

signal channel gain vector to follow an arbitrary fading distribution with an arbitrary covariance structure, although multiple interferers are still required to be i.i.d. Rayleigh faded. A generic and simple expression for the outage probability has been obtained. Its application to various faded signals is elaborated upon and yields various results for Rayleigh, Rician, and Nakagami faded signals. Future effort will include removing the i.i.d. Rayleigh fading assumption on cochannel interferers to thoroughly solve the outage problem of maximum ratio combining.

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A Redundant Residue Number System Coded Burst-by-Burst Adaptive Joint-Detection Based CDMA Speech Transceiver

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Abstract—A burst-by-burst (BbB) adaptive speech transceiver is proposed, which can drop its source coding rate and speech quality under transceiver control in order to invoke a more error resilient modem mode among less favorable channel conditions. The adaptive multirate (AMR) speech codec is operated at bit rates of 4.75 and 10.2 kb/s and combined with source sensitivity-matched redundant residue number system (RRNS) based channel codes. BbB adaptive joint detection aided code division multiple access is used for supporting the dual rate speech codec. Both the objective and subjective speech quality assessments favored the proposed BbB adaptive transceiver.

Index Terms—AMR voice codec, HSDPA, RNS, RRNS.

I. BACKGROUND

In recent years, substantial research efforts have been invested in mitigating the effects of time-variant channel quality fluctuations encountered by wireless systems [1]. As a powerful but potentially complex countermeasure, multiple-transmitter and multiple-receiver aided space-time coding [2], space-time spreading [11], and beamforming arrangements [3] have emerged. However, their performance may be matched at a lower complexity upon invoking a whole variety of adaptive transceiver schemes [1]. Among other solutions, Ottosson and Svensson proposed various multi-rate systems [4], including multiple spreading factor (SF) based, multicode and multilevel modulation schemes. Procedures that exploit the time-varying nature of the mobile channel and having different grades of sophistication [1] are already in place for all the major cellular standards worldwide [5]. However, at the time of writing, no near-instantaneously adaptive real-time or interactive speech system exists, and this motivated the research reported in this contribution.

Recently, the adaptive multirate (AMR) speech codec has been standardised by ETSI [6], [7]. The codec is capable of operating in the full rate and half rate speech traffic channels of GSM. It is also amenable to adapting the source coding and channel coding bit rates according to the quality of the radio channel. Most speech codecs employed in communication systems—such as for example, the existing GSM speech codecs (full rate—half rate, and enhanced full rate)—operate at a fixed bit rate, with a trade off between source coding and channel coding. However, estimating the channel quality and adjusting the bit rate adaptively according to the channel conditions has the potential of improving the error resilience and the speech quality over wireless channels.

The AMR concept is amenable to a range of intelligent configurations. When the instantaneous channel quality is low, the speech encoder operates at low bit rates, thus facilitating the employment of more powerful forward error control within a fixed bit rate budget,

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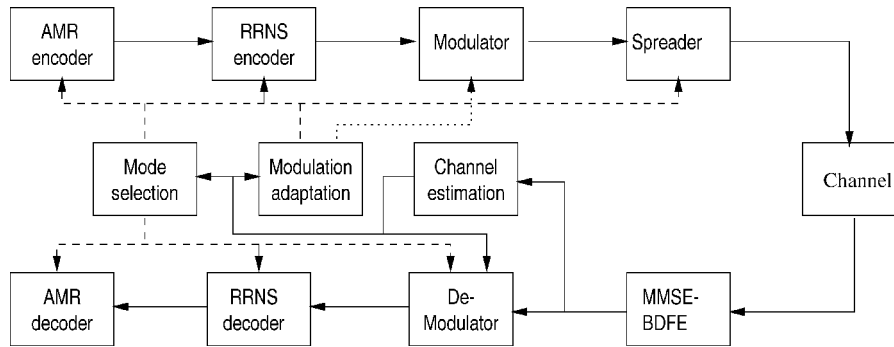


Fig. 1. Schematic of the adaptive dual-mode JD-CDMA system.

or using a more robust but lower-rate transceiver mode. By contrast, under favorable channel conditions, the speech encoder may use its highest bit rate, implying high speech quality, since in this case weaker error protection is sufficient or a less robust but higher bit rate transceiver mode can be invoked. However, the system must be designed for seamless switching between rates without encountering annoying perceptual artifacts. The evolution of speech compression research has been portrayed, for example, in [8], and hence we refrain from detailing the recent achievements in the field of speech coding.

Focusing our attention on the associated transmission aspects, in recent years, significant research interests have also been devoted to burst-by-burst adaptive quadrature amplitude modulation (BbB-AQAM) transceivers [9]–[18]. The transceiver reconfigures itself on a BbB basis, depending on the instantaneous perceived wireless channel quality [1]. More explicitly, the associated channel quality of the next transmission burst is estimated, and the specific modulation mode, which is expected to achieve the required BER performance target at the receiver, is then selected for the transmission of the current burst. Modulation schemes of different robustness and of different data throughput have also been investigated [9]–[15]. The BbB-AQAM principles have also been applied to joint detection code division multiple access (JD-CDMA) [1], [11], and OFDM [1], [10], [12]–[14]. A host of applications to wireless video telephony has been proposed and characterised in [9], while the achievable network performance and teletraffic capacity of such adaptive systems has been shown to be approximately doubled with the advent of adaptive beamforming and adaptive modulation in [3]. A range of other design issues of adaptive CDMA can be found, for example, in [19]–[22], while the synergy between the adaptive physical layer and multiple access control designed for integrated voice and data services in a cellular wireless network was addressed in [23].

Against this background characterized in terms of the recent literature [1]–[23], the main objective of this contribution is to propose a novel near-instantaneously adaptive real-time interactive speech system, which amalgamates cutting-edge system components while maximizing the perceptual speech quality. Specifically, the AMR speech codec, the maximum minimum-distance family of redundant residue number system (RRNS) assisted channel codes [2], [11], [13], [40], [41], and BbB-adaptive JD-CDMA are used [11] in conjunction with source sensitivity matched speech protection [8]. The near-instantaneously adaptive transceiver mode switching is controlled by the channel quality fluctuations imposed by the time-variant channel, which is not a desirable scenario. However, we will endeavour to contrive measures in order to mitigate the associated perceptual speech quality fluctuations. The underlying tradeoffs associated with employing two speech modes of the AMR standard speech codec in conjunction with a reconfigurable, unequal error protection BPSK/4QAM JD-CDMA modem are investigated.

The paper is structured as follows. Section II provides a brief system overview. Section III details the structure of the AMR speech codec, while the associated bit sensitivity issues are discussed in Section IV. Section V describes the RRNS channel codes applied in our system, and the associated source-matched error protection scheme is discussed in Section V-B. The BbB-AQAM JD-CDMA scheme is detailed in Section VI. Finally, before concluding, our system performance results are summarized in Section VII.

II. SYSTEM OVERVIEW

The schematic of the proposed adaptive JD-CDMA speech transceiver is depicted in Fig. 1. The encoded speech bits generated by the AMR codec at the bit rate of 4.75 or 10.2 kb/s are first mapped according to their error sensitivities into three protection classes, although for simplicity this is not shown explicitly in the figure. The sensitivity-ordered speech bits are then channel encoded using the RRNS encoder [2], and modulated using a re-configurable BPSK or 4QAM based JD-CDMA scheme [15]. We assigned the 4.75 kb/s speech codec mode to the BPSK modulation mode, and the 10.2 kb/s speech codec mode to the 4QAM mode. Therefore, this transmission scheme can provide higher speech quality at 10.2 kb/s, provided that sufficiently high channel SNRs and SIRs prevail. Furthermore, it can be reconfigured under transceiver control to provide an inherently lower but unimpaired speech quality among lower SNR and SIR conditions at the speech rate of 4.75 kb/s.

Subsequently, the modulated symbols are spread in Fig. 1 by the spreading sequence assigned to the user, where a random spreading sequence is employed. The minimum mean squared error block decision feedback equalizer (MMSE-BDFE) is used as the multiuser detector [11], where perfect channel impulse response (CIR) estimation and perfect decision feedback are assumed. The soft outputs for each user are obtained from the MMSE-BDFE and passed to the RRNS channel decoder. Finally, the decoded bits are mapped back to their original bit protection classes by using a bit-mapper (not shown in Fig. 1), and the speech decoder reconstructs the original speech information.

In BbB-AQAM/CDMA, in order to determine the best choice of modulation mode in terms of the required tradeoff between the BER and throughput, the near-instantaneous quality of the channel has to be estimated. The channel quality is estimated at the receiver, and the chosen modulation mode and its corresponding speech mode are then communicated using explicit signaling to the transmitter in a closed-loop scheme, as depicted in Fig. 1. Specifically, the channel estimation is obtained by using the channel quality metric of signal to residual interference plus noise ratio (SINR), which can be calculated at the output of MMSE-BDFE [11].

TABLE I
BIT ALLOCATION OF THE AMR SPEECH CODEC AT 4.75 kb/s AND 10.2 kb/s [6]. THE BIT POSITIONS FOR 4.75 kb/s MODE, WHICH ARE SHOWN IN PARENTHESES, ASSIST IN IDENTIFYING THE CORRESPONDING BITS IN FIGS. 2 AND 4

Mode	Parameter	1st subframe	2nd subframe	3rd subframe	4th subframe	Total per Frame
4.75 kbps	LSFs					8+8+7=23 (1-23)
	Pitch Delay	8 (24-31)	4 (49-52)	4 (62-65)	4 (83-86)	20
	Fixed Codebook Index	9 (32-40)	9 (53-61)	9 (66-74)	9 (87-95)	36
	Codebook Gains	8 (41-48)		8 (75-82)		16
	Total	95/20 ms=4.75 kbps				
10.2 kbps	LSFs					8+9+9=26
	Pitch Delay	8	5	8	5	26
	Fixed Codebook Index	31	31	31	31	124
	Codebook Gains	7	7	7	7	28
	Total	204/20 ms=10.2 kbps				

III. AMR CODEC'S MODE SWITCHING PHILOSOPHY

The AMR speech codec [6], [7], [29] employs the algebraic code-excited linear predictive (ACELP) excitation model [24], [25]. This codec operates on a 20 ms frame of 160 speech samples, and generates encoded blocks of 95, 103, 118, 134, 148, 159, 204, and 244 bits/20 ms. This leads to bit rates of 4.75, 5.15, 5.9, 6.7, 7.4, 7.95, 10.2, and 12.2 kb/s, respectively. Explicitly, the AMR speech codec provides eight different modes. Multirate coding [26] allows a variation in the total allocation of bits for a speech frame, adapting the rate to the local phonetic character of the speech signal to the channel quality or to network traffic conditions. This is particularly useful in digital cellular communications, where one of the major challenges is that of designing a codec that is capable of providing high quality speech for a wide variety of channel conditions. Ideally, a good solution must provide the highest possible quality under perfect channel conditions, while also maintaining good quality in hostile channel environments. The codec mode adaptation is a key feature of the new AMR standard that has not been used in any prior mobile standard. At a given fixed gross bit rate, this mechanism of adapting the source coding rate has the potential of altering the partitioning between the speech source bit rate and the redundancy added for error protection. Alternatively, the AMR codec can be invoked in our BbB-AQAM/CDMA transceiver.

In the AMR codec, the mode adaptation allows us to invoke a subset of at most four modes out of the eight available modes [34]. This subset is referred to as the active codec set (ACS). In the proposed BbB-AQAM/CDMA system, the codec mode adaptation is based on the channel quality, which is expressed as the MSE at the output of the multiuser CDMA detector [11]. The probability of switching from one mode to another is typically lower than the probability of sustaining a specific mode.

Intuitively, frequent mode switching is undesirable due to the associated perceptual speech quality fluctuations. It is more desirable to have a mode selection mechanism that is primarily source-controlled, assisted by a channel-quality-controlled override. During good channel conditions, the mode switching process is governed by the local phonetic character of the speech signal, and the codec will adapt itself to the speech signal characteristics in an attempt to deliver the highest possible speech quality. When the channel is hostile or the network is congested, transceiver control or external network control can take over the mode selection and allocate less bits to source coding, in order to increase the system's robustness or user capacity. By amalgamating the channel quality-motivated or network- and source-controlled processes, it results in a robust, high quality system. Surprisingly, we found from our informal listening tests that the perceptual speech qual-

ity was not affected by the rate of codec mode switching, as it will be demonstrated in Section VII. This is due to the robust ACELP structure, whereby the main bit rate reduction is related to the fixed codebook indices, as shown in Table I for the codec modes of 4.75 kb/s and 10.2 kb/s.

As expected, the performance of the AMR speech codec is sensitive to transmission errors of the codec mode information. The corruption of the codec mode information that describes which codec mode has to be used for decoding leads to complete speech frame losses, since the decoder is unable to apply the correct mode for decoding the received bit stream. Hence, robust channel coding is required in order to protect the codec mode information and the recommended transmission procedures were discussed, for example, by Bruhn *et al.* [29]. Furthermore, in transceiver-controlled scenarios, the prompt transmission of the codec mode information is required to react to sudden changes of the channel conditions. In our investigations, we assume that the signaling of the codec mode information, is free from corruption, so that we can concentrate on other important aspects of the system.

Let us now focus on the robustness of the AMR codec against channel errors.

IV. SPEECH CODEC'S ERROR SENSITIVITY

In this section, we will demonstrate that some bits are significantly more sensitive to channel errors than others, and hence must be better protected by the channel codec [8]. A commonly used approach in quantifying the sensitivity of a given bit is to invert the bit consistently in every speech frame, and evaluate the associated segmental SNR (SEGSNR) degradation. The error sensitivity of various bits for the AMR codec determined in this way is shown in Fig. 2 for the bit rate of 4.75 kb/s. Again, Fig. 2 shows more explicitly the bit sensitivities in each speech subframe for the bit rate of 4.75 kb/s, with the corresponding bit allocations shown in Table I. For the sake of visual clarity, Subframe 4 (bit positions 83–95) was not shown explicitly in Fig. 2, since it exhibited identical SEGSNR degradations to Subframe 2.

It can be observed from Fig. 2 that the most sensitive bits are those of the LSF subvectors, seen at positions 1–23. The error sensitivity of the adaptive codebook delay is the highest in the first subframe, commencing at bit 24, as shown in Fig. 2, which was encoded using eight bits in Table I. By contrast, the relative adaptive codebook delays in the next three subframes are encoded using four bits each, and a graceful degradation of the SEGSNR is observed in Fig. 2 at bit positions 49–52, 62–65, and 83–86. The next group of bits is constituted by the eight codebook gains in decreasing order of bit sensitivity, as seen in Fig. 2 at bit positions 41–48 for Subframe 1 and 75–82 for Subframe 3. The

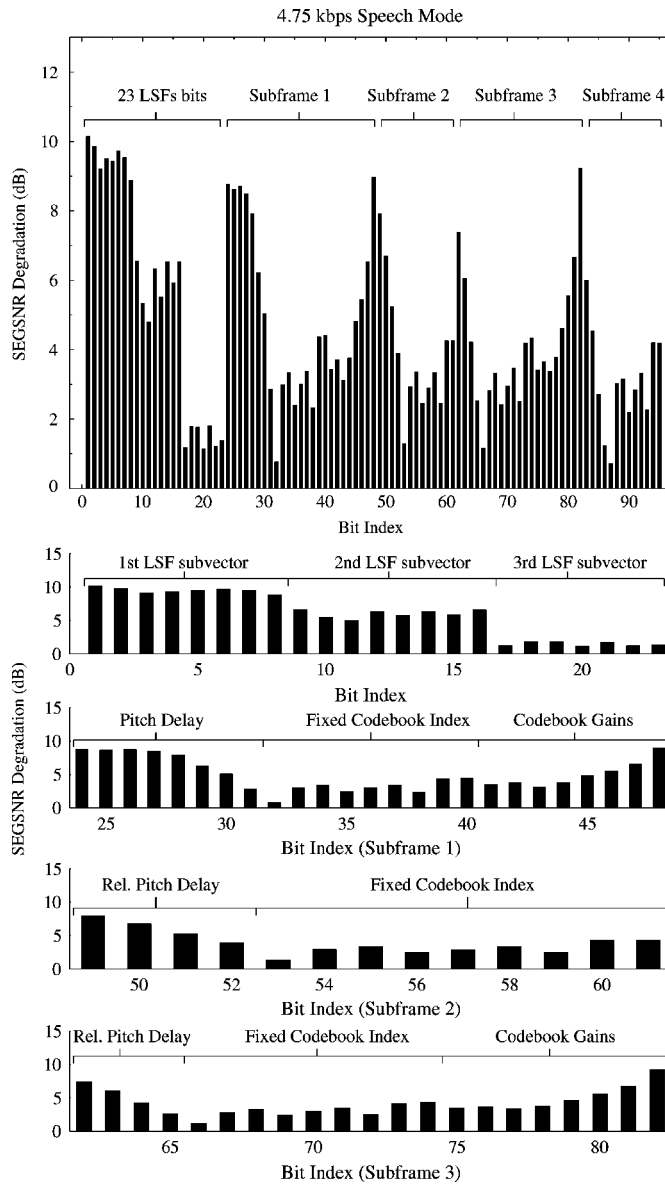


Fig. 2. SEGSNR degradations due to 100% bit error rate in the 95-bit, 20 ms AMR speech frame. The associated bit allocation can be seen in Table I.

least sensitive bits are related to the fixed codebook pulse positions, which were shown for example at bit positions 54–61 in Fig. 2. This is because if one of the fixed codebook index bits is corrupted, the codebook entry selected at the decoder will differ from that used in the encoder only in the position of one of the nonzero excitation pulses. Hence, the corrupted codebook entry will be similar to the original one. Therefore, the algebraic codebook structure used in the AMR codec is inherently quite robust to channel errors. The information obtained here will be used to design the bit mapping procedure in order to assign the channel encoders according to the bit error sensitivities.

Despite its appealing conceptual simplicity, the preceding approach used for quantifying the error sensitivity of the various coded bits does not illustrate the error propagation properties of different bits over consecutive speech frames. In order to obtain a better picture of the error propagation effects, we also employed a more elaborate error sensitivity measure. Here, for each bit, we find the average SEGSNR degradation due to a single bit error, both in the frame in which the

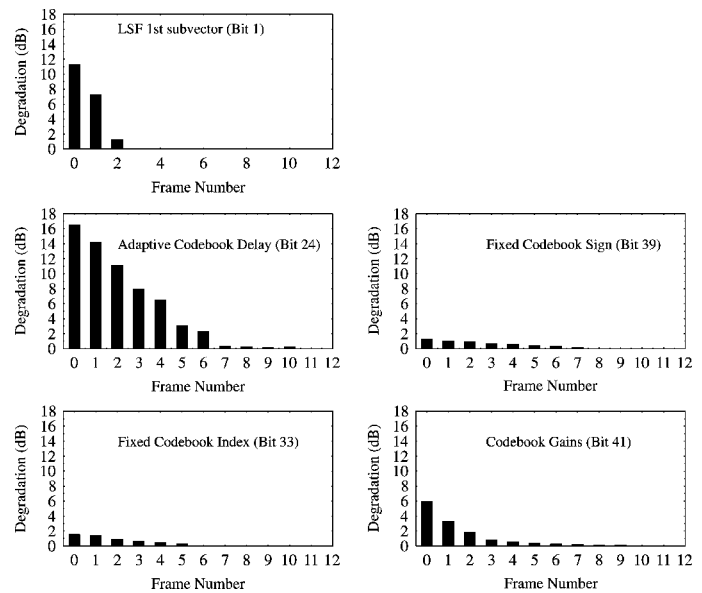


Fig. 3. SEGSNR degradation versus speech frame index for various bits.

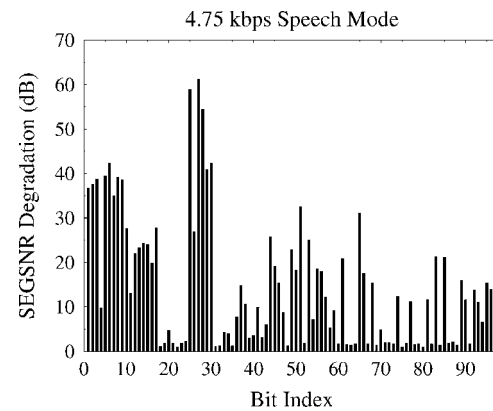


Fig. 4. Average SEGSNR degradation due to single bit errors in various speech coded bits.

error occurs, and in consecutive frames. These effects are exemplified in Fig. 3 for five different bits, where each of the bits belongs to a different speech codec parameter. More explicitly, Bit 1 represents the first bit of the first LSF subvector, which shows some error propagation effects due to the interpolation between the LSFs over consecutive frames. The associated SEGSNR degradation dies away over six frames. Bit 24, characterized in Fig. 3, is one of the adaptive codebook delay bits, and the corruption of this bit has the effect of a more prolonged SEGSNR degradation over ten frames. The fixed codebook index bits of Table I are more robust, as it was shown previously in Fig. 2. This argument is supported by the example of Bit 33 in Fig. 3, where a small and insignificant degradation over consecutive frames is observed. A similar observation also applies to Bit 39 in Fig. 3, which is the sign bit for the fixed codebook. By contrast, Bit 41 of the codebook gains produced a high and prolonged SEGSNR degradation profile.

We recomputed our bit sensitivity results of Fig. 2 using this second approach, in order to obtain Fig. 4, taking into account the error propagation effects. More explicitly, these results were calculated by summing the SEGSNR degradations over all frames, which were affected by the error. Again, these results are shown in Fig. 4, and the associated bit positions can be identified with the aid of Table I. The

importance of the adaptive codebook delay bits became more explicit. By contrast, the significance of the LSFs was reduced, although still requiring strong error protection using channel coding.

Having characterized the error sensitivity of the various speech bits, we will capitalize on this knowledge in order to assign the speech bits to various bit protection classes, as will be discussed in Section V-B. Let us now consider the channel coding aspects of our transceiver in the next section.

V. REDUNDANT RESIDUE NUMBER SYSTEM BASED CHANNEL CODING

A. Overview

In order to improve the performance of the system, we employ the novel family of the so-called redundant residue number system (RRNS) codes for protecting the speech bits unequally, depending on their respective error sensitivities.

Since their introduction, redundant residue number systems (RRNS) have been used for constructing fast arithmetics [35], [36]. In this paper, we exploit the error control properties of nonbinary systematic RRNS codes, which—similarly to Reed–Solomon codes—exhibit maximum minimum distance properties [37], [38]. Hence, RRNS codes are similar to Reed Solomon (RS) codes [32]. However, the RRNS codes chosen in our design are more amenable to implementing short codes. More explicitly, in the context of RS codes, short codes are derived by inserting dummy symbols into full-length codes. This, however, requires the decoding of the full-length RS-code. By contrast, RRNS codes simply add the required number of redundant symbols. Furthermore, RRNS codes allow us to use the low-complexity technique of residue dropping [38]. Both of these advantages will be augmented in further discourse.

An $RRNS(n, k)$ code has k so-called residues, which host the original data bits, and the additional $(n - k)$ redundant residues can be employed for error correction at the decoder. The coding rate of the code is k/n , and the associated error correction capability of the code is $t = \lfloor (n - k)/2 \rfloor$ nonbinary residues [37], [38]. At the receiver, soft decision [2] and residue dropping [39] decoding techniques are employed.

The advantages of the RRNS codes are simply stated here without proof due to lack of space [2], [39]–[41]. Since the so-called residues of the RRNS [35], [36] can be computed independently from each other, additional residues can be added at any stage of processing or transmission [2], [13]. This has the advantage that the required coding power can be adjusted according to the prevalent bit error rate (BER) of the transmission medium. For example, when the protected speech bits enter the wireless section of the network—where higher BERs prevail than in the fixed network—simply a number of additional redundant residues are computed and concatenated to the message to provide extra protection.

In our design, RRNS codes employing five bits per residue have been chosen. Three different RRNS codes having different code rates are used to protect the three different classes of speech bits. In addition, the RRNS codes employed are also switched in accordance with the modulation modes and speech rates used in our system. In Table II, we have two sets of RRNS codes for the BPSK and 4QAM modes. For the most sensitive class I speech bits, we used a $RRNS(8, 4)$ code, which has a minimum free distance of $d_{\min} = 5$ [2] and a code rate of $1/2$. At the receiver, the soft metric of each received bit was calculated and soft decoding was applied. An extra information residue was added to the $RRNS(8, 4)$ code to generate the $RRNS(8, 5)$ code for the protection class II. The extra residue enables us to apply one residue dropping [39], and soft decision decoding. The Class III bits

TABLE II
RRNS CODES DESIGNED FOR TWO DIFFERENT MODULATION MODES

Class	RRNS Code	Number of Codewords	databits	Total databits	Total codedbits
4.75 kbps/BPSK					
I	RRNS(8,4)	2	40	95	160
II	RRNS(8,5)	1	25		
III	RRNS(8,6)	1	30		
10.2 kbps/4QAM					
I	RRNS(8,4)	3	60	205	320
II	RRNS(8,5)	1	25		
III	RRNS(8,6)	4	120		

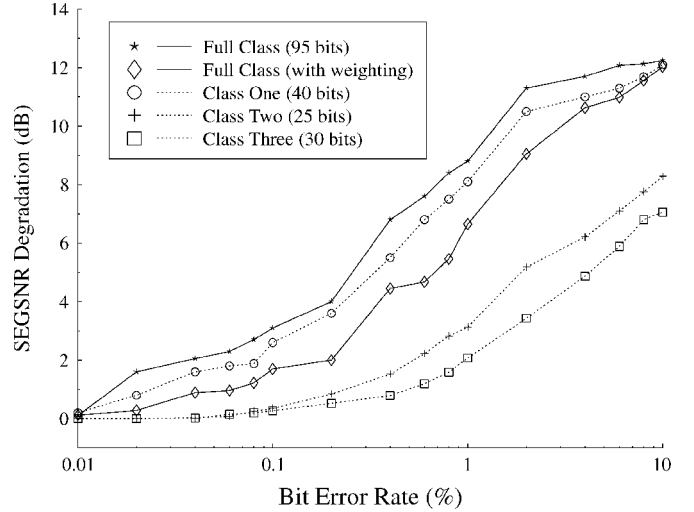


Fig. 5. SEGNSR degradation versus average BER for the 4.75 kb/s AMR codec for full-class and triple-class protection systems. When the bits of a specific class were corrupted, bits of the other classes were kept intact.

are least protected, using the $RRNS(8, 6)$ code, which has a minimum free distance of $d_{\min} = 3$ and a code rate of $2/3$. Only soft decision decoding is applied to this code.

B. Source-Matched Error Protection

The error sensitivity of the 4.75 kb/s AMR codec's source bits was evaluated in Fig. 2 and 4. The same procedures were applied in order to obtain the error sensitivity for the source bits of the 10.2 kb/s AMR codec. Again, in our system, we employed RRNS channel coding and three protection classes were deemed to constitute a suitable tradeoff between the system's complexity and performance. As shown in Table II, three different RRNS codes having different code rates are used to protect the three different classes of speech bits in a speech frame.

For the 4.75 kb/s AMR speech codec, we divided the 95 speech bits into three sensitivity classes, Class I, II, and III. Class I consists of 40 bits, while Class II and III were allocated 25 and 30 bits, respectively. Then, we evaluated the associated SEGNSR degradation inflicted by certain fixed channel BERs maintained in each of the classes using randomly distributed errors, while keeping bits of the other classes intact. The results of the SEGNSR degradations applying random errors are portrayed in Fig. 5 for both the full-class and the triple-class system. It can be seen that Class I, which consists of the 40 most sensitive bits, suffers the highest SEGNSR degradation. Class II and Class III—which are populated mainly with the fixed codebook index bits—are inherently more robust to errors. Note that in the full-class scenario, the

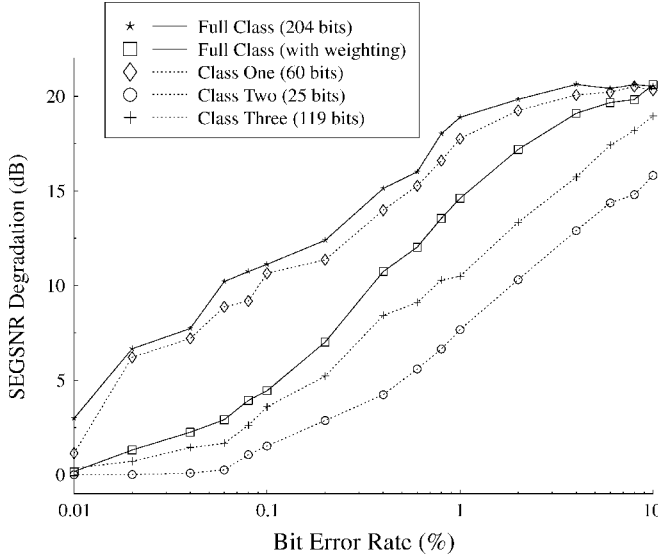


Fig. 6. SEGNSR degradation versus average BER for the 10.2 kb/s AMR codec for full class and triple-class protection systems. When the bits of a specific class were corrupted, bits of the other classes were kept intact.

associated SEGNSR degradation is higher than that of the individual protection classes. This is due to having more errors in the 95-bit frame at a fixed BER, compared to the individual protection classes, since upon corrupting a specific class using a fixed BER, the remaining classes were intact. Hence, the BER averaged over all the 95 bits was lower than that of the full-class scenario. For the sake of completeness, we decreased the BER of the full-class scheme so that, on average, the same number of errors was introduced into the individual classes as well as in the full-class scheme. In this scenario, it can be seen from Fig. 5 that, as expected, the Class I scheme has the highest SEGNSR degradation, while the sensitivity of the full-class scheme is mediocre.

Similarly, the 204 bits of a speech frame in the 10.2 kb/s AMR speech codec mode are divided into three protection classes. Class I is allocated the 60 most sensitive bits, while 25 and 119 bits are distributed to Class II and Class III, in decreasing order of error sensitivity. Their respective SEGNSR degradation results against the BER are presented in Fig. 6. Due to the fact that the number of bits in Class III is five times higher than in Class II, the error sensitivity of Class III compared to Class II appeared higher. This occurs due to the nontrivial task of finding appropriate channel codes to match the source sensitivities, and as a result, almost sixty percent of the bits are allocated to Class III. Note that after the RRNS channel coding stage, an additional dummy bit is introduced in Class III, which contains 119 useful speech bits, as shown in Table II. The extra bit can be used as a cyclic redundancy check (CRC) bit for the purpose of error detection. Having considered the source and channel coding aspects, let us now focus our attention on transmission issues.

VI. JOINT DETECTION CODE DIVISION MULTIPLE ACCESS

A. Overview

Joint detection receivers [42] constitute a class of multiuser receivers that were developed based on conventional equalization techniques [15], used for mitigating the effects of intersymbol interference (ISI). These receivers utilize the channel impulse response (CIR) estimates and the knowledge of the spreading sequences of all

the users in order to reduce the level of multiple access interference (MAI) in the received signal.

By concatenating the data symbols of all CDMA users successively (as though they were transmitted by one user) we can apply the principles of conventional TDMA-oriented channel equalization [15] to multiuser detection. In our investigations, we have used the MMSE-BDFE proposed by Klein *et al.* [42], where the multiuser receiver aims to minimize the mean square error between the data estimates and the transmitted data. A feedback process is incorporated, where in the previous data estimates are fed back into the receiver in order to remove the residual interference and assist in improving the BER performance.

B. Joint Detection Based Adaptive Code Division Multiple Access

In QAM [15], n bits are grouped to form a signaling symbol and $m = 2^n$ different symbols convey all combinations of the n bits. These m symbols are arranged in a constellation to form the m -QAM scheme. In the proposed system we used the BbB-AQAM/CDMA modes of BPSK (2-QAM) and 4QAM, conveying 1 and 2 bits per symbol, respectively. However, for a given channel signal to noise ratio (SNR), the BER performance degrades upon switching from BPSK to 4QAM, while doubling the throughput.

Previous research in BbB-AQAM schemes for TDMA transmissions has been carried out by Webb and Steele [16]; Sampei, Komaki and Morinaga [18]; Goldsmith and Chua [17]; and Torrance *et al.* [44]. This work has been extended to wideband channels, where the received signal also suffers from intersymbol interference (ISI) in addition to amplitude and phase distortions due to the fading channel. The received signal strength is not a good indicator of the wideband channel's quality, since the signal is also contaminated by ISI. Wong *et al.* [1] proposed a wideband BbB-AQAM scheme, where a channel equalizer was used to mitigate the effects of ISI on the CIR estimate.

Here we propose to combine joint detection CDMA [42] with AQAM by modifying the approach used by Wong *et al.* [1]. Joint detection is particularly suitable for combining with AQAM, since the implementation of the joint detection algorithm does not require any knowledge of the modulation mode used [11]. Hence, the associated complexity is independent of the modulation mode used.

In order to choose the most appropriate BbB-AQAM/CDMA mode for transmission, the channel quality was quantified in terms of the SINR at the output of the MMSE-BDFE by modifying the SINR expression given in [42]. More explicitly the SINR $\gamma(k, n)$ encountered at the output of the MMSE-BDFE was estimated by assuming perfect CIR estimation for the n th symbol of the k th user as follows [42]

$$\gamma(k, n) = g^2 [\mathbf{D}]_{j,j}^2 - 1 \quad (1)$$

where $n = 1, \dots, N$, and $k = 1, \dots, K$, $j = n + N(k-1)$, and g^2 is the channel-induced gain or attenuation of the transmitted signal. Furthermore, \mathbf{D} is a diagonal matrix having real-valued elements obtained from the upper triangular matrix \mathbf{U} such that $\mathbf{D}\mathbf{U}' = \mathbf{U}$, and \mathbf{U}' is also an upper triangular matrix where all the elements on the main diagonal have values of unity [42]. The data bits and noise values were assumed to be uncorrelated. The average output SINR was calculated for each transmission burst of each user. The conditions used to switch between the two AQAM/JD-CDMA modes were set according to their target BER requirements as

$$\text{Mode} = \begin{cases} \text{BPSK}, & \text{SINR} < t_1 \\ 4\text{QAM}, & t_1 \leq \text{SINR} \end{cases} \quad (2)$$

where t_1 represents the switching threshold between the two modes. The value of t_1 determines both the integrity and throughput of the system and the higher the value of t_1 , the higher the SINR, at which

TABLE III
TRANSCIVER PARAMETERS

Parameter	Value
Channel type	COST 207 Bad Urban (BU)
Paths in channel	7
Doppler frequency	80 Hz
Spreading factor	16
Chip rate	2.167 MBaud
JD block size	26 symbols
Receiver type	MMSE-BDFE
AQAM type	Dual-mode (BPSK, 4QAM)
Channel codec	Triple-class RRNS
Channel-coded Rate	8/16 kbps
Speech Codec	AMR (ACELP)
Speech Rate	4.75/10.2 kbps
Speech Frame Length	20 ms

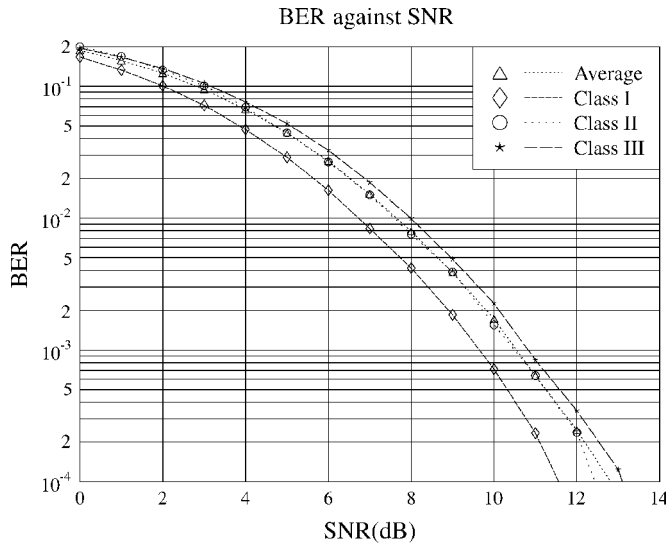


Fig. 7. BER performance of 4QAM/JD-CDMA over the COST 207 BU channel of Table III using the RRNS codes of Table II.

point the transceiver activates its higher-throughput mode, which results in maintaining a higher integrity but lower throughput. The more error-resilient the channel-coded speech codec, the lower t_1 value may be used, which results in a better perceptual speech quality. Theoretical switching threshold formulas have been developed in [1] for various scenarios, while Tang [43] proposed an intelligent learning scheme for the appropriate adjustment of the AQAM switching thresholds, where the current integrity is estimated and the switching thresholds are increased or reduced appropriately for the sake of maintaining target integrity.

With the system elements described, we now focus on the overall performance of the proposed adaptive transceiver.

VII. SYSTEM PERFORMANCE

The simulation parameters used in our AQAM/JD-CDMA system are listed in Table III. The channel profile used was the COST 207 bad urban (BU) channel [1], [45] consisting of seven paths, where each path was faded independently at a Doppler frequency of 80 Hz.

The BER performance of the proposed system is presented in Figs. 7, 8, and 9. Specifically, Fig. 7 portrays the BER performance using the 4QAM modulation mode and employing the RRNS codes of Table II

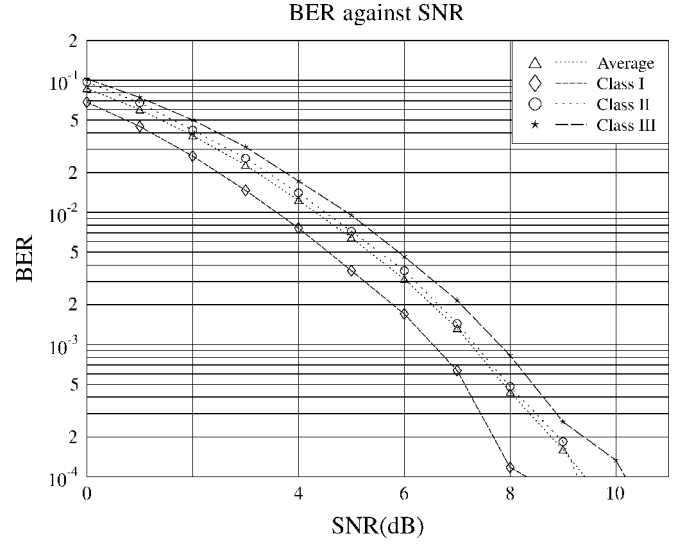


Fig. 8. BER performance of BPSK/JD-CDMA over the COST 207 BU channel of Table III using the RRNS codes of Table II.

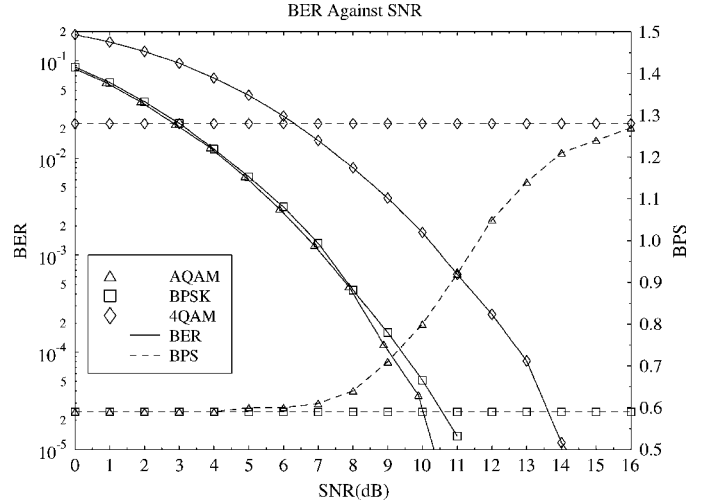


Fig. 9. BER and BPS comparisons for fixed mode BPSK and 4QAM, as well as for the AQAM/JD-CDMA system, using the RRNS codes of Table II. The switching threshold for AQAM was set to 10.5 dB and the simulation parameters are listed in Table III.

for a two-user JD-CDMA speech transceiver. As seen in Table II of Section V-B, three different RRNS codes having different code rates are used to protect the three different classes of speech bits in the speech codec. The BER of the three protection classes is shown together with the average BER of the channel coded bits versus the channel SNR. The number of bits in these protection classes was 60, 25, and 120, respectively. As expected, the Class I subchannel exhibits the highest BER performance, followed by the Class II and Class III subchannels in decreasing order of BER performance. The corresponding BER results for the BPSK/JD-CDMA mode are shown in Fig. 8.

In Fig. 9, the average BER performance of the coded fixed-mode BPSK/JD-CDMA and 4QAM/JD-CDMA systems is presented along with that of the twin-mode AQAM/JD-CDMA system supporting two users, and assuming zero-latency modem mode signaling. The performance of the AQAM scheme was evaluated by analyzing the BER and the throughput expressed in terms of the average number of bits per symbol (BPS) transmitted. The BER curve has to be read by referring

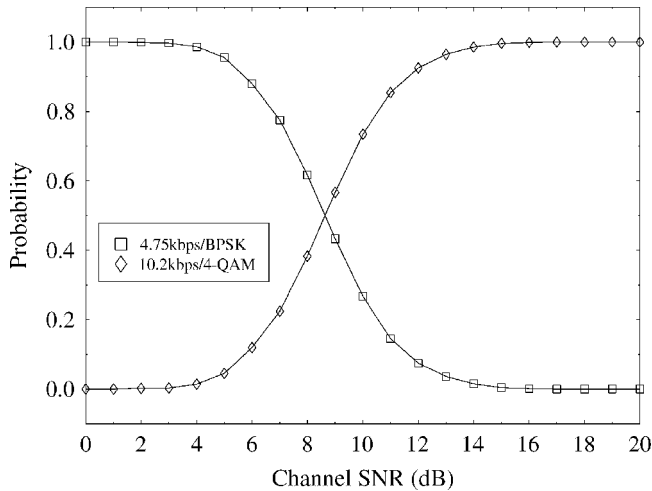


Fig. 10. Probability of each modulation mode being chosen for transmission in a twin-mode (BPSK, 4QAM), two-user AQAM/JD-CDMA system using the parameters of Table III.

to the vertical axis at the left of the figure, while the BPS throughput curve is interpreted by referring to the vertical-axis at the right that is labeled BPS. At low channel SNRs, the BER of the AQAM/JD-CDMA scheme mirrored that of BPSK/JD-CDMA, which can be explained using Fig. 10. In Fig. 10, the activation probability of the two different AQAM/JD-CDMA modes versus the channel SNR is plotted. As mentioned previously, the results were obtained using an SINR switching threshold of 10.5 dB, a value, which resulted in a good compromise in terms of maintaining the highest possible throughput at a target integrity of about 1%, which is required by the AMR speech codec for maintaining the best possible perceptual speech quality. We can see from the figure that at low *average* channel SNRs (<6 dB), the threshold of 10.5 dB instantaneous SNR was seldom reached, and therefore BPSK/JD-CDMA was the predominant mode. Hence, the performance of the AQAM/JD-CDMA scheme was similar to BPSK/JD-CDMA. However, as the channel SNR increased, the BER performance of AQAM/JD-CDMA became better than that of BPSK/JD-CDMA, as shown in Fig. 9. This is because the 4QAM mode is employed more often, reducing the probability of using BPSK, as shown in Fig. 10. Since the mean BER of the system is the ratio of the total number of bit errors to the total number of bits transmitted, the mean BER will decrease with a decreasing number of bit errors or with an increasing number of transmitted bits. For a fixed number of symbols transmitted, the total number of transmitted bits in a frame is constant for fixed mode BPSK/JD-CDMA, while for AQAM/JD-CDMA, the total number of transmitted bits increased when the 4QAM/JD-CDMA mode was used. Consequently, the average BER of the AQAM/JD-CDMA system was lower than that of the fixed-mode BPSK/JD-CDMA scheme.

The BPS throughput performance curve is also plotted in Fig. 9. As expected, the number of BPS of both BPSK and 4QAM is constant for all channel SNR values. The BPS throughput is limited by the modulation scheme used and the coding rate of the RRNS codes shown in Table II. For example, for 4QAM we have 2 BPS, but the associated channel code rate is 205/320, as shown in Table II, hence the effective throughput of the system is $2 \times (205/320) = 1.28$ BPS. For AQAM/JD-CDMA, we can see from Fig. 9 that the throughput is similar to that of BPSK/JD-CDMA at low channel SNRs. However, as the average channel SNR increased, more and more frames were transmitted using 4QAM/JD-CDMA, and the average throughput increased gradually. At high average SNRs, the throughput of AQAM/JD-CDMA became similar to that of the 4QAM/JD-CDMA scheme.

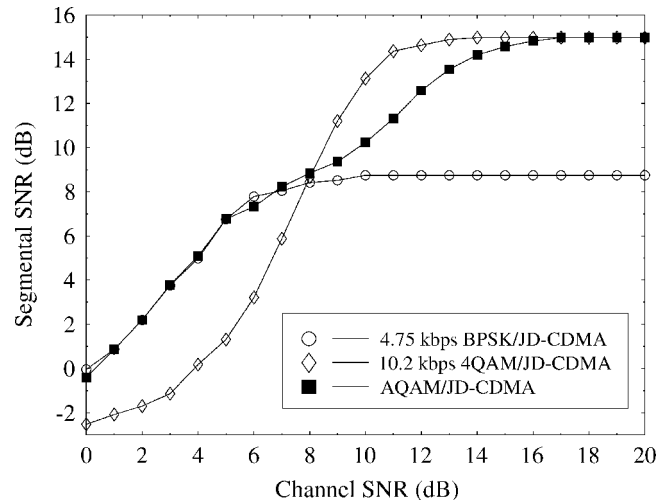


Fig. 11. SEGSR versus Channel SNR.

The overall SEGSR versus channel SNR performance of the proposed speech transceiver is displayed in Fig. 11. Observe that the source sensitivity-matched triple-class 4.75 kb/s BPSK/JD-CDMA system requires a channel SNR in excess of about 8 dB for nearly unimpaired speech quality over the COST207 BU channel of Table III. When the channel SNR was in excess of about 12 dB, the 10.2 kb/s 4QAM/JD-CDMA system outperformed the 4.75 kb/s BPSK/JD-CDMA scheme in terms of both objective and subjective speech quality. Furthermore, at channel SNRs around 10 dB, where the BPSK and 4QAM SEGSR curves cross each other in Fig. 11, it was preferable to use the inherently lower quality but unimpaired mode of operation. In the light of these findings, the application of the AMR speech codec in conjunction with AQAM constitutes an attractive tradeoff in terms of providing users with the best possible speech quality under arbitrary channel conditions. Specifically, the 10.2 kb/s 4QAM/JD-CDMA scheme has the highest source bit rate, and thus exhibits the highest SEGSR under error-free conditions. The 4.75 kb/s BPSK/JD-CDMA scheme exhibits a lower source bit rate and correspondingly lower speech quality under error-free conditions. However, due to its less robust modulation mode, the 10.2 kb/s 4QAM/JD-CDMA scheme is sensitive to channel errors and breaks down under hostile channel conditions, where the 4.75 kb/s BPSK/JD-CDMA scheme still exhibits robust operation, as illustrated in Fig. 11.

In the context of Fig. 11, ideally, a system is sought that achieves a SEGSR performance which follows the envelope of the SEGSR curves of the individual BPSK- and 4QAM/JD-CDMA modes. The SEGSR performance of the AQAM system is also displayed in Fig. 11. We observe that AQAM provides a smooth evolution across the range of channel SNRs. At high channel SNRs in excess of 12–14 dB, the system operates predominantly in the 4QAM/JD-CDMA mode. As the channel SNR degrades below 12 dB, some of the speech frames are transmitted in the BPSK/JD-CDMA mode, which implies that the lower quality speech rate of 4.75 kb/s is employed. This results in a slightly degraded average speech quality, while still offering a substantial SEGSR gain compared to the fixed-mode 4.75 kb/s BPSK/JD-CDMA scheme. At channel SNRs below 10 dB, the performance of the 10.2 kb/s 4QAM/JD-CDMA mode deteriorates due to the occurrence of a high number of errors, inflicting severe SEGSR degradations. In these hostile conditions, the 4.75 kb/s BPSK/JD-CDMA mode provides a more robust performance associated with a better speech quality. With the advent of the AQAM/JD-CDMA mode switching

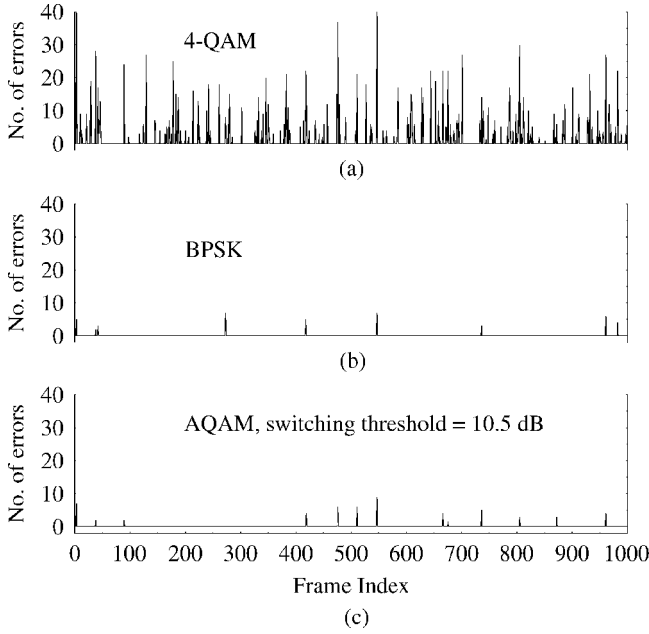


Fig. 12. Comparison of the number of errors per frame versus 20 ms frame index for the (a) 4QAM, (b) BPSK, and (c) AQAM/JD-CDMA systems with a switching threshold of 10.5 dB, at channel SNR = 10 dB for 1000 frames over the COST207 BU channel of Table III.

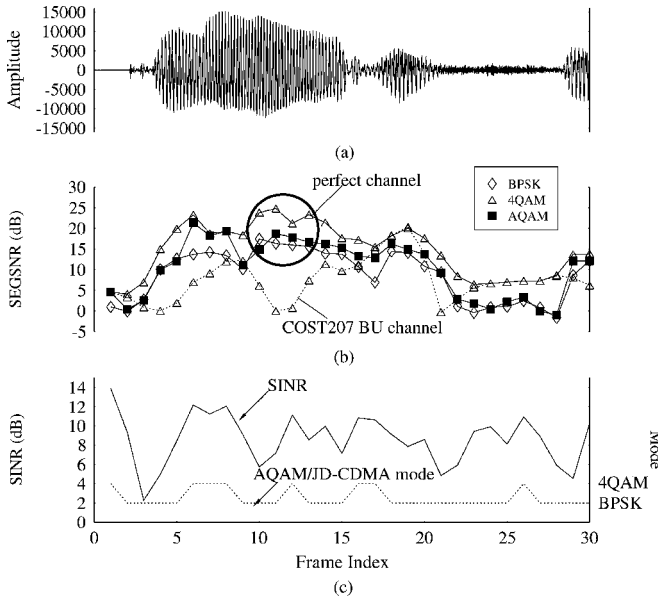


Fig. 13. Characteristic waveforms of the adaptive system. (a) Time-domain speech signal. (b) SEGSR in various transceiver modes. (c) SINR versus time and transceiver modes versus time over the COST207 BU channel of Table III.

regime, the transceiver exhibits a less bursty error distribution, than that of the conventional fixed-mode 4QAM modem, as can be seen in Fig. 12 where the error events of the BPSK/JD-CDMA scheme are also displayed.

The benefits of the proposed dual-mode transceiver are further demonstrated by Fig. 13, consisting of three graphs plotted against the speech frame index, giving an insightful characterization of the adaptive speech transceiver. Fig. 13(a) shows a speech segment of 30 frames. In the AMR codec, a speech frame corresponds to a duration

TABLE IV
DETAILS OF THE LISTENING TESTS CONDUCTED USING THE PAIRWISE COMPARISON METHOD, WHERE THE LISTENERS WERE GIVEN A CHOICE OF PREFERENCE BETWEEN TWO SPEECH FILES CODED IN DIFFERENT TRANSMISSION SCENARIOS

Speech Material A	Speech Material B	Preference		
		A (%)	B (%)	No Preference(%)
4.75 kbps(Error free)	10.2 kbps (Error free)	4.15	66.65	29.2
AQAM(9 dB)	4QAM(9 dB)	100	0.00	0.00
AQAM(9 dB)	4QAM(11 dB)	8.3	50.0	41.7
AQAM(9 dB)	BPSK(9 dB)	37.5	16.65	45.85
AQAM(12 dB)	4QAM(12 dB)	4.15	20.85	75.0
AQAM(12 dB)	4QAM(13 dB)	8.3	25.0	66.7
AQAM(12 dB)	BPSK(12 dB)	41.65	8.3	50.05

of 20 ms. In Fig. 13(b), the SEGSR versus frame index performance curves of the BPSK, 4QAM, and AQAM/JD-CDMA schemes are shown in both error-free and channel-impaired scenarios. The SINR at the output of the MMSE-BDFE is displayed in Fig. 13(c). The adaptation of the modulation mode is also shown in Fig. 13(c), where the transceiver switches to the BPSK or 4QAM mode according to the estimated SINR, using the switching threshold set to 10.5 dB.

When transmitting in the less robust 4QAM mode using the higher-rate speech mode of 10.2 kb/s, a sudden steep drop in the channel conditions—as portrayed at Frame 1 in Fig. 13—results in a high number of transmission errors, as also illustrated in Fig. 12(a). This happens to occur during the period of voice onset in Fig. 13, resulting in the corruption of the speech frame, which has the effect of inflicting impairments to subsequent frames due to the error propagation effects of various speech bits, as alluded to in Section IV. It can be seen in Fig. 13 that the high number of errors inflicted in the 4QAM mode during voiced speech segments caused a severe SEGSR degradation at frame index 10, and the 10.2 kb/s speech codec never fully recovered until the channel conditions expressed in terms of the SINR in Fig. 13(c) improved. On the other hand, the significantly more robust 4.75 kb/s BPSK/JD-CDMA scheme performed well under these hostile channel conditions, encountering a low number of errors in Fig. 12(b) while transmitting at a lower speech rate, hence at an inherently lower speech quality. For visual clarity, the performance curves of BPSK/JD-CDMA and AQAM/JD-CDMA were not displayed in Fig. 13(b) for the channel-impaired scenarios, since their respective graphs are almost identical to that of the error-free speech SEGSR curves.

A. Subjective Testing

Informal listening tests were conducted in order to assess the performance of the AQAM/JD-CDMA scheme in comparison to the fixed-mode BPSK/JD-CDMA and 4QAM/JD-CDMA schemes. It is particularly revealing to investigate how the AQAM/JD-CDMA scheme performs in the intermediate channel SNR region between 7 dB and 11 dB. The speech quality was assessed using pairwise comparison tests. The listeners were asked to express a preference between two speech files A or B, or neither. A total of 12 listeners were used in the pairwise comparison tests. Four different utterances were employed during the listening tests, where the utterances were due to a mixture of male and female speakers having American accents. Table IV details some of the results of the listening tests.

Through the listening tests, we found that for the fixed-mode BPSK/JD-CDMA scheme, unimpaired perceptual speech quality was achieved for channel SNRs in excess of 7 dB. With reference to Fig. 11, when the channel conditions degraded below 7 dB, the speech quality became objectionable due to the preponderance of channel

TABLE V
FRAME SWITCHING FREQUENCY VERSUS SEGSNR

Frame Switching Frequency	SEGSNR (dB)
1	11.38
10	11.66
100	11.68

errors. For the fixed mode 4QAM/JD-CDMA scheme, the channel SNR threshold was 11 dB, below which the speech quality started to degrade. The perceptual performance of AQAM/JD-CDMA, was to be found superior to that of 4QAM/JD-CDMA at channel SNRs below 11 dB. Specifically, it can be observed from Table IV that all the listeners preferred the AQAM/JD-CDMA scheme at a channel SNR of 9 dB due to the associated high concentration of channel errors in the less robust 4QAM/JD-CDMA scheme at the same channel SNR, resulting in a perceptually degraded reconstructed speech quality.

More explicitly, we opted for investigating the AQAM/JD-CDMA scheme at a channel SNR of 9 dB since, as shown in Fig. 10, it switches between BPSK/JD-CDMA and 4QAM/JD-CDMA according to the ratio of about 50:50. As the channel conditions improved to an SNR in excess of 11 dB, the 4QAM/JD-CDMA scheme performed slightly better than AQAM/JD-CDMA due to its inherently higher SEGSNR performance under error free conditions. Nonetheless, the AQAM/JD-CDMA scheme provided a good perceptual performance, as exemplified in Table IV at a channel SNR of 12 dB, in comparison to the 4QAM/JD-CDMA scheme at the channel SNRs of both 12 dB and 13 dB. Here, only about twenty percent of the listeners preferred the 4QAM/JD-CDMA scheme to the AQAM/JD-CDMA scheme, while the rest suggested that both sounded very similar. It can also be observed from Table IV that the AQAM/JD-CDMA scheme performed better than BPSK/JD-CDMA for a channel SNR of 7 dB and above, while in the region below 7 dB, AQAM/JD-CDMA has a similar perceptual performance to that of BPSK/JD-CDMA. As shown in Table V, we found that changing the mode switching frequency for every 1, 10, or 100 frames does not impair the speech quality either in objective SEGSNR terms, or in terms of informal listening tests.

VIII. CONCLUSION

In this contribution, a joint-detection aided adaptive CDMA speech transceiver has been designed that allows us to switch between a set of different source and channel coders as well as transmission parameters, depending on the overall instantaneous channel quality. The benefits of the multimode speech transceiver clearly manifest themselves in terms of supporting unimpaired speech quality under time-variant channel conditions, where a fixed-mode transceiver's quality would become severely degraded by channel effects. Our AQAM/JD-CDMA scheme achieved the best compromise between unimpaired error-free speech quality and robustness, which has been verified by our informal listening tests.

Our future research will be focused on improving the performance of BbB-AQAM/CDMA transceivers using wideband speech and audio codecs operated in multiple modes. Furthermore, more robust, turbo space-time coded multicarrier, frequency-hopped BbB-AQAM CDMA transceivers will be invoked.

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Multiuser Detection Assisted Time- and Frequency-Domain Spread Multicarrier Code-Division Multiple-Access

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Abstract—In this contribution, we study a reduced-complexity multiuser detection aided multicarrier direct-sequence code-division multiple-access (MC DS-CDMA) scheme, which employs both time (T)-domain and frequency (F)-domain spreading. We investigate the achievable detection performance in the context of synchronous TF-domain spread MC DS-CDMA when communicating over an additive white Gaussian noise (AWGN) channel. Five detection schemes are investigated, which include the single-user correlation based detector, the joint TF-domain decorrelating multiuser detector (MUD), the joint TF-domain MMSE MUD, the separate TF-domain decorrelating/MMSE MUD, and the separate TF-domain MMSE/decorrelating MUD. Our simulation results show that the separate TF-domain MUD schemes are capable of achieving a similar bit error rate (BER) performance to that of the significantly more complex joint TF-domain MUD schemes.

Index Terms—Code-division multiple-access (CDMA), decorrelating, frequency-domain spreading, joint detection, minimum mean square error (MMSE), multicarrier (MC), multiuser detection, separate detection, time-domain spreading.

I. INTRODUCTION

In the context of direct-sequence (DS) code-division multiple-access (DS-CDMA) communications, there are two types of spread-spectrum schemes. The first of these spread-spectrum schemes [1]–[3] spreads the original data stream using a signature code in the time (T)-domain, and the spread-spectrum signal is transmitted using a single-carrier. In contrast, the second DS spread-spectrum scheme [4]–[7] spreads the original data stream to a number of subcarriers using a signature code in the frequency (F)-domain, and each chip of the resultant spread-spectrum signal is transmitted by a different carrier. Hence, this scheme is also referred to as multicarrier CDMA (MC-CDMA) in the literature [5], [6], [8], [9]. Furthermore, there is a family of multicarrier CDMA in which each subcarrier signal constitutes a T-domain DS spread signal, but no F-domain spreading is employed. This family of multicarrier CDMA is usually referred to as MC DS-CDMA [5], [10]–[16]. An amalgam of these spread-spectrum schemes was proposed in [17]. This extended spread-spectrum scheme spreads the transmitted data stream using two signature codes, where one of the signature codes corresponds to the T-domain spreading, while the other corresponds to the F-domain spreading. Since the proposed multicarrier DS-CDMA scheme employs both the previously mentioned T-domain spreading and F-domain spreading, it is referred to as TF-domain spread MC DS-CDMA.

The benefits of employing both T-domain spreading and F-domain spreading in MC DS-CDMA systems are multifold. First, the future generations of broadband multiple-access systems [18] are expected to have a bandwidth on the order of tens or even hundreds of MHz. When single-carrier based DS-CDMA or MC-CDMA using solely T-domain spreading or solely F-domain spreading is utilized, the total system bandwidth is related to either the T-domain spreading factor or to the

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