MULTI-MODE WIRELESS VIDEOPHONY

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ABSTRACT

A comparative study of arbitrarily programmable, but fixed-rate videophone codecs using quarter common intermediate format (QCIF) video sequences scanned at 10 frames/s is offered. These codecs were designed to allow direct replacement of mobile radio voice codecs in second generation wireless systems, such as the Pan-European GSM, the American IS-54 and IS-95 as well as the Japanese systems, operating at 13, 8, 9.6 and 6.7 kbps, respectively, although better video quality is maintained over higher-rate, 32kbps cordless systems, such as the Japanese PHS and the European DECT and CT2 systems. Best overall performance was achieved by our vector-quantised codecs, followed by the discrete cosine transformed and the quadtree-based schemes, which were characterised by the bitallocation schemes of Table 1. The associated video Peak Signal-to-Noise Ratio (PSNR) was around 30 dB, while the subjective quality can be viewed under `http://www-mobile.ecs.soton.ac.uk`. A range of multimode wireless transceivers were also proposed, which are characterised by Table 2.

1. MOTIVATION

In recent years the concept of intelligent multi-mode transceivers (IMT) has emerged in the context of wireless systems [1]. Their aim is to provide the best possible compromise amongst 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 [2]. The conceptual framework of IMTs is supported by flexible implementations often referred to as software radios [3]. Due to lack of space in this treatise we have to limit our discourse to a small subset of these issues, mainly concentrating on the associated programmable, but fixed-rate video codec design aspects and referring the reader for a deeper exposure of the transmission concepts to the literature [25, 26, 27]. A further advantage of the IMTs of the near future is that due to their flexibility they are likely to be able to reconfigure themselves in various operational modes in order to ensure backwards compatibility with existing, so-called second generation standard wireless systems, such as the Japanese Digital Cellular [4], the Pan-American IS-54 [5] and IS-95 systems [6], as well as the Global System of Mobile Communications (GSM) standards [7].

The Journal of the Institute of Image Engineers of Japan, 1997, Vol. 26, No.6, pp 648-658

In order to provide wireless videophony services in the context of these existing so-called second-generation wireless systems, an additional speech channel has to be allocated to videophone users for the transmission of the video information. In this contribution we review some of the associated fixed but programmable-rate video coding aspects. We note, however that a substantial amount of further work has to be carried out in the area of intelligent interfaces, such as appropriate miniaturized video cameras, low-consumption video displays and in particular in the field of ergonomic hand-held portable multi-media communicator construction and design.

The corresponding speech channel rates of the above wireless systems are 6.7, 8, 9.6 and 13 kbps, respectively, and the proposed video codecs are capable of operating at a scanning rate of 10 frames/s, while maintaining such low bit rates, provided that low-dynamic head-and-shoulder video-sequences of the 176x144-pixel so-called Quarter Common Intermediate Format (QCIF) or 128x96-pixel Sub-QCIF video resolution are employed. We note, however that for high-dynamic sequences the 32kbps typical speech bitrate of cordless telephone systems, such as the Japanese 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 3rd generation adaptive multimedia, multimode terminal of the near future, which is currently under intensive study worldwide, for example within the European Community's Fourth Framework Programme in the field of Advanced Communications Technologies and Services (ACTS) [1].

In this short treatise we cannot consider the performance of the proposed video codecs in all of the above 2nd and 3rd generation mobile radio systems. Our main goal is to describe the design philosophy of our prototype video codecs and document their performance using two characteristic fixed bit rates, namely 8 and 11.36kbps within the above mentioned typical speech coding rate range. We will show that these codecs exhibit similar video quality, but different error resilience. Furthermore, we will devote special attention to transmission robustness issues and briefly summarise the features of a range of appropriate transceivers. Speech source coding aspects are beyond the scope of this paper, the reader is referred to for example [8, 9] for the choice of the appropriate speech codecs. Channel coding issues are addressed in Reference [43], while a detailed discussion of reconfigurable modulation is given for example in Chapter 13 of [32].

Following the above brief motivational notes, Section 2 outlines the design of a variety of programmable, but fixed-rate video source codecs and analyses their bit sensitivity. Specifically, Subsection 2.2 describes the gain-cost quantised, fixed but arbitrarily programmable rate discrete cosine transformed (DCT) video codec [25], while Subsections 2.3 and 2.4 highlight the vector-quantised (VQ) [26] and quad-tree (QT) [27] coded schemes, leading on to a brief comparison in Section 2.5 [30]. This is followed by a brief summary of the associated transmission aspects in Section 3 [26], before concluding in Section 4. Let us now commence our discourse on the choice of the appropriate video codec.

2. VIDEO COMPRESSION

2.1. Background

The theory and practice of image compression has been consolidated in a number of established monographs, such as for example Reference [10] by Jain. A plethora of video codecs have been proposed in the excellent special issues edited by Tzou, Mussmann and Aizawa [11] as well as Girod et al [12] for a range of bitrates and applications. Khansari, Jalali, Dubois and Mermelstein [16] as well as Mann Pelz [17] reported promising results on adopting the H.261 codec [15] for wireless applications by invoking powerful signal processing and error-control techniques in order to remedy the inherent source coding problems due to stretching its application domain to hostile wireless environments. Cherriman et al [24] proposed H.263-codec [13] based programmable transceivers or IMTs. Further important contributions in the field of video compression were due to Chen et al [18], Illgner and Lappe [19] Zhang [20], Ibaraki, Fujimoto and Nakano [21], Watanabe et al [22] etc, the MPEG4 consortium's endeavours [23], but the individual contributions by other prestigeous researchers are too numerous to review.

In contrast to existing and forthcoming standard variable-rate video compression schemes, such as the H.261 [15], H.263 [13, 14] and MPEG2, MPEG4 codecs [23], which rely on bandwidth-efficient but error-sensitive variable-length coding (VLC) techniques combined with a complex self-descriptive bitstream structure, our proposed codecs exhibit a more robust, regular bitstream and a constant bitrate. For the sake of improved robustness it is often advantageous to refrain from using variable-length coding. Hence in this treatise we attempt to offer a comparative study of a range of fixed but arbitrarily programmable-rate 176×144 pixel head-and-shoulder Quarter Common Intermediate Format (QCIF) video codecs specially designed for videotelephony over existing and future mobile radio speech systems on the basis of a recent research programme [24]-[30].

2.2. Low Bitrate DCT Codecs [25]

2.2.1. DCT Codec Schematic

The proposed programmable codec was designed to switch between intra- and inter-frame modes of operation, as seen in Figure 1. At the commencement of communications we transmit a low-resolution intra-frame coded frame, in order to initialise the reconstructed frame buffers. Once switched to inter-frame mode, any further mode switches are optional and only required if a drastic change of scene occurs. Spe-

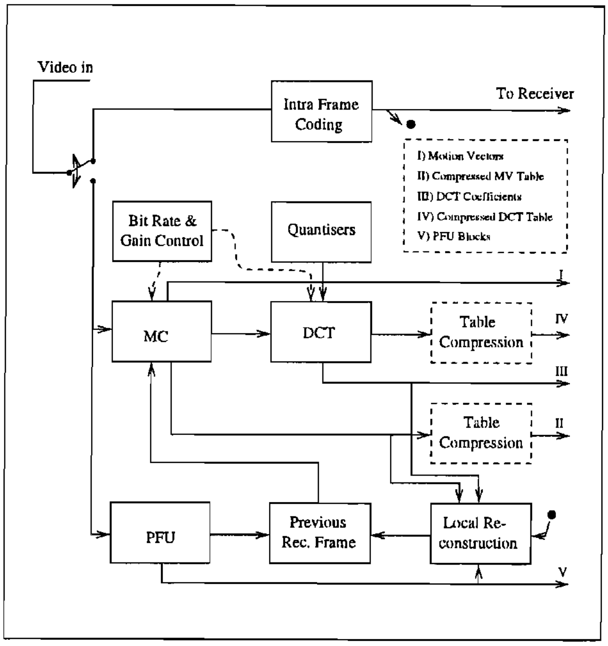

Figure 1: Schematic of the Multi-Class DCT codec ©ETT 1997, Streit, Hanzo [30]

cific algorithmic details of the codec were documented in Reference [25], hence here we refrain from elaborating on the motion active/passive and DCT active/passive block classification technique employed, which operates under the instructions of the bitrate control algorithm of Figure 1. We found that earmarking about 30-40 of the 396 8×8 QCIF blocks as motion-active and 30-40 as DCT-active was a good compromise in terms of video quality and implementational complexity for bitrates around 10 kbps. The side-information represented by the corresponding activity-tables was amenable to further compression at the cost of reduced robustness against channel errors, which is indicated by the optional activity-table compression printed in broken lines in Figure 1.

2.2.2. Adaptive Bit Allocation Strategy

The adaptive codec's bitallocation is summarised in Table 1. We found that the best subjective and objective videophone quality was achieved, when the number of active blocks for the motion compensation (MC) and DCT was roughly the same, although not necessarily the same 30 blocks were processed by the two independent algorithms. For the limited search scope of head-and-shoulders videophone sequences the encoding of the motion vectors (MV) requires only 4 bits per motion-active block, while that of the DCT coefficients needs 12 bits/block, including a 2-bit DCT coefficient quantiser classifier. Hence we earmarked between 1/2 and 2/3 of the available bit rate budget to the encoding of the DCT activity table, indicating, which of the blocks had high-energy DCT coefficients and hence were deemed to be DCT-active and to the quantisation of the DCT coefficients. The remaining bits were used for the MC and for the so-called Partial Forced Up-date (PFU) procedure employed to improve the codec's robustness, as detailed in Reference [25] . The PFU was typically configured to refresh 22 out of the 396 blocks in each frame. Therefore

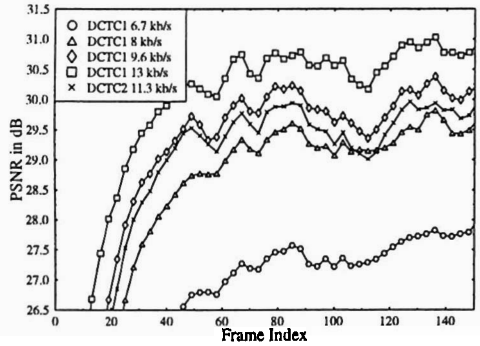

Figure 2: PSNR versus frame index performance of DCTC1 at various bit rates and for DCTC2 at 11.3 kb/s for the 'Claire' sequence ©ETT 1997, Streit, Hanzo [30]

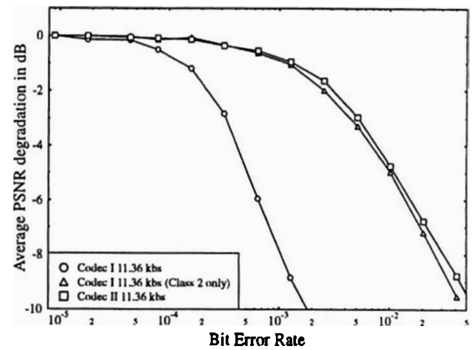

Figure 3: PSNR degradation versus BER for DCTC1 and DCTC2 ©ETT 1997, Streit, Hanzo [30]

$4 \times 22 = 88$ bits were reserved for the PFU. The actual number of encoded DCT blocks and MVs depended on the selected bit rate and typically varied between 30 and 50 for bit rates between 8 and 12 kbps at a scanning rate of 10 frames/s.

According to the optional dashed-line blocks in Figure 1, we designed two different codecs, a lower-rate but more error-sensitive scheme and a more robust, but slightly higher-rate codec. The output of the lower-rate codec contains two different-resilience classes of bits. Namely, the entropy encoded MC- and DCT-activity tables on one hand, which constitute the more vulnerable Class 1, and the less sensitive Class 2 MV, DCT and PFU bits on the other hand. The first class of information is, due to the reliance of the encoding procedure on Huffman coding, extremely vulnerable against any corruption. A corrupted bit is likely to create a code associated with a different length and, as a result, the entire frame may have to be dropped or re-transmitted. In our further discourse we will refer to this DCT codec as DCTC1.

However, since the high vulnerability of the Huffman-coded DCTC1 to channel errors is unacceptable in some applications, we also contrived another, more robust codec, which sacrifices coding efficiency and abandons the Huffman coding concept for the sake of improved error resilience. Explicitly, in DCTC2 we decided to transmit the index of each active DCT block and MV without the optional compression of Figure 1, requiring 9 bits to identify one of the 396 indices using the so-called enumerative method. The increased robustness of the codec is associated with an approximately 35 % increased bit rate. As Figure 2 reveals, DCTC1 at 8 kbps achieves a similar quality to that of DCTC2 at 11.3 kbps.

2.2.3. DCT Codec Robustness

The performance of DCTC1 was tested at 10 frames/s and 6.7, 8, 9.6 and 13 kbps, which are the speech rates of the Japanese Digital Cellular [4], American IS-54 [5], IS-95 [6] and the Pan-European GSM system [7], respectively. The results for DCTC2 are similar at a 35 % higher bit rate. [1]

As regards to the codec's error sensitivity, we have to differentiate between two possible error events. If the run-length encoded Class 1 bits are corrupted, it is likely that a codeword of a different length is generated and the decoding process becomes corrupted. Although this error is often detectable, since the erroneously decoded frame length becomes different from the currently expected number of bits per video frame, a single bit error can force the decoder to drop an entire frame. If, however, one of the more error-resilient Class 2 PFU, DCT or MV bits is corrupted, the decoder is unable to detect the error event, but only a maximum of two blocks are affected by such a single bit error, rather than dropping the entire frame. The error sensitivity difference between the run-length and non-run-length encoded bits is highlighted in Figure 3. If the whole bit stream of DCTC1 is subjected to random bit errors, a BER of $2 \cdot 10^{-4}$ is sufficient to inflict unacceptable video degradation. If, however, bit errors only affect the non-run-length encoded Class 2 bits, while the RL-coded bits remain intact, the codec can tolerate BERs up to $2 \cdot 10^{-2}$. In reference [25] we proposed an appropriate transmission scheme, which takes advantage of this characteristic. As evidenced by Figure 3, the absence of run-length encoded bits increases the error resilience of DCTC2 by an order of magnitude. Therefore DCTC2 is better suited for mobile applications over Rayleigh fading channels. Further issues of un-equal protection FEC and ARQ schemes are discussed in reference [25]. Having studied the algorithmic and performance issues of DCT-based codecs let us now concentrate our attention on a similar rudimentary performance study of vector quantised (VQ) codecs.

2.3. Vector-quantised Video Codecs [26]

Vector quantisation (VQ) is a generalisation of scalar quantisation, a technique lavishly documented in an excellent monograph by Gray and Gersho [31]. A comparative analysis of various VQ algorithms and their details were given in Reference [26], hence here we concentrate mainly on the performance of the codec. The VQ codec's schematic is akin to that of the DCT codec shown in Figure 1, with the exception of invoking a specially trained codebook for rep-

[1] DCTC1- and DCTC2-coded sequences at various bit rates can be viewed under the WWW address http://www-mobile.ecs.soton.ac.uk

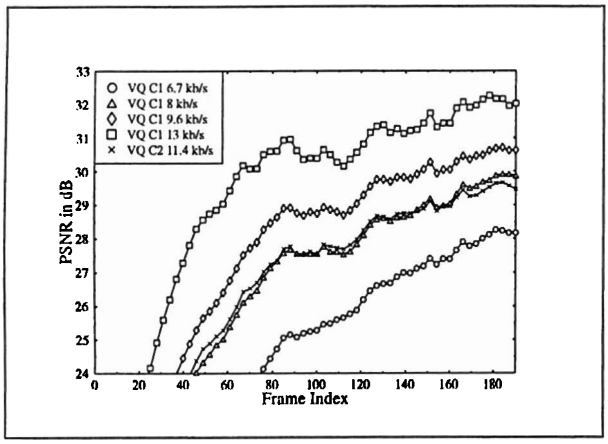

Figure 4: PSNR versus frame index performance of VQC1 at various bit rates and for VQC2 at 11.3 kb/s for the 'Claire' sequence ©ETT 1997, Streit, Hanzo [30]

resenting the 8×8 pixel blocks instead of encoding their DCT coefficients.

Similarly to the previously proposed DCT-based codecs, we contrived two VQ schemes, the lower-rate VQC1 and the more error resilient VQC2, both of which are characterised by Table 1. VQC1 achieved a higher compression ratio due to using the previously proposed table compression algorithms for the signalling of the location of the motion-active blocks and for that of the blocks exhibiting high motion compensated error residual (MCER). By contrast, VQC2 refrained from using vulnerable table compression techniques and hence exhibited a higher innate robustness against channel errors. Both codecs were based on the so-called classified VQ principle, using a codebook size of 256 which lead to an overall codec complexity of around 15 Mflops, when employing the previously mentioned active / passive block classification [26]. The MCER was generated for all 396 8×8 blocks and a bit-rate constrained fraction of the highest-energy 20-50 % MCER blocks were vector quantised.

The peak signal-to-noise ratio (PSNR) versus frame index performance of the VQC1 scheme is portrayed in Figure 4 for the 'Claire' sequence at the previously introduced 2nd generation mobile radio speech bit rates of 6.7, 8, 9.6 and 13 kbps. Lastly, the associated bit allocation schemes are summarised in Table 1 in contrast to our other prototype codecs. [2] This Table reveals a range of interesting aspects, showing for example that while the DCT codec allocated 12 bits/block for DCT-based MCER coding, for a similar video quality the VQ scheme required only an 8-bit, 256-entry codebook. Here we refrain from elaborating on the robustness issues of VQC1 and VQC2 due to lack of space. Their fundamental behaviour under erroneous channel conditions [26] is akin to that of DCTC1 and DCTC2, respectively, which was shown in Figure 3. These issues will be comparatively studied for DCT, VQ and QT codecs in Section 2.5. Let us now briefly consider quad-tree (QT) coded schemes.

[2] VQC1 and VQC2 encoded sequences at various bit rates can be viewed under the WWW address http://www-mobile.ecs.soton.ac.uk

Algorithm 1 *This algorithm adaptively adjusts the required QT resolution, the number of QT description bits and the number of MCER encoding bits required in order to arrive at the required target bit rate [27].*

1. Develop the full tree from the required minimum to maximum number of QT levels, for exampe from the video frame-level to pixel level.
2. Determine the *mse* gains associated with all decomposition steps for the full QT.
3. Determine the average decomposition gain over the full set of leaves at each decomposition level.
4. If the potentially required number of coding bits is more than twice the target number of bits for the given video frame, then delete all leaves having less than average gains and repeat Step 3.
5. Otherwise delete leaves on an individual basis, starting with the lowest gain leaf, until the required number of bits is attained for the video frame concerned.

2.4. Quad-tree based Codecs [27]

The proposed QT codecs also obey the structure of Figure 1, but the DCT-based MCER compression was replaced by QT-based compression. Again, for reasons of space economy, here we refrain from detailing the algorithmic design of the fixed-rate cost-gain quantised QT codec, the interested reader is referred to [27] for a detailed discussion. The codec's bit allocation scheme is summarised in Table 1 in contrast to our other benchmarkers.

Suffice to say here that upon assessing the potential of a number of different approaches to contriving an appropriate adaptive bit allocation scheme we finally arrived at Algorithm 1 [27]. Accordingly, the codec develops the QT structure from the top of the tree at frame level, down to a given maximum number of decomposition levels, which in the extreme may be the pixel level. The codec then determines the 'gain' of each decomposition step by evaluating the difference between the mean squared video reconstruction error of the 'parent block' and the total *mse* contribution associated with the sum of its four 'child blocks'. Observe that the purpose of Steps 3 and 4 is to introduce a bitrate-adaptive, computationally efficient way of pruning the QT to the required resolution, yielding the required target bitrate for the videoframe concerned. This allows us to incorporate an element of cost-gain quantised coding, while arriving at the required target bit rate without many times tentatively decomposing the image in various ways in an attempt to find the optimum fixed bit allocation scheme. The algorithm typically encountered 4-5 such fast QT pruning recursions, before branching out to Step 5, which facilitated a slower converging fine-tuning phase during the bit allocation optimisation.

In summary of our QT-coding investigations we concluded that due to the inherent error sensitivity of the QT-description code these schemes are similarly error sensitive to DCTC1 and VQC1 and their compression ratio is slightly

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 + 30×12	-	-	6	1136
DCTC1	22	22×4	< 350 (VLC)	< 350 (VLC)	-	-	VLC	800
VQC2	22	22×4	38×9 + 38×4	-	31×9 + 31×8	-	5	1136
VQC1	22	22×4	< 350 (VLC)	-	< 350 (VLC)	-	VLC	800
QTC	22	20×4	< 500 (VLC)	-	-	< 565 + 1 or 80	VLC	1136

Table 1: Bit Allocation Table ©ETT 1997, Streit, Hanzo [30]

more modest than that of the similar-robustness equivalent variable-length coded DCTC1 and VQC1 schemes. Viewed from a different angle, the Type 1 QT codecs exhibit similar bitrates to the more robust Type 2 DCTC2 or VQC2 arrangements. The PSNR versus bitrate performance of the QT-codec will be shown in Figure 6 in contrast to the DCT- and VQ-based schemes [3], while its robustness evaluated in terms of PSNR versus BER will be compared to that of our other benchmarkers in Figure 7. Having highlighted the salient features of the proposed QT codec let us now focus our attention on the performance comparison of the schemes considered.

2.5. Comparison of the candidate codecs [30]

We comparatively studied five different fixed-rate QCIF video codecs suitable for wireless videotelephony and evaluated their robustness. The corresponding bitallocation schemes were summarised in Table 1. The associated transmission issues, including source sensitivity-matched forward error correction (FEC) coding, adaptive modulation and automatic repeat request (ARQ) schemes have been discussed in depth in a series of companion papers published by the authors in references [25], [26], [27] and [29].

Let us finally compare our proposed inter-frame codecs to two widely used standard codecs, namely the MPEG-2 and H261 [15] codecs. These standard schemes are typically variable rate codecs, which make extensive use of variable-length compression techniques, such as RL-coding and entropy coding [10], although it is possible to invoke appropriate adaptive packetisation and multiple encoding operations in order to arrive at a required near-constant bitrate [24, 37]. An often employed alternative solution in distributive video applications is to use a buffer with a feedback to the quantiser, instructing the codec to invoke more coarse quantisers, when the buffer fullness exceeds a certain critical limit. Using buffering in interactive videotelephony is not a realistic alternative, since in case of 10 frames/s scanning the inherent latency is 100 ms. The voice signal's latency or delay is becoming objectionable for delays of 100 ms, hence perfect lip-synchronisation cannot be realistically achieved.

The above standard codecs also require the transmission of at least one intra-frame (I) coded frame at the commencement of transmission in order to provide a reference for the operation of the motion compensation. The transmission of I frames can be repeated at selectable regular intervals, in order to replenish the reconstructed frame buffer of the decoder, thereby mitigating the effect of prolonged transmission errors, yielding a regular surge in the bit rate. This is unacceptable in conventional fixed-rate second-generation mobile radio systems. In distributive video systems these

[3]Examples of QT-coded sequences can be viewed under the following WWW address: http:/www-mobile.ecs.soton.ac.uk

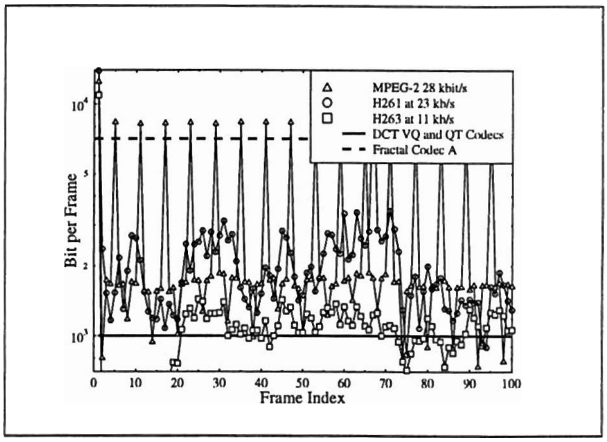

Figure 5: Bit rate fluctuation versus frame index for the proposed adaptive codecs and two standard codecs ©ETT 1997, Streit, Hanzo [30]

surges are smoothed by the adaptive buffers at the cost of a slight delay. Furthermore, if the I-frame is corrupted, it may inflict more severe video degradation than that due to previous inter-frame coded frame errors. In addition to the I frames, the H.263 and MPEG-2 codecs use two more modes of operation, namely, the so-called predictive coding (P) and bi-directional (B) coding modes, where the B-frames rely on differential coding strategies invoked with reference to the surrounding I and P frames. Due to the above robustness and delay problems we found that our distributed partial forced update (PFU) scheme was more appropriate for the targeted mobile radio applications.

In our experiments portrayed in Figure 5 we stipulated a fixed bit rate of 10 kbps for our three prototype codecs and adjusted the parameters of the H261 and MPEG-2 codecs to provide a similar video quality associated with a similar average PSNR performance. We note, however that the more recent variable-rate H.263 scheme slightly outperforms our fixed-rate schemes. The corresponding PSNR curves of our candidate codecs are displayed in Figure 6. Observe in Figure 5 that the number of bits / frame for our proposed codecs is always 1000, corresponding to 10 kbps and it is about twice as high for the two standard codecs, exhibiting a random fluctuation for the H261 codec. The MPEG codec exhibits three different characteristic bit rates, corresponding to the I, B and P frames in decreasing order from around 8000 bits / frame, to about 1800 and 1300, respectively.

A direct comparison of the above five codecs in Figures 5-7 reveals the following findings [30]:

- Our codecs achieve a similar performance to the MPEG-2 codec at less than half the bit rate. The H-261 codec

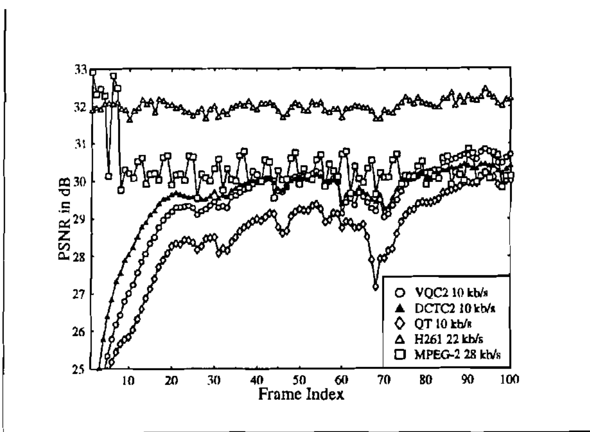

Figure 6: PSNR versus frame index performance of the proposed adaptive codecs and three standard codecs ©ETT 1997, Streit, Hanzo [30]

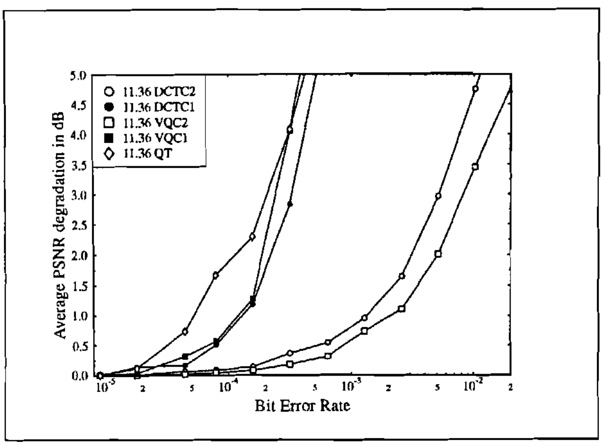

Figure 7: PSNR degradation versus BER for the proposed codecs ©ETT 1997, Streit, Hanzo [30]

at 22 kb/s, ie more than twice the bit rate, outperforms our codecs by about 2 dB in terms of PSNR. Note furthermore that our fixed-rate DCT and VQ codecs require about 20 frames to reach their steady-state video quality due to the fixed bit rate limitation, which is slightly prolonged for the QT codec.

- The delay of our codecs and that of the H-261 codec is in principle limited to one frame only. The delay of the H.263 and MPEG-2 codecs may stretch to several frames due to the P- and B-frames. In order to smoothe the teletraffic demand fluctuation of the MPEG-2 codec typically adaptive feedback controlled output buffering is used, which further increases the delay.
- The error resilience of the Type 1 codecs of Table 1 - namely that of DCTC1, VQC1 and QT - which use the runlength-compressed active / passive table concept is very limited, as is that of the standard MPEG2 and H.263 codecs. These arrangements have to invoke Automatic Repeat Request (ARQ) assistance over error-prone channels. Hence in these codecs single bit errors can corrupt an entire frame, or in fact several frames in case of the MPEG-2 codec. Thes problems are avoided by the slightly less bandwidt efficient non-run-length encoded Type 2 schemes, wh therefore exhibit an improved error resilience.
- Overall, the vector quantised codecs VQC1 and VQC were less prone to blocking artifacts than the DCT based codecs, while exhibiting also slightly higher bi error resilience and lower complexity. Hence the VQ based codecs constitute the best compromise in term of quality, compression ratio and computational de mand, closely followed by the DCTC1 and DCTC: candidate codecs. The QT codec does not lend it self to Type 2 implementations, since the QT code i rather vulnerable to channel errors, as it is evidencec by Figure 7, portraying the PSNR-degradation, rathe than the PSNR itself, which is more convenient fo: the comparison of different codecs, exhibiting different error-free PSNRs.

Let us now briefly highlight the associated transmissior aspects in the next Section.

3. 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 Reference [26]. The features of these systems are summarised in Table 2, although here due to lack of space we can only highlight these features with reference to [26].

Specifically, six different systems were designed, which used twin-class error correction coding, protecting the more vulnerable class 1 (C1) video bits more strongly than the less important C2 bits, whenever possible. This is however not always explicit in the Table, as seen for example for System 1, since we exploited that 16-level Quadrature Amplitude Modulation (16QAM) schemes possess two different integrity internal subchannels [32], which - when combined with various BCH codecs - can be invoked for the high-integrity delivery of the more vulnerable video bits. Pilot symbol assisted QAM (PSAQAM) was used [32]. Adaptive QAM schemes, which can reconfigure themselves on a burst-by-burst basis and hence are amenable to IMT implementations were considered for example by Webb, Steele, Morinaga, Sampei, Komaki, Chua, Goldsmith, Torrance, Cherriman et al in References [33]-[42].

All systems have a BCH-coded [43] rate of 20.32 kbps, implying that the lower video coding rates were more strongly BCH-coded, in order to resolve, as to whether it is worthwhile reducing the video codec's rate by variable-length decoding, in order to be able to incorporate a stronger BCH code. In some of the scenarios Automatic Repeat Request (ARQ) was invoked, which acted upon either any errors, ie Class One/Class Two video bit errors, or only in case of errors in the more sensitive Class One video bit stream, noting that the packets, which are retransmitted, would occupy additional transmission slots and hence reduce the system's teletraffic capacity. This is also shown in the Table. The corresponding user Baud-rates are 18 or 9KBd, depending on whether 4QAM or 16QAM was invoked by the transceiver. At a multi-user signalling rate of 144KBd, which fits in the 200KHz 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 re-transmission

Feature	System 1	System 2	System 3	System 4	System 5	System 6
Video Codec	VQC1	VQC2	VQC1	VQC2	VQC2	VQC1
Video rate (kbps)	11.36	8	11.36	8	8	11.36
Frame Rate (fr/s)	10	10	10	10	10	10
C1 FEC C2 FEC Header FEC	BCH(127,71,9) BCH(127,71,9) BCH(127,50,13)	BCH(127,50,13) BCH(127,92,5) BCH(127,50,13)	BCH(127,71,9) BCH(127,71,9) BCH(127,50,13)	BCH(127,50,13) BCH(127,50,13) BCH(127,50,13)	BCH(127,50,13) BCH(127,50,13) BCH(127,50,13)	BCH(127,71,9) BCH(127,71,9) BCH(127,50,13)
FEC-coded Rate (kbps)	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) 4/16QAM	5/11	11	4.5/10.5	6/11	8/12	12
Min. Rayleigh SNR (dB) 4/16QAM	10/22	15	9/18	9/17	13/19	17

Table 2: Summary of System Features ©IEEE 1997, Streit, Hanzo [26]

assistance, the number of users supported by Systems 2-6 is hence reduced by two. The effective user bandwidth was computed by dividing the 200KHz bandwidth by the number of users supported. The minimum required channel SNR values over both Additive White Gaussian Noise (AWGN) and Rayleigh channels was recorded in the Table for second-order switch-diversity reception using the appropriate PSNR versus channel SNR curves of Reference [26], along with the maximum number of videophone users supported within the 200KHz GSM-bandwidth. A range of interesting conclusions can be drawn from Table 2, which were set out more explicitly in Reference [26].

4. SUMMARY AND CONCLUSION

Our discussions centered around the choice of a fixed-, but programmable-rate video codec for constant-rate wireless videophony over existing second-generation mobile radio systems. In Section 2.5 the VQ-codecs were found slightly superior to the other candidate codecs and hence the associated system design trade-offs and transmission aspects were summarised in Table 2 in the context of the favoured 8 and 11.36 kbps, 10 frames/s QCIF VQ-codecs. The associated head-and-shoulders videophone quality of various codecs can be studied under `http://www-mobile.ecs.soton.ac.uk` using an MPEG player. The interested reader is referred to [25, 27, 26, 30] for further details on the various candidate codecs, on their comparison and on the associated transmission aspects.

5. ACKNOWLEDGEMENT

The financial support of the following organisations is gratefully acknowledged: Motorola ECID, Swindon, UK; European Community, Brussels, Belgium; Engineering and Physical Sciences Research Council, Swindon, UK; Mobile Virtual Centre of Excellence, UK.

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