrized 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 264kBd 20-slot PRMA scheme supports 36 users while maintaining virtually imperceptible speech degradation for channel SNRs in excess of about 25 dB and for mobile speeds above 2 mph.

1 Introduction

With the ever increasing traffic demands of high capacity personal communication networks (PCNs) 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. The typically high coherence bandwidth removes the need for channel equalization which reduces the portable stations'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 [1] by means of an appropriate increase in the power budget. This further reduces the signalling rate and hence the detrimental effects of dispersion. 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].

In most contemporary digital mobile radio (DMR) systems time division multiple access (TDMA) has been favoured. 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) and the principle of packet reservation multiple access (PRMA), which was originally proposed by D. J. Goodman [3, 4].

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.

2 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 PSs to a central BS. In contrast to packet data transmission, where packet dropping is not acceptable but higher delays are tolerable, in packet speech communications low delays, typically $< 30\,\mathrm{ms}$, are required, but a packet dropping probability of about 1% is acceptable and hardly perceivable.

The operation of PRMA is based on the VAD [5] 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 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, in particular, if forward error correction (FEC) coding is used; a phenomenon referred to as packet capture by Goodman [3], 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 the 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 PSs almost instantaneously applies in 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 PSs via the broadcast channel.

3 Speech Model and Voice Activity Detector

The efficiency of PRMA is fundamentally dependent on the talk and silence statistics of human speech, both of which have been shown to possess negative exponentially distributed durations [6]. This model has been used by Goodman et. al [3, 4] in their investigations. In our simulations one of the speech subscribers was using pre-recorded real speech, while the others the negative exponential packet generation model. This allowed us to deploy a real communication link consisting 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 described by a simple two-state Markov model, where the probability that a talk spurt with mean length of t_1 finishes within a time-slot duration of τ 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 $t_1 = 1\mathrm{s}$ and $t_2 = 1.35\mathrm{s}$ [6].

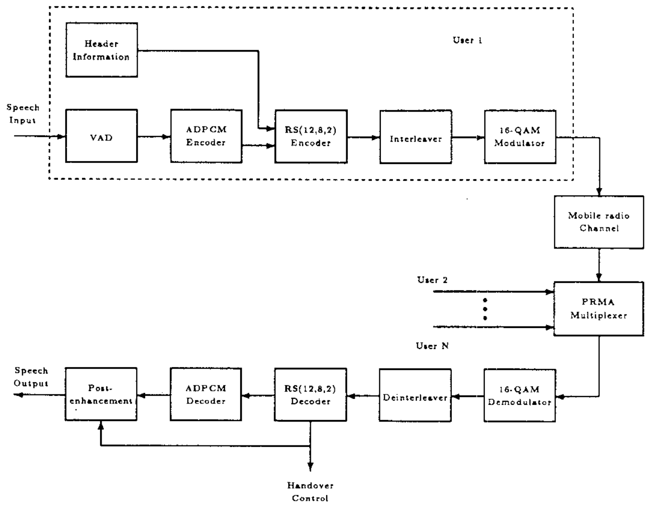

Figure 1: System architecture

Any VAD must be designed to find a compromise amongst the following inherently contradictory requirements [7]: (1) minimising false triggering on high-level noise; (2) transmission of low-level speech; (3) fast speech recognition to minimise initial talk spurt clipping or delay; (4) short switch off delay (hangover or HO) to minimise unwanted activity, while preventing clipping of the low-energy end of talk spurt. Our low-complexity adaptive time-domain VAD implementation is based on reference [5] and fulfils the above requirements.

4 System Architecture

The schematic diagram of our cordless telecommunication (CT) system is depicted in Figure 1. In the spirit of our introductory arguments for low complexity here we favor the 32kbps CCITT G.721 ADPCM speech codec generating four-bit symbols. Then, exploiting the friendly propagation environment, we deploy a four bit/symbol 16-StQAM modem [1, 8], which compensates for the relatively high bit rate of the ADPCM codec and yields a Baud rate of 8 kBd.

Although the maximum-distance square 16-QAM constellation [8] 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 CTs. Consequently, the differentially Gray-coded non-coherent 16-StQAM scheme proposed in reference [1] is more appropriate for CT/PRMA. While 16-StQAM has a lower average distance amongst the constellation points, it is rotationally symmetric and hence neatly lends itself to differential coding. Further, differential coding can be advantageously combined with robust, low-complexity non-coherent 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 three 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 onto the phasor constellation the probability of any of the four 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 four 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 two, logical zero is inferred, otherwise logical one. The remaining three bits are differentially Gray-decoded from the phase rotation experienced.

In order 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 amongst 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 favour hard-decision Berlekamp-Massey decoding. A further advantage of the RS codes is that their error detection capability can be exploited to control handovers or speech post-enhancement [8].

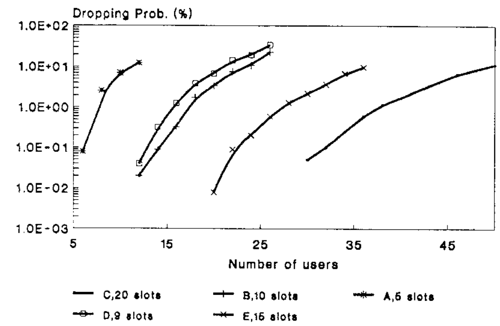

Figure 2: PRMA packet dropping probability vs. number of users for various systems

PRMA parameter	
Channel rate	264 kBd
Source rate	8 ksymbols/sec
Frame duration	20 ms
No. of slots	20
Slot duration	1 ms
speech frame length	20 ms
Header length	64 bits
Maximum speech delay	32 ms
Permission probability	0.3

Table 1: PRMA parameters

The principle of speech post-enhancement 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 [8], [9]. Should however less than six RS codewords be corrupted, the subjective effects are less objectionable, when no post-processing is involved, because the channel errors are confined to a short (< 6 ms) speech segment.

When using the 16-ary transmission system shown in Figure 1, the overall signalling rate is computed as follows. The 32kbps 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 de-interleaving at the receiver. Transmitting 264 4-bit symbols per 1 ms slot is equivalent to a signalling rate of 264kBd. The parameters of our PRMA scheme are summarised in Table 1.

5 Results and Discussion

5.1 Non-binary PRMA

All the system elements of Figure 1 were simulated and tested separately. Then in a comparative experiment we embarked upon studying the effects of using 1, 2 and 4 bit/symbol binary and non-binary signalling on PRMA

System	Modem	Speech Rate (kbps)	PRMA Source Rate (kBd)	TDMA User Bandw. (kHz)	No of TDMA Users/Carrier	No of PRMA Users/Carrier	No of PRMA Users/slot	PRMA User Bandw. (kHz)
A	GMSK	32	52.8	39	5	8	1.6	22
B	$\frac{\pi}{4}$-DQPSK	32	26.4	33	10	18	1.8	18
C	16-StQAM	32	13.2	22	20	38	1.9	11.6
D	GMSK	16	28.8	21.3	9	16	1.78	12.25
E	GMSK	8	16.8	12.4	15	27	1.8	7.26
F	16-StQAM	16	7.2	12	36	70	1.94	6.3
G	16-StQAM	8	4.2	7	62	125	2	3.52

Table 2: Overall PRMA system comparison

efficiency, while fixing the channel signalling rate or Baud-rate at 264 kBd, as proposed for our CT scheme. We refer to these schemes as System A, B and C, respectively, as seen in Table 2. 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 [11] along with objective and subjective speech performance, robustness and complexity issues. Here we only concentrate on the PRMA-related problems of these trade-offs, emphasizing that these modem schemes have different bandwidth requirements, when using identical channel Baud-rates.

For example, using our bandwidth estimates from reference [11] for a 264 kBd 1 bit/symbol Gaussian minimum shift keying (GMSK) scheme of System A with a typical bandwidth efficiency of 1.35 bit/Hz a bandwidth of about 196 kHz was required. As regards to the 2 bit/symbol 264 kBd $\pi/4$-DQPSK modem of System B, where root raised cosine Nyquist filtering with a roll-off factor of 0.35 was used, the bandwidth requirement was approximately 325 kHz. Lastly, for a Nyquist filtered 4 bit/symbol 16-StQAM scheme, such as that in System C with a roll-off factor of 1 a channel spacing of about 440 kHz was required. As a consequence, the number of PRMA users derived for these scenarios must be offset by the actual bandwidth efficiency figures of 1.36 bit/Hz, 1.6 bit/Hz and 2.4 bit/Hz for the 1, 2 and 4 bit/symbol modems, as we will show at a later stage.

It should be noted that the number of channels supported by the higher level schemes would be reduced in a cellular system with frequency re-use. This is because these higher level schemes require an increased signal to interference ratio (SIR) to operate at the same BER as the lower level schemes, necessitating an increased cluster size with corresponding reduction in the bandwidth assigned to each base station. 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 reference [10] for cellular systems.This showed that the efficiency is dependent on the required BER, but in cellular systems multi-level 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 re-use. 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 floors provide high attenuations as do walls. By ensuring that co-channel BSs are on every other floor, and spaced widely apart on the same floor, the co-channel interference does not significantly effect the BER performance. It is as if each BS operates like a telepoint node, providing overlapping coverage with its neighbouring BSs, which use a different set of channels. In this situation multilevel modulation will result in more users being supported per BS, as we will now show.

In case of binary transmissions at 264 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 only five time slots to be created, as seen in columns 4 and 6 of Table 2. With 2 bits/symbol $\pi/4$-DQPSK modem schemes the 26.4 kBd signalling rate yielded 10 slots, while the 13.2 kBd 16-StQAM rate supported 20 time slots.

The number of users supported in these three scenarios at $P_{drop} = 1\%$ is seen from Figure 2 to be about 8, 18 and 38, respectively, where the packet dropping versus number of users curves of systems A-E are portrayed. These figures are listed in column 7 of Table 2. The equivalent number of users per time slot supported was $8/5 = 1.6$, $18/10 = 1.8$ and $38/20 = 1.9$, respectively, which are tabulated in column 8 of Table 2. When using multi-level 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 5-slot binary System A supported 8 users at $P_{drop} = 1\%$ within a bandwidth of 196 kHz, requiring 196 kHz/8 users = 22 kHz/users, as shown in column 5 of Table 2. For the 10-slot $\pi/4$-DQPSK System B the user bandwidth was 325 kHz/18 users $\simeq$ 18 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 2. Observe that even when we off-set the PRMA bandwidth-efficiency figures with the different modem filtering requirements, the QAM modem had the best PRMA performance. As mentioned, all attributes of Systems A, B and C are summarised for ease of comparison in Table 2.

5.2 Reduced source-rate PRMA

Similarly to using multi-level modems, the number of time slots created can also be increased, when using half-rate or quarter-rate speech codecs. However, the gains obtainable are more modest due to the increasing dominance of the header in terms of channel capacity. For example, when using a 16 or 8 kbps speech codec in a 264 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 bits/20 ms = 19.2kbps. Again, assuming the same $R = 2/3$-rate error correction coding the source-rate becomes 28.8 kbps, creating $264/28.8 \simeq 9$ time slots for PRMA users, as shown for System D in Table 2. An 8 kbps speech codec produces $160+64 = 224$ bits per 20 ms, giving an FEC-coded PRMA source rate of 16.8 kbps, which allows 15 slots to be supported, when using System D. These characteristics of Systems D and E are tabulated along with other system features in columns 2-5 of Table 2. The packet dropping probability versus number of users curves for Systems D and E were plotted for comparison with those of the non-binary schemes in Figure 2.

Observe that at $P_{drop} = 1\%$ about two more users were supported, when using a 2 bit/symbol $\pi/4$-DQPSK modem along with a 32 kbps speech codec, as seen for System B, than in case of a 16 kbps codec and binary modem combination, used in System D, while also the lower complexity was associated with the latter scheme. This difference became even more pronounced in case of an 8 kbps speech codec and binary modem as in System E, compared to the 32 kbps speech coded and 16-StQAM modem combination (System C). As demonstrated by Figure 2, approximately 11 more users were supported by the latter scheme, again at a lower complexity. The equivalent number of users per time slot supported in case of the 16 and 8 kbps speech codecs at $P_{drop} = 1\%$ is seen from Figure 2 to be $16/9 = 1.78$ and $27/15 = 1.8$, respectively, compared to 1.8 and 1.9 in case of the 2 bit/symbol and 4 bit/symbol scenarios. These values are tabulated in the last but one column of Ta-

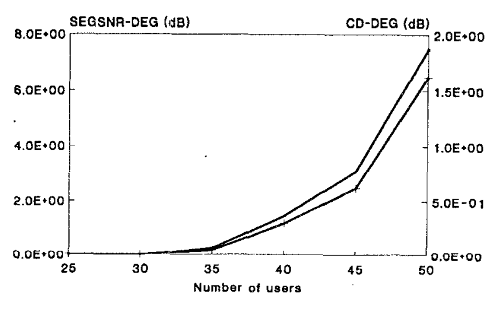

Figure 3: Objective speech quality degradation with PRMA vs. number of users

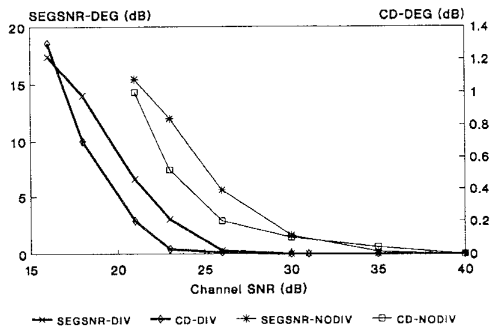

Figure 4: Objective speech quality degradation vs. channel SNR with and without diversity

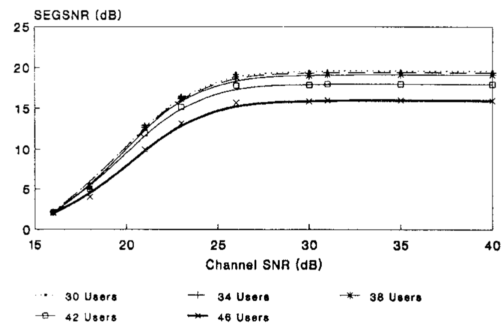

Figure 5: SEGSNR degradation vs. channel SNR with different number of PRMA users

ble 2. The relative user bandwidth requirements are $196/16 = 12.25$ kHz and $196/27 = 7.26$ kHz, respectively, as demonstrated by the last column of of Table 2.

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 by increasing the number of modulation levels than by more complex low-rate speech codecs. When bandwidth efficiency is at absolute premium, half- and quarter-rate speech codecs further improve the PRMA- and overall bandwidth efficiency, as seen for Systems F and G in Table 2.

Finally, when using higher slot numbers guaranteed by multi-level transmission schemes the packet dropping probability versus user number curves in Figure 2 become less steep, implying a more graceful grade-of-service degradation characteristic.

5.3 Speech degradation due to PRMA

With the PRMA system parameters settled, we then focussed 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_{drop}). Different objective speech quality measures quantify different types of degradations, and hence for confident assessment we used both the segmental signal-to-noise ratio degradation (SEGSNR-DEG) and the cepstral distance degradation (CD-DEG), as in reference [2]. Our findings for the proposed CT system are shown in Figure 3 in terms of SEGSNR-DEG and CD-DEG as a function of the number of users supported. Both the SEGSNR-DEG and CD-DEG objective degradations became more steep for $N > 38$, which corresponded to $P_{drop} > 1\%$, 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.

5.4 Speech degradation due to Rayleigh fading

We consider a Rayleigh channel, a propagation frequency of 1.9 GHz, a pedestrian speed of 2 mph and a signalling rate of 264kBd. In Figure 4 we restricted the number of PRMA users to $N = 20$ in order to maintain a packet dropping probability of $P_{drop} = 0$, and investigated the SEGSNR-DEG and CD-DEG objective speech degradations as a function of the channel SNR with and without diversity. Observe that for SNRs 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 favoured the diversity assisted 16-StQAM modem. The overall combined system performance is characterized with the aid of Figure 5, 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.

6 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 mph corresponding to a Doppler frequency of 5.6 Hz, when the channel SNR was in excess of 23–25 dB. The speech quality degradation introduced by PRMA packet dropping was more modest than that due to channel impairments. For higher Doppler frequencies the system performance improves, because the fading becomes more rapid, decreasing the length of the error bursts and thereby improving the performance of the RS codec and 16-StQAM modem. An improvement by a factor of 1.8 in spectral efficiency measured in terms of bits/Hz was achieved by deploying PRMA.

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