Channel-Adaptive WIDEBAND Wireless Video Telephony

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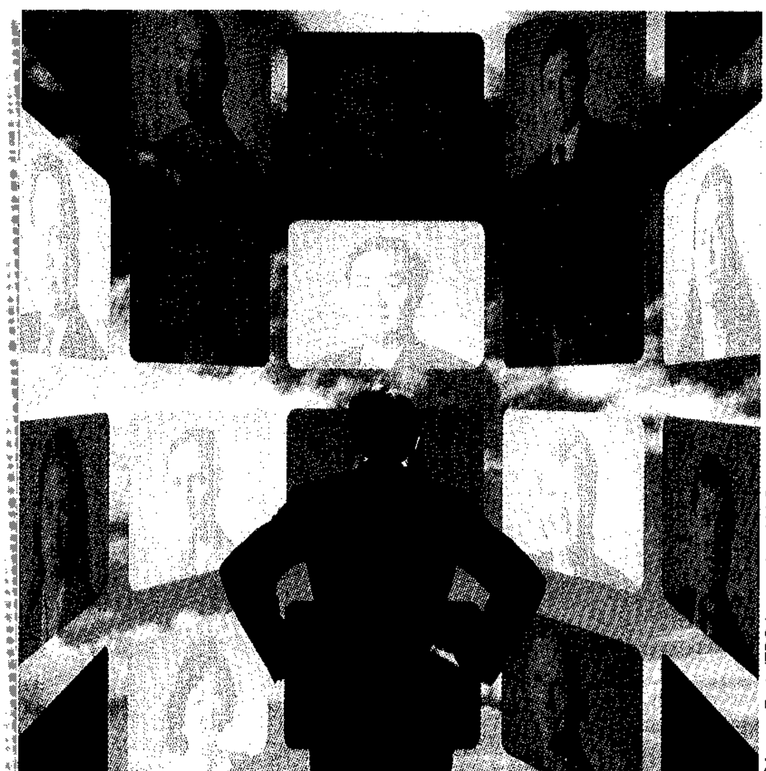

Design Tradeoffs and Performance Comparisons

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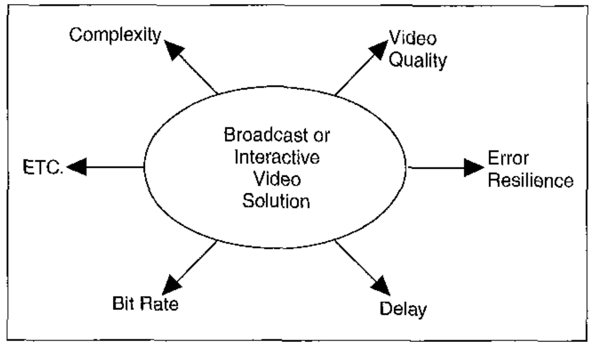
In recent years the concept of intelligent multimode, multimedia transceivers (IMMTs) has emerged in the context of wireless systems [1], and the range of various existing solutions that have found favor in existing standard systems was summarized in the excellent overview by Nanda et al. [2]. The aim of these adaptive transceivers is to provide mobile users with the best possible compromise among a number of contradicting design factors, such as the power consumption of the hand-held portable station (PS), robustness against transmission errors, spectral efficiency, teletraffic capacity, audio/video quality, and so forth [1]. We will limit our discourse to a small subset of the associated wireless video transceiver design issues, referring the reader for a deeper exposure to [3]-[13]. A further advantage of the IMMTs of the near future is that due to their flexibility they are likely to be able to reconfigure themselves in various operational modes to ensure backwards compatibility with existing, so-called second generation (2G) standard wireless systems, such as the Japanese digital cellular [14], the Pan-American IS-54 [15] and IS-95 [16] systems, as well as the global system of mobile communications (GSM) [17] standards. The basic features of these systems are outlined in Table 1.

The fundamental advantage of burst-by-burst (BbB) adaptive IMMTs is that—irrespective of the propagation environment encountered—when the mobile roams across different environments subject to pathloss; shadow- and fast-fading; co-channel-, intersymbol-, and multiuser interference, while experiencing power control errors, the system will always be able to configure itself in the highest possible throughput mode, while maintaining the required transmission integrity. Furthermore,



▲ *1. Contradictory system design requirements of various video communications systems.*

while powering up under degrading channel conditions may put other users in the system at a disadvantage, invoking a more robust—although lower throughput—transmission mode will not. The employment of the above BbB adaptive modems in the context of code division multiple access (CDMA) is fairly natural and it is motivated by the fact that all three third-generation (3G) mobile radio system proposals employ CDMA [18]-[20].

In simple terms, finding a specific solution to a distributive or interactive video communications problem has to be based on a compromise in terms of the inherently contradictory constraints of video quality, bit rate, delay, robustness against channel errors, and the associated implementational complexity, as suggested by Fig. 1. Considering some of these tradeoffs and proposing a range of attractive solutions to various video communications problems is the basic aim of this overview.

The article is structured as follows. The first section portrays a range of proprietary video codecs and com-

Table 1. Speech Coding Rates of Second-Generation Mobile Systems.

	Cellular Systems					Cordless System			
System	GSM [17]	DCS-1800	IS-95A/B [16], [18] CDMA	IS-54/IS-136 [15], [21], [22]	JDC [14]	CT2 [23]	DECT [24]	PHS [25]	PACS [26]
	Europe 22.8 13	Euorpe 22.8 13	USA 8/Var 1.2-9.6	USA 11.2 7.95/7.4	Japan 13 6.7	USA 32 32	Europe 32 32	Japan 32 32	USA 32 32
Speech FEC	Conv. (2, 1, 5)	Conv. (2, 1, 5)	Conv. UL: (2, 1, 9) DL: R=1/3	Conv. (2, 1, 5)	Conv. R=9/17	No	No	CRC	CRC

DCS-1800: GSM-like European system in the 1800 MHz band
IS-95: American CDMA system
PHS: Japanese personal handyphone system
JDC: Japanese digital cellular
CDMA: Code Division Multiple Access
GSM: Global system of mobile communications
Conv.: Convolutional coding with a rate of R

IS-54: American digital advanced mobile phone system (DAMPS)
CT2: British cordless telephone system
PACS: Personal access communications system
GSM: Global system of mobile communications
UL: Uplink, DL: Downlink

In our proposed systems the wideband channel-induced degradation is combated not only by the employment of adaptive modulation but also by equalization.

pares them to some of the existing standard video codecs. A number of multimode video transceivers are also characterized. In contrast, the next section is dedicated to systems employing the standard H.263 video codec in the context of wideband BbB adaptive video transceivers. The concept of BbB-adaptive video transceivers is then extended to CDMA-based systems, before we present our conclusions.

Proprietary Wireless Videophone Systems

System Constraints

To provide wireless videophony services in the context of proprietary or 2G wireless systems, an additional speech channel has to be allocated for the transmission of the video information. Here we briefly review some of the associated fixed but programmable-rate video coding aspects with reference to some of the recent literature on the topic [3]-[13]. We note, however, that a substantial amount of work has to be carried out in the area of intelligent transducers and interfaces, such as appropriate miniaturized video cameras, low power-consumption video displays, and in particular in the field of ergonomic hand-held portable multimedia communicator construction and design.

The speech channel rates of the currently operational 2G wireless systems are 6.7, 8, 9.6, and 13 kb/s, respectively, as seen in Table 1. For these constant, but rather low, video rates there are two possible types of video codecs, which can be realistically invoked, namely, fixed-rate proprietary schemes [4]-[7], [13] or the standard H.263 video codec in conjunction with an appropriate rate control scheme, using also adaptive packetization and a strongly protected transmission packet acknowledgment flag [9]-[13]. In this section we concentrate on fixed-rate proprietary video codecs and transceivers [4]-[7], [13]. Later we will consider a range of video transmission schemes based on the standard H.263 video codec [9]-[13].

In our former work we designed a suite of fixed-rate, proprietary video codecs [4]-[7] that are capable of operating at a video scanning or refreshment rate of 10 frames/s (f/s). These codecs were capable of maintaining sufficiently low bit rates for the provision of videophony over an additional speech channel in the context of the operational 2G wireless systems of Table 1, provided that low-dynamic head-and-shoulder video sequences of the 176×144-pixel so-called quarter common intermediate format (QCIF) or 128×96-pixel sub-QCIF video resolution are employed. We note, however, that for high-dynamic sequences the 32kb/s typical speech bit rate of the cordless telephone systems of Table 1, such as the Japanese personal handyphone system (PHS), the digital European cordless telephone (DECT), or the British CT2 system, is more adequate in terms of video quality. Furthermore, the proposed programmable video codecs are capable of multirate operation in the 3G universal mobile telecommunications system (UMTS) [20] or in the so-called IMT2000 and cdma2000 [20] systems.

In this treatise we cannot consider the performance of the proposed video codecs [7], [13] in all of the above 2G and 3G mobile radio systems, hence we will mainly concentrate on a range of intelligent multimode schemes based on quadrature amplitude modulation (QAM) [30]. These multimode schemes will allow us to assess the various design tradeoffs in terms of throughput, complexity, channel quality requirements, etc. The main goal initially is to describe the design philosophy of our fixed-rate 8 or 11.36 kb/s prototype video codecs and document their performance using two characteristic fixed bit rates, namely, 8 and 11.36 kb/s within the above typical speech coding rate range, namely, around 10 kb/s, as discussed in the context of the systems portrayed in Table 1. We will show that these 8 and 11.36 kb/s codecs exhibit similar video quality, but different error resiliences. Furthermore, we will devote some attention to transmission robustness issues and briefly summarize the features of a range of appropriate transceivers. Speech source coding aspects are beyond the scope of this contribution; the reader is referred to, for example, [27] and [28], for the choice of the appropriate speech codecs. Channel coding issues are addressed in [29], while a detailed discussion of reconfigurable modulation is given for example in [30, Chap. 13].

Fixed-Rate Video Compression

The topic of video compression is well documented [31]-[42], and some of the associated design tradeoffs were highlighted previously. In contrast to the existing and forthcoming standard variable-rate video compression schemes, such as the H.261, H.263 [33] and MPEG-1 [31], MPEG-2 [32], [35], and MPEG-4 [41] codecs, which rely on bandwidth-efficient but error-sensitive variable-length coding (VLC) techniques [35] combined with a complex self-descriptive bit stream structure, our proposed fixed-rate codecs exhibit a more robust bit stream and a constant bit rate. For the sake of improved robustness it is often advantageous to refrain from using VLC. Hence here we attempt to offer an overview of a suite of constant, but arbitrarily programmable-rate 176×144 pixel head-and-shoulder QCIF video codecs specially designed for videotelephony over existing and future mobile radio speech systems on the basis of a recent research program [4]-[13].

All the codecs considered here share the schematic of Fig. 2, although in the figure vector-quantized (VQ) motion compensated error residual (MCER) coding is shown, as an example. Discrete cosine transform (DCT) and quad-tree (QT) based MCER compression were the topic of [4], [13] and [6], [13], respectively. Since the codecs discussed in this section were designed for operation at 10 f/s and for bit rates of 800 or 1136 kb/s, the number of bits per frame becomes 800 or 1136, respectively. Let us now consider the video codec schematic of Fig. 2 in a little more detail, initially concentrating on the intraframe coding mode, which is necessary in all video codecs employing additionally also interframe compression techniques for the sake of high coding efficiency.

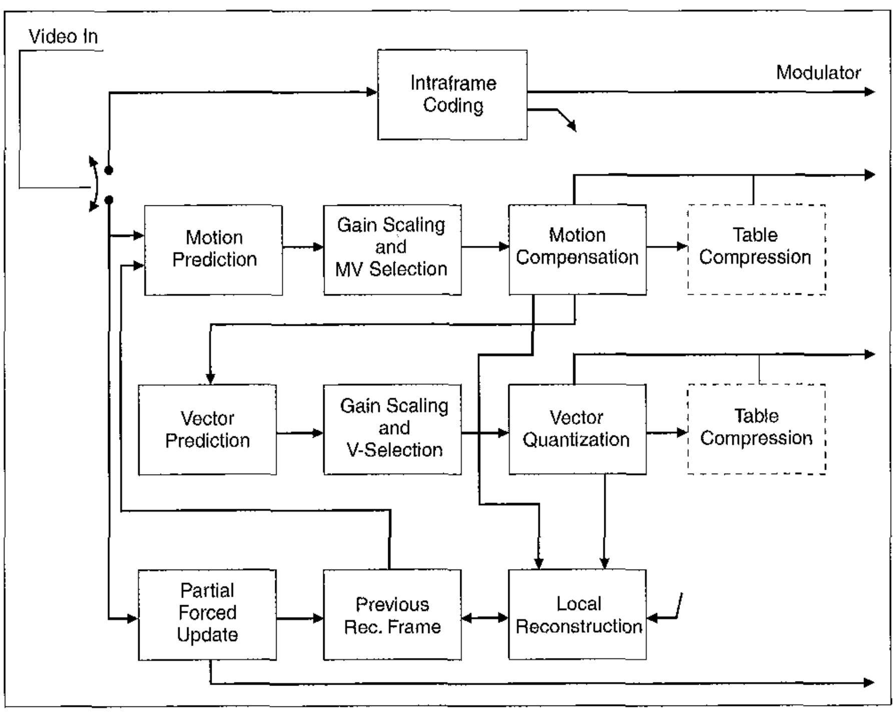
Intraframe Mode

Since the first frame used in motion compensation is unknown to the decoder, the 4-bit encoded block averages are transmitted first in the intraframe mode, which are also used by the local decoder. Due to the paucity of bits per video frame—namely, 800 or 1136—we can only afford to transmit a rather low-resolution intracoded initial frame. For example, in the 800-bit/frame codec a maximum of $800/4=200$ such intraframe coded blocks can be used, which requires a block-size of about 144×176 pixels$/200 \approx 11 \times 11$ pixels. This results in a rather coarse intraframe coded initial frame, which is used by the interframe coded mode of operation to improve the video quality in successive coding steps. This initial "warm-up" phase of the codec typically is imperceptible to the untrained viewer, since it typically takes less than 1-2 s and it is a consequence of the fixed bit-rate constraint imposed for example by 2G mobile radio systems.

Furthermore, the intraframe mode is also invoked during later coding stages in a number of blocks to mitigate the effect of transmission errors and hence to prevent encoder/decoder misalignment, as it will be detailed later. In this context this operation is often referred to as partial forced updating (PFU), since the specific blocks concerned are partially updated by superimposing an attenuated version of the intraframe coded block averages. The number of the PFU blocks per frame is bit-rate dependent, and it is automatically determined by our programmable codec. For example, in our 11.36 kb/s prototype codec in every frame 22 out of the 396 8×8-pixel blocks, scattered over the entire frame, are periodically updated using the 4-bit quantized block means, which are partially overlayed on to the contents of the reconstructed frame buffer. Specifically, the overlaying is performed such that the block's contents in the local buffer is weighted by 0.7 and superimposed on to the received block average, which is scaled by 0.3. The bit-rate contribution of this PFU process is a moderate $22 \times 4=88$ bits per QCIF frame, and it refreshes about 5.6 of each QCIF video frame.

Interframe Mode

We note, however that we employ motion compensation (MC) for a given 8×8 pixel block only, if it is deemed sufficiently beneficial in terms of MCER energy reduction. For most of the blocks the MCER energy reduction does not justify the additional transmission overhead associated with the motion vector (MV), a principle which we refer to as cost-gain motivated quantization [4], [13]. These operations are represented by the processing blocks in the second row of Fig. 2. In fact, our results suggested that using the 38 most important active MVs out of the 396 blocks' MVs was sufficient to achieve a near-maximum MCER-reduction. As seen in Table 2, for our 8 kb/s and 11.36 kb/s candidate codecs each of the 38 motion-active blocks' MVs were coded using 4 bits, while their position by 9 bits. The 9-bit active block address was necessitated by having to differentiate among 396 8×8-pixel blocks. For the remaining blocks simple frame differencing was invoked, which does not require MVs.



▲ *2. Basic schematic of the fixed-rate VQ-based video codec.*

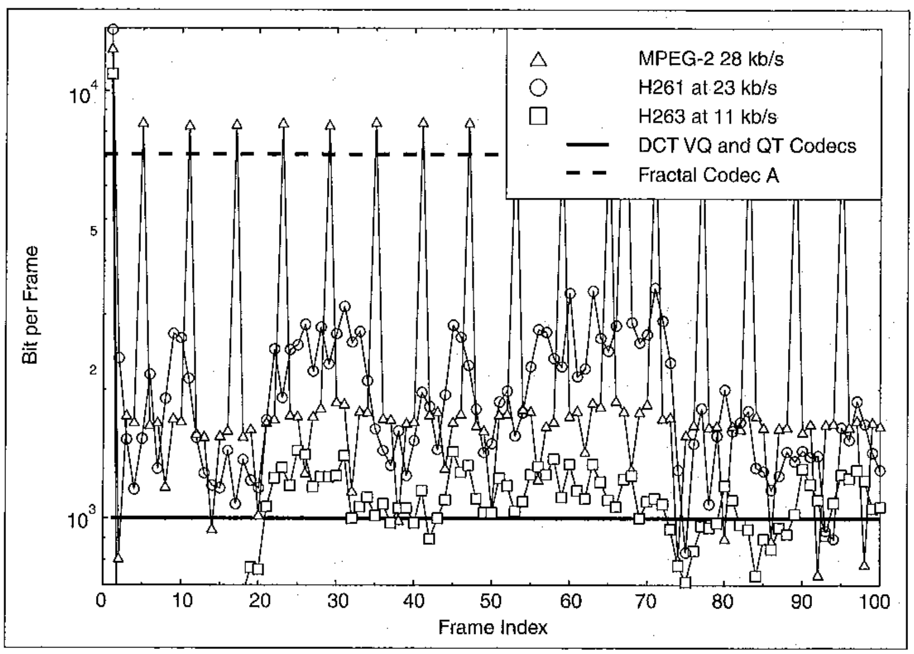
Table 2. Bit-Allocation of Fixed- but Programmable-Rate Codecs.								
Codec	FAW	PFU	MV Index + MV	DCT Ind. + DCT	VQ Ind. + VQ	QT + PC	Padding	Total
DCTC2	22	22 × 4	30 × 9 + 30 × 4	30 × 9+ 30 × 8	—	—	6	1136
DCTC1	22	22 × 4	<350 (VLC)	<350 (VLC)	—	—	VLC	800
VQC2	22	22 × 4	38 × 9 + 38 × 4	—	30 × 9+ 30 × 8	—	5	1136
VQC1	22	22 × 4	<350 (VLC)	—	<350 (VLC)	—	VLC	800
QTC	22	20 × 4	<500 (VLC)	—	—	565 + 1 or 80	VLC	1136

The MCER coding invokes a second-stage cost- gain quantization process, where again a fraction of the 396 MCER blocks are coded only. These processing steps are constituted by the blocks in the third row of Fig. 2. The remaining MCER blocks are set to zero. To identify the most important MCER blocks, the so-called classified VQ process of [5] and [13] can be used for the sake of maintaining a low implementation complexity. On this basis in the codecs considered the 31 most important MCER-active blocks were found for each QCIF frame and were represented with the aid of a 256-entry trained codebook. The codebook training procedure was also outlined in [5] and [13]. The above coding steps finally resulted in the bit allocation scheme seen in the third line of Table 2.

Observe, however, in Fig. 2 that the associated motion-activity tables and the associated MVs, as well as the MCER activity tables and VQ addresses, can be further compressed in the blocks drawn in dashed line upon invoking efficient, but error-sensitive VLC techniques. Hence, apart from the above 11.36 kb/s VQ-based codec, which we refer to as VQ2, we also contrived an 8 kb/s version of this scheme, VQ1. Explicitly, note in the fourth line of Table 2 that by VLC both the motion-activity and the MCER activity tables, as well as the MVs and the VQ codebook addresses, can be further compressed to around 350 bits per QCIF video frame. Lastly, following a similar philosophy, we also designed a VQ-based [5], [13] and a QT-based [6], [13] codec, which are again, characterized in lines 1, 2, and 5 of Table 2, respectively. The associated transmission issues were reported along with the video compression issues in [4]-[6] and [13].

The various performance aspects of these video codecs were compared in [7], which are summarized in Figs. 3, 4, and 5, suggesting the following conclusions.

- The proposed codecs exhibit a similar peak signal-to-noise ratio (PSNR) performance at 10 kb/s to the MPEG-2 codec operated at an average bit rate of 28 kb/s. The H-261 codec at 22 kb/s has a PSNR of about 2 dB higher than the 10 kb/s fixed-rate codecs.
- The inherent latency of the H-261 codec and that of the 10 kb/s codecs is one frame or 100 ms due to MC. The latency of the H.263 and MPEG-2 codecs may become significantly higher due to using so-called predicted or P-frames, relying on a number of previously received frames.
- The robustness of the lower-rate 8 kb/s codecs, which further compress the active/passive tables is quite limited,



▲ 3. *Bit-rate fluctuation versus frame index for the proposed adaptive codecs and the MPEG-2, H.261, and H.263 standard codecs.*

as is that of the VLC-based standard codecs. The slightly less bandwidth efficient schemes refraining from using the optional table compression drawn in dashed line in Fig. 2 exhibit an improved error resilience.

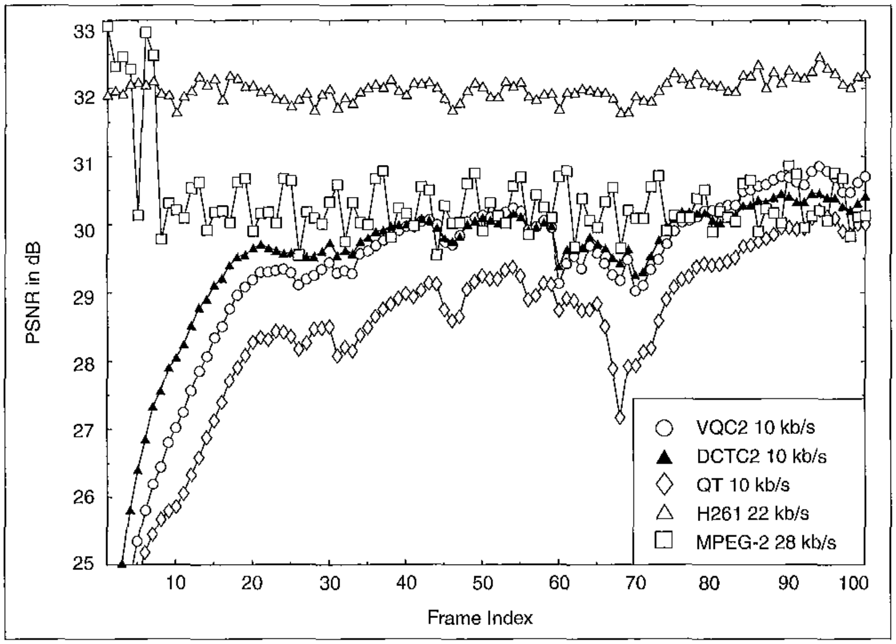
▲ Overall, the vector quantized codecs VQC1 and VQC2 were shown to constitute the best compromise in terms of video quality, compression ratio, and computational complexity, closely followed by the DCTC1 and DCTC2 candidate codecs [7]. The QT codec does not lend itself to robust implementations, since the variable-length QT code is rather vulnerable to channel errors, as it is evidenced by Fig. 5.

Transmission Aspects

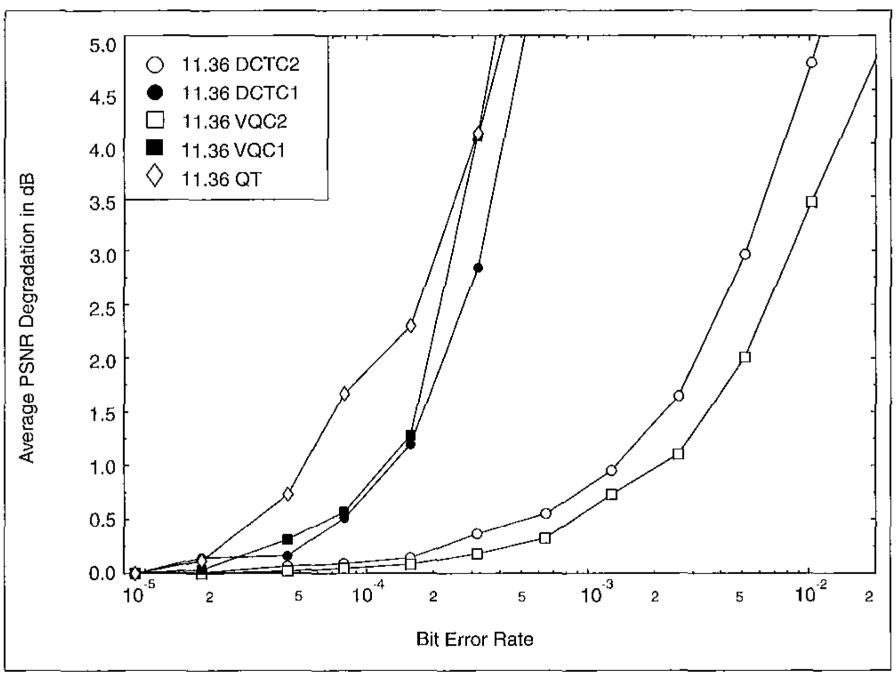
On the basis of our preference concerning the VQ codecs, we then contrived a suite of source-sensitivity matched programmable videophone transceivers, which were detailed in [5] and [13]. The features of these systems are summarized in Table 3, although here due to lack of space we can only briefly highlight these features, again, referring the interested readers to [5] and [13].

Specifically, six different systems were designed, which used twin-class binary Bose-Chaudhuri-Hocquenghem (BCH) error correction coding [20], protecting the more vulnerable Class 1 (C1) video bits more strongly than the less important C2 bits, whenever possible. This is not always explicit in Table 3, as seen for example for System 1, since we exploited that 16-level quadrature amplitude modulation (16 QAM) schemes possess two different integrity internal subchannels [30]. When combined with various BCH codecs [29], these 16 QAM subchannels can be invoked for the high-integrity delivery of the more vulnerable video bits. Pilot symbol assisted QAM (PSAQAM) was used [30]. Adaptive QAM schemes, which can reconfigure themselves on a BbB basis and, hence, are amenable to IMMT implementations, were considered for example in [30] in the context of both single-carrier QAM and multicarrier QAM.

All systems have a BCH-coded [29] rate of 20.32 kb/s, implying that the 8 kb/s video codec modes were more heavily BCH coded, to resolve, as to whether it is worthwhile reducing the video codec's rate by VLC, to be able to incorporate a stronger BCH code. In some of the scenarios



▲ *4. PSNR versus frame index performance of the proposed adaptive codecs and the MPEG-2, H.261, and H.263 standard codecs.*



▲ *5. PSNR degradation versus BER for the proposed fixed-rate video codecs.*

automatic repeat request (ARQ) was invoked. Specifically, ARQ acted upon either all errors, such as the C1/C2 video bit errors, or only in case of errors in the more sensitive C1 video bit stream. In this context we have to note that the packets, which are retransmitted, would occupy additional transmission slots and hence reduce the system's teletraffic capacity. This is also shown in Table 3. The corresponding user baud rates are 18 or 9 KBd, depending on whether 4 QAM or 16 QAM was invoked by the transceiver. At a multiuser signaling rate of 144 KBd, which fits in the 200 KHz channel bandwidth, 8 or 16 videophone users could be accommodated without ARQ. When occupying two slots per frame for ARQs, assuming that only one of the users can rely at any moment on a maximum of two retransmission attempts, the number of users supported by Systems 2-6 is hence reduced by two. The effective user bandwidth was computed by dividing the 200 KHz system bandwidth by the number of users supported. The minimum required channel signal-to-noise ratio (SNR) values over both additive white Gaussian noise (AWGN) and Rayleigh channels were recorded in Table 3 for second-order switch-diversity reception using the appropriate pseudo-SNR versus channel SNR curves of [5] along with the maximum number of videophone users supported within the 200 KHz bandwidth used. A range of interesting conclusions can be drawn from Table 3, which were set out more explicitly in [5]. Having reviewed a range of fixed- but programmable-rate video codecs and statically reconfigurable video transceivers, let us now consider the family of more agile BbB reconfigurable video systems and their performance benefits in the next section.

Table 3. Summary of GSM-Like Video System Features.

Feature	System 1	System 2	System 3	System 4	System 5	System 6
Video Codec	VQC1	VQC2	VQC1	VQC2	VQC2	VQC1
Video Rate (kb/s)	11.36	8	11.36	8	8	11.36
Frame Rate (f/s)	10	10	10	10	10	10
C1 FEC	BCH(127,71,9)	BCH(127,50,13)	BCH(127,71,9)	BCH(127,50,13)	BCH(127,50,13)	BCH(127,71,9)
C2 FEC	BCH(127,71,9)	BCH(127,92,5)	BCH(127,71,9)	BCH(127,50,13)	BCH(127,50,13)	BCH(127,71,9)
Header FEC	BCH(127,50,13)	BCH(127,50,13)	BCH(127,50,13)	BCH(127,50,13)	BCH(127,50,13)	BCH(127,50,13)
FEC-Coded Rate (kb/s)	20.32	20.32	20.32	20.32	20.32	20.32
Modem	4/16-PSAQAM	4/16-PSAQAM	4/16-PSAQAM	4/16-PSAQAM	4/16-PSAQAM	4/16-PSAQAM
ARQ	None	Cl. One	Cl. One & Two	Cl. One & Two	None	Cl. One
User Signal Rate (kBd)	18 or 9	9	18 or 9	18 or 9	18 or 9	9
System Signal Rate (kBd)	144	144	144	144	144	144
System Bandwidth (kHz)	200	200	200	200	200	200
No. of Users	8-16	(16–2)=14	6-14	6-14	8-16	(16–2)=14
Eff. User Bandwidth (kHz)	25 or 12.5	14.3	33.3 or 14.3	33.3 or 14.3	33.3 or 14.3	14.3
Min. AWGN SNR (dB) 4/16 QAM	5/11	11	4.5/10.5	6/11	8/12	12
Min. Rayleigh SNR (dB) 4/16 QAM	10/22	15	9/18	9/17	13/19	17

BbB Adaptive QAM Video Transceivers

Narrowband BbB Adaptive Modulation

In BbB adaptive quadrature amplitude modulation (BbB-AQAM) a high-order, high-throughput modulation mode is invoked, when the instantaneous channel quality is favorable [30]. By contrast, a more robust lower order BbB-AQAM mode is employed, when the channel exhibits inferior quality, for improving the average bit error rate (BER) performance. To support the operation of the BbB-AQAM modem, a high-integrity, low-delay feedback path has to be invoked between the transmitter and receiver for signaling the estimated channel quality perceived by the receiver to the remote transmitter. This strongly protected message, for example, can be superimposed on the reverse-direction messages of a duplex interactive channel. The transmitter then adjusts its AQAM mode according to the instructions of the receiver, to be able to meet its BER target.

The salient feature of the proposed BbB-AQAM technique is that irrespective of the channel conditions, the transceiver achieves always the best possible multimedia source-signal representation quality—such as video, speech, or audio quality—by automatically adjusting the achievable bit rate and the associated multimedia source-signal representation quality to match the channel quality experienced. The AQAM modes are adjusted on a near-instantaneous basis under given propagation conditions to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference (CCI), multiuser interference, etc. Furthermore, when the mobile is roaming in a hostile outdoor—or even hilly terrain—propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high-source signal representation quality modes are employed.

BbB-AQAM has been originally suggested by Webb and Steele [43], stimulating further research in the wireless community, for example by Sampei et al. [44], showing promising advantages when compared to fixed modulation in terms of spectral efficiency, BER performance, and robustness against channel delay spread. Various systems employing AQAM were also characterized in [30]. The numerical upper bound performance of narrowband BbB-AQAM over slow Rayleigh flat-fading channels was evaluated by Torrance et al. [53], while over wideband channels by Wong et al. [46], wide-AQAM. Following these developments, the optimization of the BbB-AQAM switching thresholds was carried employing Powell-optimization using a cost-function, which was based on the combination of the target BER and target bit per symbol (b/symbol) performance [48]. Adaptive modulation was also studied in conjunction with channel coding and power control techniques by Matsuoka et al. [49] as well as Goldsmith and Chua [50].

In the early phase of research more emphasis was dedicated to the system aspects of adaptive modulation in a narrowband environment. A reliable method of transmitting the modulation control parameters was proposed by Otsuki et al. [52], where the parameters were embedded in the transmission frame's midamble using Walsh codes. Subsequently, at the receiver the Walsh sequences were decoded using maximum likelihood detection. Another technique of estimating the required modulation mode used was proposed by Torrance et al. [53], where the modulation control symbols were represented by unequal error protection five-phase-shift-keying (PSK) symbols. The adaptive modulation philosophy was then extended to wideband multipath environments by Kamio et al. [54] by utilizing a bidirectional decision feedback equalizer (DFE) in a micro- and macro-cellular environment. This equalization technique employed both forward- and backward-oriented channel estimation based on the pre-amble and post-amble symbols in the transmitted frame. Equalizer tap gain interpolation across the transmitted frame was also utilized to reduce the complexity in conjunction with space diversity [54]. The authors concluded that the cell radius could be enlarged in a macro-cellular system and a higher area-spectral efficiency could be attained for micro-cellular environments by utilizing adaptive modulation. The latency effect, which occurred when the input data rate was higher than the instantaneous transmission throughput was studied and solutions were formulated using frequency hopping [55] and statistical multiplexing, where the number of slots allocated to a user was adaptively controlled [56].

In [57], symbol rate adaptive modulation was applied, where the symbol rate or the number of modulation levels was adapted by using 1/8-rate 16 QAM, 1/4 rate 16 QAM, and 1/2-rate 16 QAM, as well as full-rate 16 QAM, and the criterion used to adapt the modem modes was based on the instantaneous received SNR and channel delay spread. The slowly varying channel quality of the uplink (UL) and downlink (DL) was rendered similar by utilizing short frame duration time division duplex (TDD), and the maximum normalized delay spread simulated was 0.1. A variable channel coding rate was then introduced by Matsuoka et al. in conjunction with adaptive modulation in [49], where the transmitted burst incorporated an outer Reed-Solomon code and an inner convolutional code to achieve high-quality data transmission. The coding rate was varied according to the prevalent channel quality using the same method as in adaptive modulation to achieve a certain target BER performance. A so-called channel margin was introduced in this contribution, which adjusted the switching thresholds to incorporate the effects of channel quality estimation errors. As mentioned above, the performance of channel coding in conjunction with adaptive modulation in a narrowband environment was also characterized by Chua and Goldsmith [50]. They used trellis and lattice codes without channel interleaving, invoking a feedback path between the transmitter and receiver for modem mode control purposes. The effects of the delay in the feedback path on the adaptive modem's performance were studied, and this scheme exhibited a higher spectral efficiency when compared to the nonadaptive trellis coded performance.

Subsequent contributions by Suzuki et al. [58] incorporated space diversity and power adaptation in conjunction with adaptive modulation, for example, to combat the effects of the multipath channel environment at a 10 Mb/s transmission rate. The maximum tolerable delay spread was deemed to be one symbol duration for a target mean BER performance of 0.1. This was achieved in a time division multiple access (TDMA) scenario, where the channel estimates were predicted based on the extrapolation of previous channel quality estimates. Variable transmitted power was then applied in combination with adaptive modulation in [51], where the transmission rate and power adaptation was optimized to achieve an increased spectral efficiency. In this treatise, a slowly varying channel was assumed, and the instantaneous received power required to achieve a certain upper bound performance was assumed to be known prior to transmission. Power control in conjunction with a predistortion type nonlinear power amplifier compensator was studied in the context of adaptive modulation in [59]. This method was used to mitigate the nonlinearity effects associated with the power amplifier when QAM modulators were used.

Results were also recorded concerning the performance of adaptive modulation in conjunction with different multiple access schemes in a narrowband channel environment. In a TDMA system, dynamic channel assignment was employed by Ikeda et al., where in addition to assigning a different modulation mode to a different channel quality, priority was always given to those users in reserving time slots which benefit from the best channel quality [60]. The performance was compared to fixed channel assignment systems, where substantial gains were achieved in terms of system capacity. Furthermore, a lower call termination probability was recorded. However, the probability of intracell handoff increased as a result of the associated dynamic channel assignment (DCA) scheme, which constantly searched for a high-quality, high-throughput time slot for the existing active users. The application of adaptive modulation in packet transmission was introduced by Ue et al. [61], where the results showed improved data throughput. Recently, the performance of adaptive modulation was characterized in conjunction with an ARQ system in reference [62], where the transmitted bits were encoded using a cyclic redundant code (CRC) and a convolutional punctured code to increase the data throughput.

A recent treatise was published by Sampei et al. [63] on laboratory test results concerning the utilization of adaptive modulation in a TDD scenario, where the modem mode switching criterion was based on the SNR and on the normalized delay spread. In these experimental results, the channel quality estimation errors degraded the performance, and consequently a channel estimation error margin was devised to mitigate this degradation. Explicitly, the channel estimation error margin was defined as the measure of how much extra protection margin must be added to the switching threshold levels to minimize the effects of the channel estimation errors. The delay spread also degraded the performance due to the associated irreducible BER, which was not compensated by the receiver. The performance of the adaptive scheme in a delay-spread impaired channel environment was better, however, than that of a fixed modulation scheme. Lastly, the experiment also concluded that the AQAM scheme can be operated for a Doppler frequency of $f_d = 10$ Hz with a normalized delay spread of 0.1 or for $f_d = 14$ Hz with a normalized delay spread of 0.02, which produced a mean BER of 0.1 at a transmission rate of 1 Mb/s.

Lastly, the latency and interference aspects of AQAM modems were investigated in [71] and [72]. Specifically, the latency associated with storing the information to be transmitted during severely degraded channel conditions was mitigated by frequency hopping or statistical multiplexing. As expected, the latency is increased when either the mobile speed or the channel SNR are reduced, since both of these result in prolonged low instantaneous SNR intervals. It was demonstrated that as a result of the proposed measures, typically more than 4 dB SNR reduction was achieved by the proposed adaptive modems in comparison to the conventional fixed-mode benchmark modems employed. The achievable gains, however, depend strongly on the prevalent co-channel interference levels and hence interference cancellation was invoked in [72] on the basis of adjusting the demodulation decision boundaries after estimating the interfering channel's magnitude and phase.

Having reviewed the developments in the field of narrowband AQAM, we now consider wideband AQAM modems in the next section.

Wideband BbB Adaptive Modulation

In the above narrowband channel environment, the quality of the channel was determined by the short-term SNR of the received burst, which was then used as a criterion to choose the appropriate modulation mode for the transmitter, based on a list of switching threshold levels l_n [43]-[45]. In a wideband environment, however, this criterion is not an accurate measure for judging the quality of the channel, where the existence of multipath components produces not only power attenuation of the transmission burst, but also intersymbol interference. Consequently, appropriate channel quality criteria have to be defined to estimate the wideband channel quality for invoking the most appropriate modulation mode.

The most reliable channel quality estimate is the BER, since it reflects the channel quality, irrespective of the source or the nature of the quality degradation. The BER can be estimated with a certain granularity or accuracy, provided that the system entails a channel decoder or—synonymously—forward error correction (FEC) decoder employing algebraic decoding [20]. If the system contains a so-called soft-in-soft-out (SISO) channel decoder, such as a turbo decoder [73], the BER can be estimated with the aid of the logarithmic likelihood ratio (LLR), evaluated either at the input or the output of the channel decoder. Hence a particularly attractive way of invoking LLRs is employing powerful turbo codecs, which provide a reliable indication of the confidence associated with a particular bit decision. The LLR is defined as the logarithm of the ratio of the probabilities associated with a specific bit being binary zero or one. Again, this measure can be evaluated at both the input and the output of the turbo channel codecs and both of them can be used for channel quality estimation.

In the event that no channel encoder/decoder (codec) is used in the system, the channel quality expressed in terms of the BER can be estimated with the aid of the mean-squared error (MSE) at the output of the channel equalizer or the closely related metric, the pseudo-SNR [46]. The MSE or pseudo-SNR at the output of the channel equalizer has the important advantage that it is capable of quantifying the severity of the intersymbol interference (ISI) and/or CCI experienced, in other words quantifying the signal-to-interference plus noise ratio (SINR).

In our proposed systems the wideband channel-induced degradation is combated not only by the employment of adaptive modulation but also by equalization. In following this line of thought, we can formulate a two-step methodology in mitigating the effects of the dispersive wideband channel. In the first step, the equalization process will eliminate most of the intersymbol interference based on a channel impulse response (CIR) estimate derived using the channel sounding midamble and consequently, the signal to noise and residual interference ratio at the output of the equalizer is calculated.

We found that the residual channel-induced ISI at the output of the DFE is near-Gaussian distributed and that if there are no decision feedback errors, the pseudo-SNR at the output of the DFE, γ_{dfe} can be calculated as [64], [46], [47]:

$$\gamma_{dfe} = \frac{\text{Wanted Signal Power}}{\text{Residual ISI Power} + \text{Effective Noise Power}} = \frac{E\left[\left|S_k \sum_{m=0}^{N_f-1} C_m h_m\right|^2\right]}{\sum_{q=-(N_f-1)}^{-1} E\left[\left|f_q S_{k-q}\right|^2\right] + N_o \sum_{m=0}^{N_f-1} |C_m|^2}, \tag{1}$$

where C_m and h_m denote the DFE's feed-forward coefficients and the channel impulse response, respectively. The transmitted signal and the noise spectral density is represented by S_k and N_o. Lastly, the number of DFE feed-forward coefficients is denoted by N_f. By utilizing the pseudo-SNR at the output of the equalizer, we are ensuring that the system performance is optimized by employing equalization and AQAM [30] in a wideband environment according to the following switching regime:

$$\text{Modulation Mode} = \begin{cases} \text{No TX}, & \text{if } \gamma_{DFE} < f_0 \\ \text{BPSK}, & \text{if } f_0 < \gamma_{DFE} < f_1 \\ \text{4 QAM}, & \text{if } f_1 < \gamma_{DFE} < f_2 \\ \text{16 QAM}, & \text{if } f_2 < \gamma_{DFE} < f_3 \\ \text{64 QAM}, & \text{if } \gamma_{DFE} > f_3 \end{cases} \tag{2}$$

where f_n, $n = 0,\dots,3$ are the pseudo-SNR thresholds levels, which are set according to the system's integrity requirements and the modem modes may assume 0,...,6 b/symbol transmissions corresponding to no transmissions (No TX), binary PSK (BPSK), as well as 4-, 16-, and 64 QAM [30]. We note, however that in our interactive videophone schemes proposed at a later stage we refrained from employing the No Tx mode in an effort to avoid the associated latency of the buffering required for storing the information, until the channel quality improved sufficiently for allowing transmission of the buffered bits.

In [65]-[68] a range of novel radial basis function (RBF) assisted BbB-AQAM channel equalizers have been proposed, which exhibits a close relationship with the so-called Bayesian schemes. Decision feedback was introduced in the design of the RBF equalizer to reduce its computational complexity. The RBF DFE was found to give similar performance to the conventional DFE over Gaussian channels using various BbB-AQAM schemes, while requiring a lower feedforward and feedback order. Over Rayleigh-fading channels similar findings were valid for binary modulation, while for higher-order modems the RBF-based DFE required increased feedforward and feedback orders to outperform the conventional MSE DFE scheme. Then turbo BCH codes were invoked [67] for improving the associated BER and b/symbol performance of the scheme, which was shown to give a significant improvement in terms of the mean b/symbol performance compared to that of the uncoded RBF equalizer-assisted adaptive modem. Finally, a novel turbo equalization scheme was presented in [68], which employed an RBF DFE instead of the conventional trellis-based equalizer, which was advocated in most turbo equalizer implementations. The so-called Jacobian logarithmic complexity reduction technique was proposed, which was shown to achieve an identical BER performance to the conventional trellis-based turbo equalizer, while incurring a factor 4.4 lower "per-iteration" complexity in the context of 4 QAM.

In summary, in contrast to the narrowband, statically reconfigured multimode systems of [4]-[7], we invoked wideband, near-instantaneously reconfigured or BbB adaptive modulation to quantify the video performance benefits of such systems. It is an important element of the system that when the binary BCH [20] or turbo codes [73], [74], protecting the video stream are overwhelmed by the plethora of transmission errors, we refrain from decoding the video packet to prevent error propagation through the reconstructed frame buffer [9], [11]. Instead, these corrupted packets are dropped and the reconstructed frame buffer will not be updated, until the next packet replenishing the specific video frame area arrives. The associated video performance degradation is fairly minor for packet dropping or frame error rates (FERs) below about five. These packet-dropping events are signaled to the remote decoder by superimposing a strongly protected one-bit packet acknowledgment flag on the reverse-direction packet, as outlined in [9] and [11]. In the proposed scheme we also invoked the adaptive rate control and packetization algorithm of [9] and [11], supporting constant baud-rate operation.

Having reviewed a suite of various fixed-rate video coding assisted statically reconfigured video transceivers earlier, in the forthcoming subsection we will consider higher-rate H.263 based BbB-AQAM video transceivers.

Wideband BbB-AQAM Video Transceiver

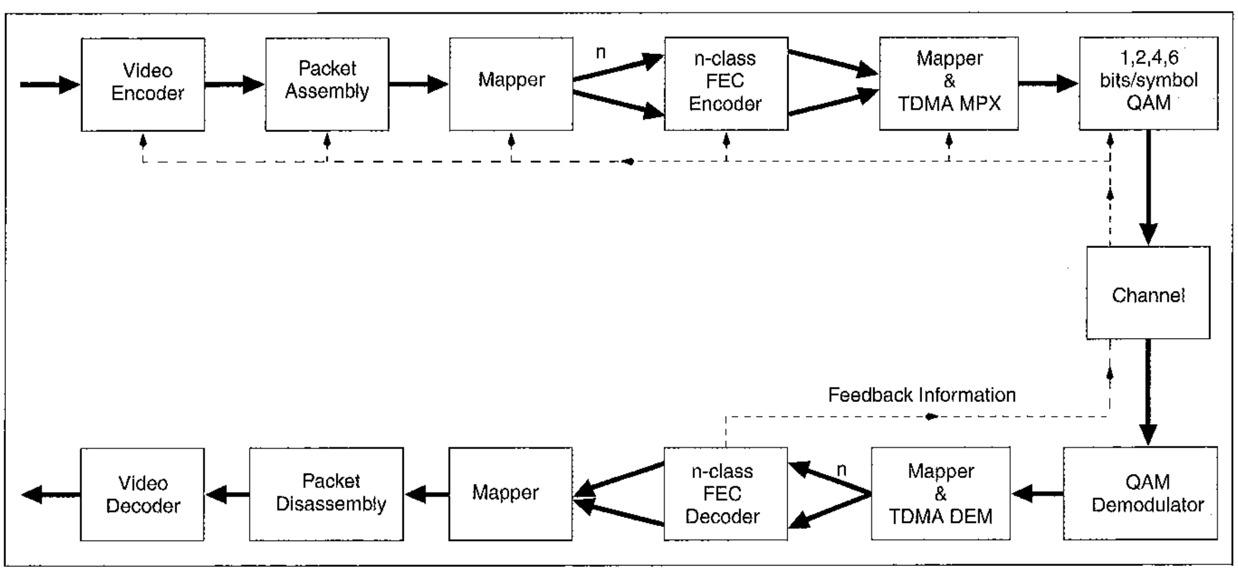
In this section we demonstrate the benefits of our wideband BbB-AQAM videophone system employing the programmable H.263 video codec in conjunction with our adaptive packetizer. The system's schematic is shown in Fig. 6, which will be referred to in more depth during our further discourse.

In these investigations, again, we used 176 × 144 pixel QCIF-resolution, 30 f/s video sequences encoded by the H.263 video codec [33] at bit rates resulting in high perceptual video quality. Table 4 shows the modulation- and channel parameters employed. The COST207 [69] four-path typical urban (TU) channel model was used, which is characterized by its CIR in Fig. 7. We used the Pan-European FRAMES proposal [70] as the basis for our wideband transmission system, invoking the frame structure shown in Fig. 8. Employing the FRAMES Mode A1 (FMA1) so-called nonspread data burst mode required a system bandwidth of 3.9 MHz, when assuming a modulation excess bandwidth of 50 [30]. A range of other system parameters are shown in Table 5. Again, it is important to note that the proposed AQAM transceiver of Fig. 6 requires a duplex system, since the AQAM mode required by the receiver during the next received video packet has to be signaled to the transmitter. In this system we employed TDD and the feedback path is indicated in dashed line in the schematic of Fig. 6.

Again, the proposed video transceiver of Fig. 6 is based on the H.263 video codec [33]. The video coded bitstream was protected by near-half-rate binary BCH coding [20] or by half-rate turbo coding [73], [74] in all of the BbB adaptive wideband AQAM modes [30]. The AQAM modem can be configured either under network control on a more static basis or under transceiver control on a near-instantaneous basis, to operate as a 1, 2, 4, and 6 b/symbol scheme, while maintaining a constant signaling rate. This allowed us to support an increased throughput expressed in terms of the average number of b/symbol, when the instantaneous channel quality was high, leading ultimately to an increased video quality in a constant bandwidth.

The transmitted bit rate for all four modes of operation is shown in Table 6. The unprotected bit rate before approximately half-rate BCH coding is also shown in the table. The actual useful bit rate available for video is slightly less than the unprotected bit rate due to the required strongly protected packet acknowledgment information and packetization information. The effective video bit rate is also shown in the table.

To be able to invoke the inherently error-sensitive variable-length coded H.263 video codec in a high-BER wireless scenario, a flexible adaptive packetization algorithm was necessary, which was highlighted in [9]. The



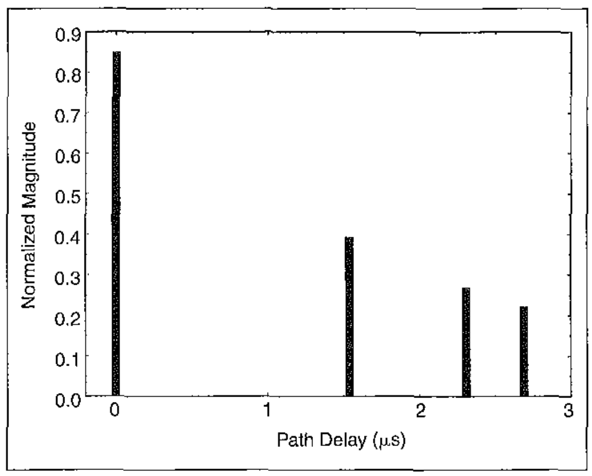
▲ *6. Reconfigurable transceiver schematic.*

Table 4. Modulation and Channel Parameters.	
Parameter	Value
Carrier Frequency	1.9 GHz
Vehicular Speed	30 mph
Doppler Frequency	85 Hz
Norm. Doppler Fr.	3.27×10^{-5}
Channel Type	COST 207 Typ. Urban (Fig. 7)
No. of Channel Paths	4
Data Modulation	Adaptive QAM (BPSK, 4-QAM, 16-QAM, 64-QAM)
Receiver Type	Decision Feedback Equalizer No. of Forward Filter Taps = 35 No. of Backward Filter Taps = 7

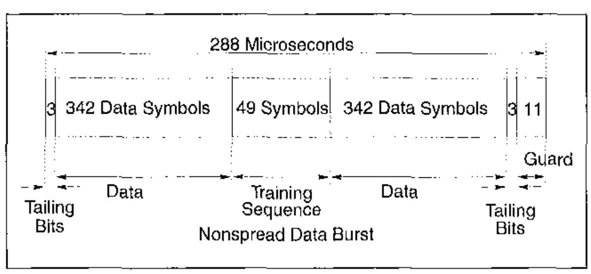
technique proposed exhibits high flexibility, allowing us to drop corrupted video packets, rather than allowing erroneous bits to contaminate the reconstructed frame buffer of the H.263 codec. This measure prevents the propagation of errors to future video frames through the reconstructed frame buffer of the H.263 codec. More explicitly, corrupted video packets cannot be used by either the local or the remote H.236 decoder, since that would result in unacceptable video degradation over a prolonged period of time due to the error propagation inflicted by the associated motion vectors and run-length coding. Upon dropping the erroneous video packets, both the local and remote H.263 reconstruction frame buffers are updated by a blank packet, which corresponds to assuming that the video block concerned was identical to the previous one.

A key feature of our proposed adaptive packetization regime is therefore the provision of a strongly error protected binary transmission packet acknowledgment flag [9], [11], which instructs the remote decoder not to update the local and remote video reconstruction buffers in the event of a corrupted packet. This flag, for example, can be conveniently repetition coded, to invoke majority logic decision (MLD) at the decoder. Explicitly, the binary flag is repeated an odd number of times and at the receiver the MLD scheme counts the number of binary ones and zeros and opts for the logical value, constituting the majority of the received bits. These packet acknowledgment flags are then superimposed on the forthcoming reverse-direction packet in our advocated time division duplex (TDD) regime [9], [11] of Table 5, as seen in the schematic of Fig. 6.

The proposed BbB-AQAM modem maximizes the system capacity available by using the most appropriate modulation mode for the current instantaneous channel conditions. As stated before, we found that the pseudo-SNR at the output of the channel equalizer was an adequate channel quality measure in our BbB adaptive wideband modem. A more explicit representation of the wideband AQAM regime is shown in Fig. 9, which displays the variation of the modulation mode with respect to the pseudo SNR at channel SNRs of 10 and 20 dB. In these figures, it can be seen explicitly that the lower-order modulation modes were chosen, when the pseudo-SNR was low. In contrast, when the pseudo-SNR was high, the higher-order modulation modes were selected to increase the transmission throughput. These figures can also be used to exemplify the application of wideband AQAM in an indoor and outdoor environment. In this respect, Fig. 9(a) can be used to characterize a hostile outdoor environment, where the perceived channel quality was low. This resulted in the utilization of predominantly more robust modulation modes, such as BPSK and 4 QAM. Conversely, a less hostile indoor environment is exemplified by Fig. 9(b), where the perceived channel quality was high. As a result, the wideband QAM regime can adapt suitably by invoking higher-order modulation modes, as evidenced by Fig. 9(b). Again, this simple example demonstrated that wideband AQAM can be utilized, to provide a seamless, near-instantaneous reconfiguration between for example indoor and outdoor environments.



▲ 7. Normalized channel impulse response for the COST 207 [69] four-path TU channel.



▲ 8. Transmission burst structure of the FMA1 nonspread data burst mode of the FRAMES proposal [70].

Table 5. Generic System Features of the Reconfigurable Multimode Video Transceiver, Using the Nonspread Data Burst Mode of the Frames Proposal [70] Shown in Fig. 8.

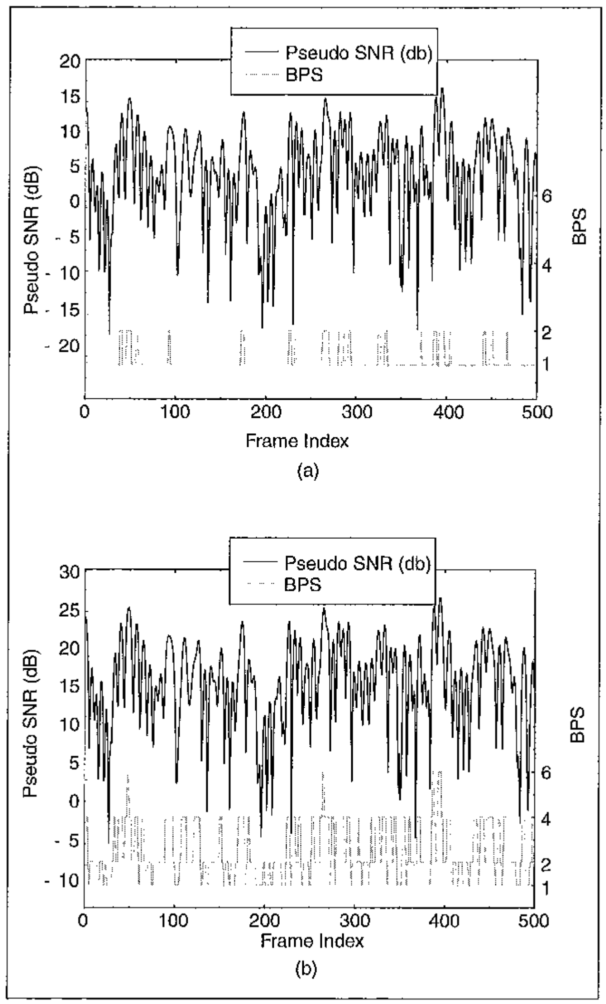
Features	Value
Multiple Access	TDMA
Duplexing	TDD
Number of Slots/Frame	16
TDMA Frame Length	4.615 ms
TDMA Slot Length	288 μs
Data Symbols/TDMA Slot	684
User Data Symbol Rate (KBd)	148.2
System Data Symbol Rate (MBd)	2.37
Symbols/TDMA slot	750
User Symbol Rate (KBd)	162.5
System Symbol Rate (MBd)	2.6
System Bandwidth (MHz)	3.9
Eff. User Bandwidth (kHz)	244

BbB-AQAM Performance

The mean BER and b/symbol performances were numerically calculated [46], [47] for two different target BER systems, namely for the **High-BER** and **Low-BER** schemes, respectively. The results are shown in Fig. 10 over the COST207 TU Rayleigh fading channel of Fig. 7. The targeted mean BERs of the **High-BER** and **Low-BER** regime of 1 and 0.01 was achieved for all average channel SNRs investigated, since this scheme also invoked a no-transmission mode, when the channel quality was extremely hostile. In this mode only dummy data was transmitted, to facilitate monitoring the channel's quality.

At average channel SNRs below 20 dB, the lower-order modulation modes were dominant, producing a robust system to achieve the targeted BER. Similarly, at high average channel SNRs the higher-order modulation mode of 64 QAM dominated the transmission regime, yielding a lower mean BER than the target, since no higher-order modulation mode could be legitimately invoked. This is evidenced by the modulation mode probability results shown in Fig. 11 for the COST207 TU Rayleigh fading channel of Fig. 7. The targeted mean BPS values for the **High-BER** and **Low-BER** regime of 4.5 and 3 were achieved at approximately 19 dB channel SNR for the COST207 TU Rayleigh fading channels. However, at average channel SNRs below 3 dB the above-mentioned no-transmission or transmission blocking mode was dominant in the **Low-BER** system and thus the mean BER performance was not recorded for that range of average channel SNRs.

The transmission throughput achieved for the **High-BER** and **Low-BER** transmission regimes is shown in Fig. 12. The transmission throughput for the **High-BER** transmission regime was higher than that of the **Low-BER** transmission regime for the same transmitted signal energy due to the more relaxed BER requirement of the **High-BER** transmission regime, as evidenced by Fig. 12. The achieved transmission throughput of the wideband AQAM scheme was higher than that of the BPSK, 4 QAM, and 16 QAM schemes for the same average channel SNR. However, at higher average channel SNRs the throughput performance of both schemes converged, since 64 QAM became the dominant modulation mode for the wideband AQAM scheme. SNR gains of 1-3 dB and 7-9 dB were recorded for the



▲ *9. Modulation mode variation with respect to the PSNR defined by (1) over the TU Rayleigh fading channel. The b/symbol throughputs of 1, 2, 4, and 6 represent BPSK, 4 QAM, 16 QAM, and 64 QAM, respectively.*

Table 6. Operational-Mode Specific Transceiver Parameters.

Features	Multirate System			
Mode	BPSK	4 QAM	16 QAM	64 QAM
Bits/Symbol	1	2	4	6
FEC	Near Half-Rate BCH			
Transmission Bit Rate (kb/s)	148.2	296.4	592.8	889.3
Unprotected Bit Rate (kb/s)	75.8	151.7	303.4	456.1
Effective Video Rate (kb/s)	67.0	141.7	292.1	446.4
Video Fr. Rate (Hz)	30			

High-BER and **Low-BER** transmission schemes, respectively. These gains were considerably lower than those associated with narrowband AQAM, where 5-7 dB and 10-18 dB of gains were reported for the **High-BER** and **Low-BER** transmission scheme, respectively [71], [72]. This was expected, since in the narrowband environment the fluctuation of the instantaneous SNR was more severe, resulting in increased utilization of the modulation switching mechanism. Consequently, the instantaneous transmission throughput increased, whenever the fluctuations yielded a high received instantaneous SNR. Conversely, in a wideband channel environment the channel quality fluctuations perceived by the DFE were less severe due to the associated multipath diversity, which was exploited by the equalizer.

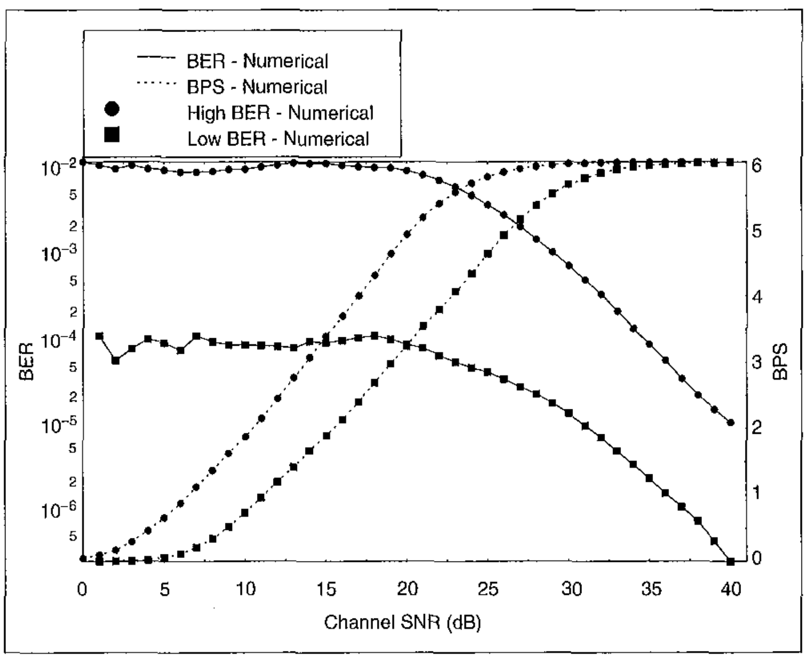
Having characterized the wideband BbB-AQAM modem's performance, let us now consider the entire video transceiver of Fig. 6 and Tables 4-6.

Wideband BbB-AQAM Video Performance

As a benchmarker, the statically reconfigured modems of the previous section were invoked in Fig. 13 to indicate how a system would perform, which cannot act on the basis of the near-instantaneously varying channel quality. As it can be inferred from Fig. 13, such a statically reconfigured transceiver switches its mode of operation from a lower-order modem mode—such as, for example, BPSK to a higher-order mode—such as 4 QAM, when the channel quality has improved sufficiently for the 4 QAM mode's FER to become lower than five after reconfiguring the transceiver in this more long-term 4 QAM mode.

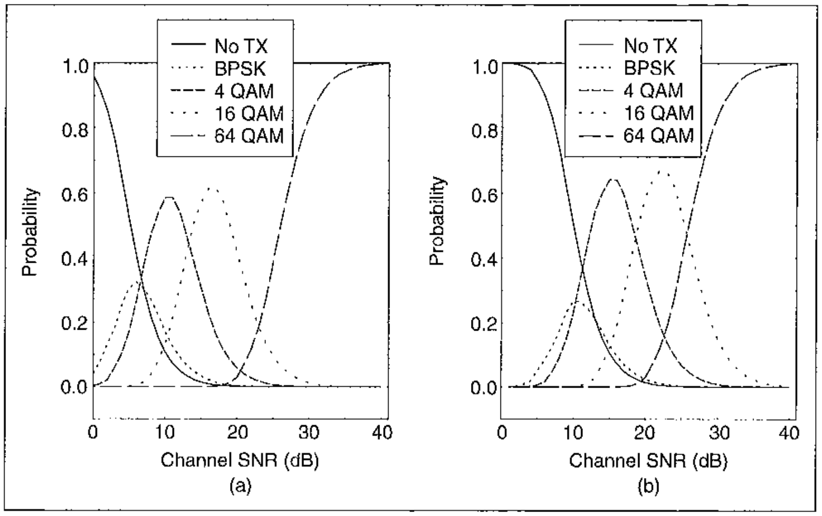
To assess the effects of imperfect channel estimation on BbB-AQAM, we considered two scenarios. In the first scheme the adaptive modem always chose the perfectly estimated AQAM modulation mode to provide a maximum upper bound performance. In the second scenario the modulation mode was based upon the perfectly estimated AQAM modulation mode for the previous burst, which corresponded to a delay of one TDMA frame duration of 4.615 ms. This second scenario represents a practical BbB adaptive modem, where the one-frame channel quality estimation latency is due to superimposing the receiver's required AQAM mode on a reverse-direction packet, for informing the transmitter concerning the best mode to be used for maintaining the target performance.

Figure 13 demonstrates on a logarithmic scale that the "one-frame channel estimation delay" AQAM modem manages to maintain a similar FER performance to the fixed rate BPSK modem at low SNRs, although we will see during our further discourse that AQAM provides increasingly higher bit rates, reaching six times higher values than BPSK for high channel SNRs, where the employment of 64 QAM is predominant. In this

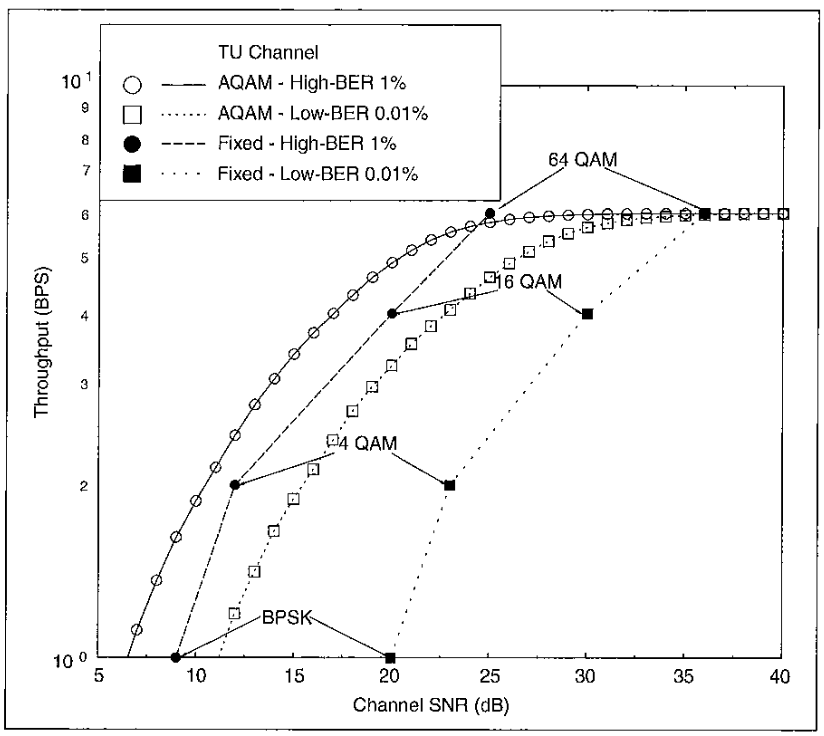


▲ *10. Numerical mean BER and b/symbol performance of the wideband equalized AQAM scheme for the* **High-BER** *and* **Low-BER** *regime over the COST207 TU Rayleigh fading channel.*

high-SNR region the FER curve asymptotically approaches the 64 QAM FER curve for both the realistic and the ideal AQAM scheme, although this is not visible in the figure for the ideal scheme, since this occurs at SNRs outside the range of Fig. 13. Again, the reason for this performance discrepancy is the occasionally misjudged channel quality estimates of the realistic AQAM scheme. Additionally, Fig. 13 indicates that the realistic AQAM modem exhibits a near-constant three FER at medium SNRs. The issue of adjusting the switching thresholds to achieve the target FER will be addressed in detail at a later stage in this section and the thresholds invoked will be detailed with reference to Table 7. Suffice to say at this stage that the average number of bits per symbol—and potentially also the associated video quality—can be increased upon using more "aggressive" switching thresholds. This results in an increased FER, however, which tends to decrease the video quality, as it will be discussed later. Having shown the effect of the BbB-AQAM modem on the transmission FER, let us now demonstrate the effects of the AQAM switching thresholds on the system's performance in terms of the associated FER performance.



▲ *11. Numerical probabilities of each modulation mode utilized for the wideband AQAM and DFE scheme over the TU Rayleigh fading channel for the (a)* **High-BER** *transmission regime and (b)* **Low-BER** *transmission regime.*

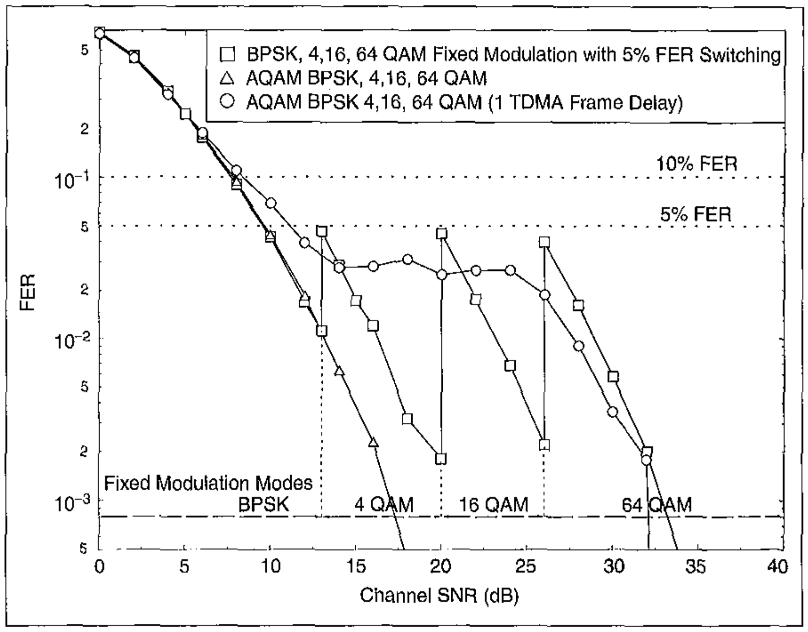


▲ *12. Transmission throughput of the wideband AQAM and DFE scheme and fixed modulation modes over the TU Rayleigh Fading channel for both the* **High-BER** *and* **Low-BER** *transmission regimes.*

AQAM Switching Thresholds

The set of switching thresholds used in all the previous graphs was the "standard" set shown in Table 7, which was determined on the basis of the required channel SINR for maintaining the specific target video FER. To investigate the effect of different sets of switching thresholds, we defined two new sets of thresholds, a more "conservative" set, and a more "aggressive" set, employing less robust, but more bandwidth-efficient modem modes at lower SNRs. The more conservative switching thresholds reduced the transmission FER at the expense of a lower effective video bit rate. By contrast, the more aggressive set of thresholds increased the effective video bit rate at the expense of a higher transmission FER. The transmission FER performance of the realistic BbB adaptive modem, which has a one TDMA frame delay between channel quality estimation and mode switching, is shown in Fig. 14 for the three sets of switching thresholds of Table 7. It can be seen that the more "conservative" switching thresholds reduce the transmission FER from about three to about one for medium channel SNRs, while the more "aggressive" thresholds increase the transmission FER from about 3% to 4-5%. Since FERs below 5% are not objectionable in video quality terms, however, this FER increase is an acceptable compromise for attaining a higher effective video bit rate.

The effective video bit rate for the realistic adaptive modem with the three sets of switching thresholds is shown in Fig. 15. The more conservative set of switching thresholds reduces the effective video bit rate but also reduces the transmission FER. The aggressive switching thresholds increase the effective video bit rate, but also increase the transmission FER. Therefore the optimal switching thresholds should be set such that the transmission FER is deemed acceptable in the range of channel SNRs considered. Let us now consider the performance improvements achievable, when employing powerful turbo codecs.



▲ 13. Transmission FER (or packet loss ratio) versus channel SNR comparison of the four fixed modulation modes (BPSK, 4 QAM, 16 QAM, 64 QAM) with 5% FER switching and adaptive BbB modem (AQAM). AQAM is shown with a realistic one TDMA frame delay between channel estimation and mode switching, and a zero delay version is included as an upper bound. The channel parameters were defined in Table 4 and near-half-rate BCH coding was employed [12].

Let us now demonstrate the additional performance gains that are achievable when a somewhat more complex turbo codec [73], [74] is used in comparison to similar-rate algebraically decoded binary BCH codecs [29]. The generic system parameters of the turbo-coded reconfigurable multimode video transceiver are the same as those used in the BCH-coded version summarized in Table 5. Turbo-coding schemes are known to perform best in conjunction with square-shaped turbo interleaver arrays and their performance is improved upon extending the associated interleaving depth, since then the two constituent encoders are fed with more independent data. This ensures that the turbo decoder can rely on two quasi-independent data streams in its efforts to make as reliable bit decisions, as possible. A turbo interleaver size of 18×18 bits was chosen, requiring 324 bits for filling the interleaver. The required so-called recursive systematic convolutional (RSC) component codes had a coding rate of 1/2 and a contraint length of $K=3$. After channel coding the transmission burst length became 648 bits, which facilitated the decoding of all AQAM transmission bursts independently. The operational-mode specific turbo transceiver parameters are shown in Table 8, which should be compared to the corresponding BCH-coded parameters of Table 6. The turbo-coded parameters result in a ten lower effective throughput bit rate compared to the similar-rate BCH-codecs under error-free conditions. However, Fig. 16 demonstrates that the pseudo-SNR video quality versus channel SNR performance of the turbo-coded AQAM modem becomes better than that of the BCH-coded scenario when the channel quality degrades. Having highlighted the operation of wideband single-carrier BbB AQAM modems, let us now consider how the above BbB adaptive principles can be extended to CDMA.

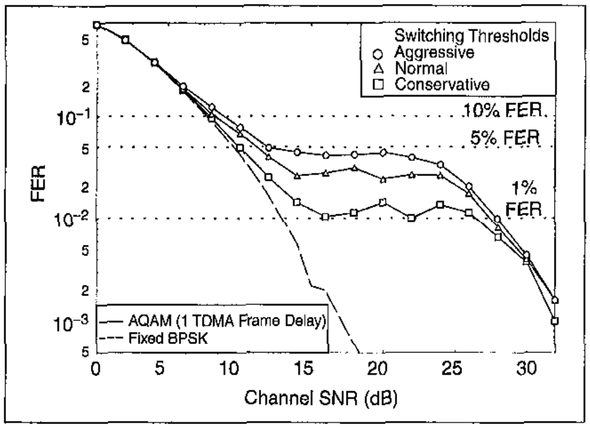
BbB Adaptive CDMA Video Transceivers

The application of the above BbB AQAM modems in the context of CDMA is fairly natural, and it is motivated by the fact that all three 3G mobile radio system proposals employ CDMA [19], [20]. The generation of the transmitted signal requires only one additional step, namely that of direct sequence (DS) spreading. Due to lack of space here we refrain from providing an introduction to DS-CDMA, since the associated literature of DS-CDMA is extremely rich [75]-[80].

Suffice to say that the previously characterized AQAM signals are spread with the aid of a set of so-called orthogonal spreading codes to a frequency range typically

Table 7. SINR Estimate at Output of the Equalizer Required for each Modulation Mode in BbB Adaptive Modem, i.e., Switching Thresholds.

	BPSK	4 QAM	16 QAM	64 QAM
Standard	<10 dB	≥10 dB	≥18 dB	≥24 dB
Conservative	<13 dB	≥13 dB	≥20 dB	≥26 dB
Aggressive	<9 dB	≥9 dB	≥17 dB	≥23 dB



▲ *14. Transmission FER (or packet loss ratio) versus channel SNR comparison of the fixed BPSK modulation mode and the adaptive BbB modem (AQAM) for the three sets of switching thresholds described in Table 7. AQAM is shown with a realistic one TDMA frame delay between channel estimation and mode switching. The channel parameters were defined in Table 4 [12].*

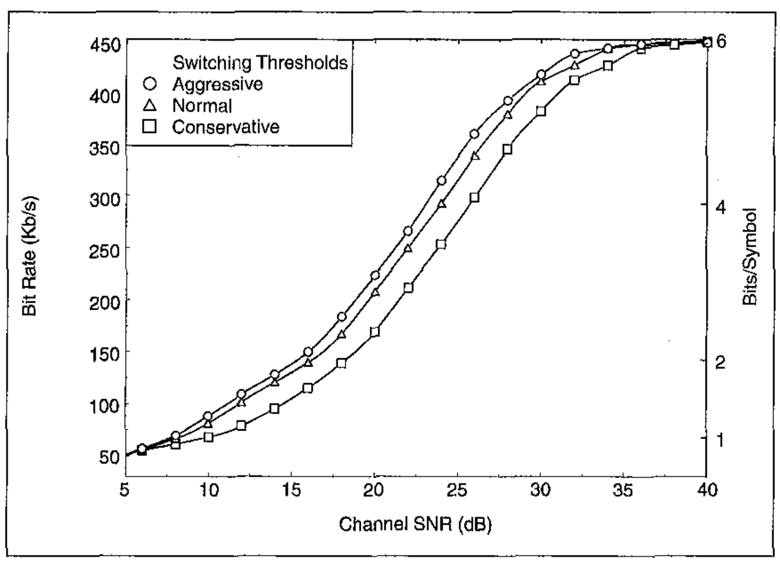
wider than that of the unspread AQAM signal. The spreading codes—provided that they are perfectly orthogonal—allow us to transmit the information of a number of mobiles within the same bandwidth without interfering with each other, as long as the number of users is lower than the so-called spreading factor given by the ratio of the unspread symbol duration to the chip-duration of the spreading sequence. The main advantage of spreading the signal is that the resultant spread or wideband signals are now subject to more significant frequency-selective multipath propagation. This is not a disadvantage, however, in the context of CDMA, since the inherent multipath diversity can be advantageously exploited to achieve a high diversity combining gain. More explicitly, the higher the spreading factor, the higher the number of so-called resolvable multipath components. The advantage of this is that—provided the associated receiver diversity combining complexity can be tolerated—the effects of fading can be mitigated and near-Gaussian performance can be achieved.

Over the years a plethora of CDMA detectors have been proposed [75]-[80]. However, the most significant contribution was the invention of the multiuser detector [76], [77], which allows us to remove the multiuser interference (MUI) at the receiver and hence to support a number of users within a given bandwidth, while maintaining a performance similar to that of a single-user scenario.

To quantify the associated near-instantaneous channel quality of our AQAM/CDMA system, the SINR is estimated at the output of the multiuser receiver for all the K users. This can be carried out by invoking the estimated CIRs and spreading sequences of all the users [81] and determining the MSE or pseudo-SNR at the output of the multiuser equalizer. The multiuser detector of [81] is amenable to BbB adaptive AQAM/CDMA operation, since it detects the received symbols, rather than their individual constituent bits and hence it does not have to be reconfigured each time, when the AQAM/CDMA mode is reconfigured.

In this context the MUI becomes analogous to the channel-induced ISI, where the ISI is generated by K propagation paths, rather than by K users. Following the estimation of the channel quality based on the SINR, the associated AQAM mode is chosen accordingly and signalled to the transmitter. The AQAM/CDMA switching regime is similar to that of (2). We note here that the lower the bandwidth of the spread CDMA signal, the lower the number of resolvable multipath components and hence the more dramatic the channel envelope fading. Hence in this scenario the performance benefits of AQAM/CDMA become more substantial. By contrast, when the chip-rate is high and hence there is a high number of resolvable multipath components, the channel's envelope fading becomes less dramatic, since there will always be a number of paths that are not faded, while some others are. Hence in this highly dispersive scenario the associated performance benefits of AQAM/CDMA may become more modest.

The fundamental advantage of AQAM/CDMA is that—irrespective of the propagation environment encountered—when the mobile roams across different environments subject to pathloss, shadow- and fast-fading

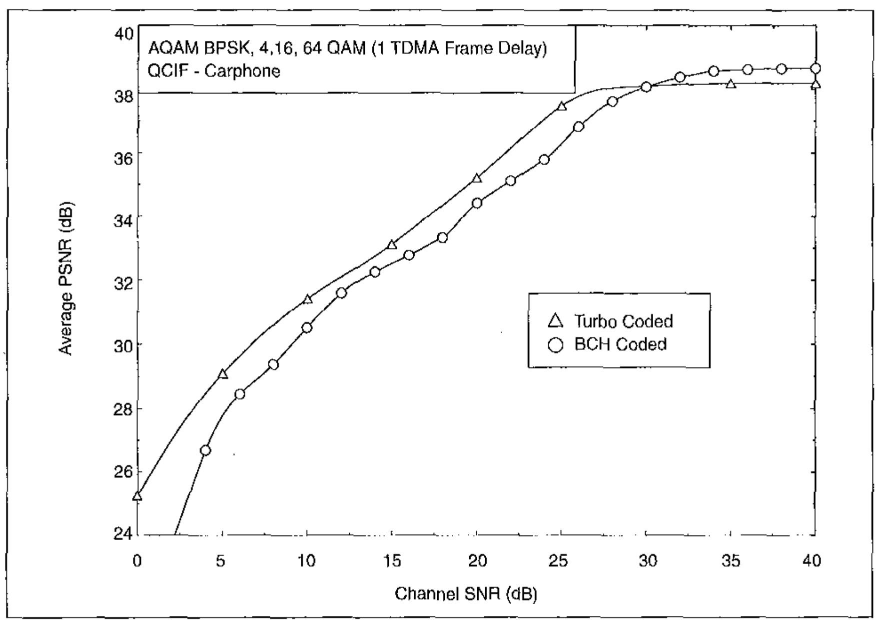


▲ *15. Video bit rate versus channel SNR comparison for the adaptive BbB modem (AQAM) with a realistic one TDMA frame delay between channel estimation and mode switching for the three sets of switching thresholds as described in Table 7. The channel parameters were defined in Table 4 [12].*

Table 8. Operational-Mode Specific Turbo-Coded Transceiver Parameters.				
Features	Multirate System			
Mode	BPSK	4 QAM	16 QAM	64 QAM
Bits/Symbol	1	2	4	6
FEC	Half-Rate Turbo Coding with CRC			
Transmission Bit Rate (kb/s)	140.4	280.8	561.6	842.5
Unprotected Bit Rate (kb/s)	66.3	136.1	275.6	415.2
Video Fr. Rate (Hz)	30			

[20], co-channel interference, ISI, MUI, power control errors, etc., the system will always be able to configure itself in the highest possible throughput mode, while maintaining the required transmission integrity. Furthermore, while powering up under degrading channel conditions may disadvantage other users in the system, invoking a more robust AQAM/CDMA mode will not.

In video performance terms, the slowly varying effective throughput of the AQAM/CDMA transceiver results in time-variant video bit rate. It is preferable, however, to have an inherently reduced-quality single video frame transmitted in a more robust, but lower-rate AQAM mode, rather than an originally higher-rate, higher-quality frame, which is corrupted in a more error-sensitive AQAM mode. The associated video pseudo-SNR is portrayed as a function of the channel SNR in Fig. 17 at an FER of 5% in contrast to that of the individual AQAM/CDMA modes. Observe in the figure that for example the 16 QAM fixed mode appears to slightly outperform the AQAM scheme in objective pseudo-SNR terms, when the system opts for more frequently employing the 4 QAM mode due to the gradually decaying channel quality. In reality in this scenario the perceptual video quality of the AQAM scheme is actually superior to that of the fixed 16 QAM mode, which experiences bursts of errors due to its inability to invoke a more robust modem mode. The same phenomenon was observed also near the BPSK/4 QAM switching threshold.

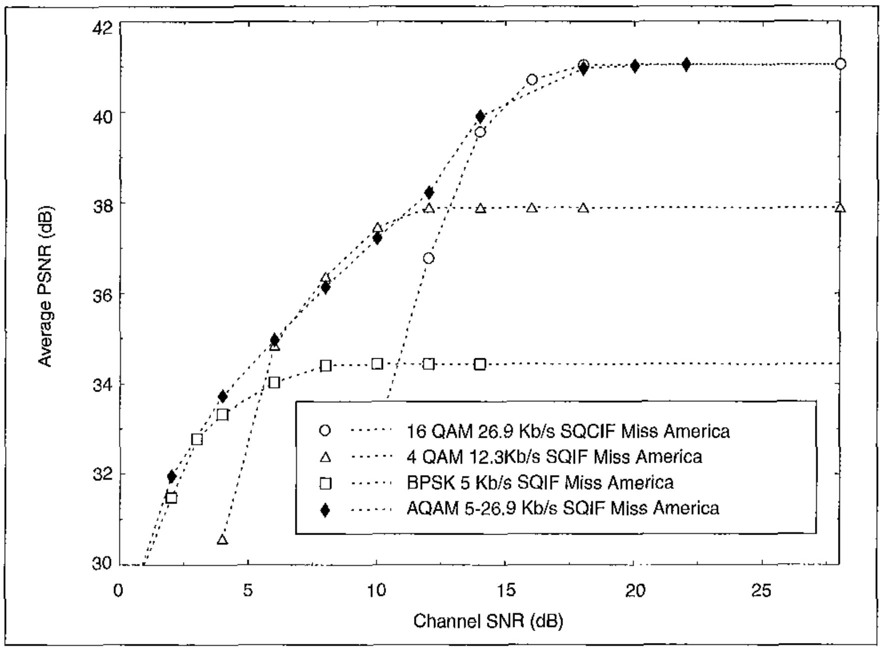


▲ 16. *Decoded video quality (PSNR) versus transmission FER (or packet loss ratio) comparison of the realistic adaptive BbB modems (AQAM) using either BCH or turbo coding. The channel parameters were defined in Table 4 [12].*

Summary and Conclusion

Initially our discussions centered around the choice of fixed- but programmable-rate video codecs for constant-rate wireless videophony. The VQ-codecs were found slightly superior to the other fixed-rate candidate codecs and hence the associated system design tradeoffs and transmission aspects were summarized in Table 3 in the context of the favored 8 and 11.36 kb/s, 10 f/s QCIF VQ-codecs. The associated head-and-shoulders videophone quality of various codecs can be studied under the URL address of http://www-mobile.ecs.soton.ac.uk using an MPEG player. The interested reader is referred to [4]-[7] for further details on the various fixed-rate proprietary candidate codecs and for a discussion on the associated transmission aspects. A range of H.263-based videophone schemes were contrived in [9]-[11].

In contrast to the statically reconfigured narrowband multimode video transceivers [4]-[7], we have advocated BbB-AQAM and AQAM/CDMA video transceivers, which employ the channel quality perceived by the channel equalizer or multiuser equalizer as the quality measure for controlling the AQAM modes. While in [46] the throughput upper bound of such an AQAM modem was analyzed, in this contribution a practical video transceiver was introduced and the achievable video performance



▲ *17. Video PSNR versus channel SNR comparision of the fixed modulation modes of BPSK, 4 QAM, and 16 QAM, and the BbB AQAM/CDMA modem, supporting two users with the aid of joint detection. These results were recorded for the Miss America video sequence at SQCIF resolution (128 x 96 pels) over the COST 207 bad urban channel model at an FER of 5% [3].*

gains due to employing the proposed wideband BbB adaptive modem was quantified. When our adaptive packetizer is used in conjunction with the AQAM modem, it continually adjusts the video codec's target bit rate, to match the instantaneous bit rate capacity provided by the adaptive modem.

We have also shown that the delay between the instants of channel estimation and AQAM mode switching has an effect on the performance of the proposed AQAM video transceiver. This performance penalty can be mitigated by reducing the modem mode signaling delay. It was also demonstrated that the system can be tuned to the required FER performance using appropriate AQAM switching thresholds. In harmony with our expectations, we found that the more complex turbo channel codecs were more robust against channel effects than the lower-complexity binary BCH codecs. Lastly, the AQAM principles were extended to joint-detection assisted AQAM/CDMA systems, where similar findings were confirmed to those found in the context of unspread AQAM.

In conclusion, wireless video telephony is technically realistic in the context of the existing 2G and the forthcoming 3G wireless systems. The 2G systems are more amenable to the employment of fixed-rate proprietary video codecs, rather than incorporating the H.263 codec, mapping their bit streams to an additional speech channel. For 3G systems the standard H.263 and MPEG-4 codecs can also be applied and videophony is likely to become one of their most important value-added service. The employment of the BbB-AQAM and AQAM/CDMA principles is realistic in the 3G systems and will ensure a substantially enhanced video quality in comparison to statically reconfigured transceivers. Future research has to concentrate on achieving tranceiver performance improvements with the aid of the MPEG-4 codec [41], turbo equalization [87], [88], space-time coding [82]-[86], adaptive frequency hopping [89], etc. to benefit the users of wireless videophone services.

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