HDTV

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DCT source coding and current implementations for HDTV

M. Barbero (RAI)

H. Hofmann (IRT)

N. D. Wells (BBC)

Following a brief explanation of the principal bit-rate reduction techniques employing discrete cosine transforms (DCT) which are applicable to wideband HDTV, the article describes their implementation in three codecs currently under development in Europe.

The picture quality achieved by these codecs, which reduce the gross bit-rate of a studio-quality HDTV signal by a factor of eight, or more, is typified by that of the EU256 system which was the subject of detailed performance tests early in 1991. These tests confirm the claim that DCT-based codecs are capable of a performance which is virtually transparent to the quality achieved in the studio.

1. Introduction

This article describes the techniques used in current European implementations of bit-rate-reduction codecs intended for HDTV transmission at bit-rates ranging from 70 Mbit/s to 140 Mbit/s. At these bit-rates, the received picture quality is very close to that of the studio original and this ensures that the full impression and impact of good high-definition pictures can be maintained at the receiver.

The source bit-rate of studio-quality HDTV pictures is in excess of 1000 Mbit/s. For digital transmission, this bit-rate must be reduced by a factor of 8 or more. For such bit-rate-reduction factors, a combination of techniques such as transform coding, motion-compensated interframe prediction and variable-length coding can be used. Systems using these techniques for the high-quality coding of conventional-definition television pictures within 34 Mbit/s have been developed, and a standard system for 34 Mbit/s coding has been specified by Interim Working Party CMTT/2 in conjunction with the European Broadcasting Union (EBU) and the European Telecommunications Standards Institute (ETSI) [1, 2, 3]. Much of the work within Europe reported in this paper is related to the development of 34 Mbit/s systems for conventional television. Systems for HDTV transmission have been developed within the Eureka 256 Project, the RACE HIVITS project No. 1018 and by the Institut für Rundfunktechnik in conjunction with Siemens.

Section 2 gives a general description of bit-rate-reduction techniques used in the current European HDTV codecs. *Section 3* then describes, in more detail, the different implementation strategies and finally *Section 4* reports on the measured performance of a typical codec.

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2. Methods of bit-rate reduction

2.1. Transform coding

In the technique of transform coding a block of *N* samples is transformed to give *N* coefficients. Each coefficient represents the amplitude of a specific pattern within the block. The inverse transform operating on the *N* coefficients gives back the *N* original samples. The aim of the transform is to take advantage of the correlation between samples within a block and to concentrate the picture information within as small a number of coefficients as possible. There are many different forms of transform but, among those which can be readily implemented, the discrete cosine transform (DCT) is considered the most efficient. Bit-rate reduction is more efficient if two-dimensional blocks are used and a common block size for television coding is 8 samples by 8 lines (taken either from a frame or a field, depending on the system).

The DCT is similar to the discrete Fourier transform in that the amplitudes of the coefficients relate closely to the amplitudes of the spatial frequencies present in the block. Bit-rate reduction in the DCT domain is effective because in the majority of blocks only a few coefficients have an amplitude which is significantly different from zero and only these coefficients need be transmitted. For typical pictures, the amplitudes of the coefficients are smaller for the higher frequencies. The average number of significant coefficients varies from block to block and therefore the bit-rate generated per block will inevitably be variable.

2.2. Coefficient quantization

After a block has been transformed, the transform coefficients are quantized. Different quantization is applied to each coefficient, according to the spatial frequency within the block that it represents. More quantization error can be tolerated in the high-frequency coefficients because high-frequency noise is less visible than low-frequency quantization noise. Also, quantization noise is less visible in the chrominance components than in the luminance component.

The coefficients are quantized by first dividing each coefficient by an appropriate weighting function for that coefficient which is derived from standard noise visibility curves. The weighted coefficients are then passed through a fixed quantization law which is a piecewise linear law with the spacing of the outer levels made more coarse in order to limit the total number of quantized values. There seems to be little advantage in having a non-linear quantization characteristic except for the inclusion of an increased threshold around zero level. This threshold is included in an attempt to maximize the number of coefficients which are quantized to zero.

The quantization parameters are varied to control the average bit-rate, for example, by affecting the number of non-zero coefficients. By making the quantizer more coarse the average bit-rate is reduced. In order to avoid the visibility of block boundaries, the coefficient representing the average DC level within the block is always coded with a given accuracy. The quantization parameters are held constant for a stripe of blocks.

Quantization noise is more visible in some blocks than in others, for example in blocks which contain a high-contrast edge between two plain areas. In such blocks, the quantization parameters can be modified to limit the maximum quantization error, particularly in the high-frequency coefficients.

2.3. Variable-length coding

The numbers corresponding to the quantized coefficients are passed to a variable-length coder (VLC). Variable-length coding is a technique in which the number of bits assigned to a given quantized value depends on the probability of occurrence of that value. Coefficient values which occur most frequently are assigned short code words and values which occur less often are assigned longer code words. In order to be able to decode the bit stream, it is arranged that no short code word is the first part of any longer code word.

Ideally, the lengths of the code words should be matched to the statistical distribution of the values to be coded. For each coefficient in a block, except the DC term, the distribution is centred around zero amplitude and the distribution becomes more peaked around zero for the higher-frequency coefficients. Therefore, for optimum performance, the way in which variable-length codes are assigned to coefficient amplitudes should vary with coefficient frequency. Also, the variable-length coding could adapt according to the picture content.

The most common order in which the coefficients of a block are fed to the VLC is such that those corresponding to lower spatial frequencies are sent first. This has the effect that several quantized coefficients in succession often have zero value. By coding these run lengths of zeros with a single code word, a useful saving in bit-rate can be obtained.

Other very efficient, variable-length codes have been proposed which code the individual bits of related coefficients rather than coding the complete coefficient amplitudes [2].

An example of a variable-length code is the one chosen within Europe for the 34 Mbit/s coding standard in which the code words are built up from pairs of bits. If the first bit of a pair is a zero then the code word finishes after the next bit. This code is used to code coefficient values and run lengths of zeros. A code word is also reserved to signal that the block has ended (after the last non-zero coefficient in a block). The maximum length of a code word is 18 bits. This code structure has the advantage that it is particularly easy to decode without any significant loss in efficiency.

An optimal VLC is designed by measuring the probabilities of occurrence of coefficient amplitudes and then assigning code words, for example by Huffman's algorithm. Such an optimum VLC may not have a convenient and fast decoding algorithm. In general, the bit-rate does not depend critically on the choice of VLC and in practice there seems to be around 10% variation in bit-rate between the best and the worst performing codes.

2.4. Prediction modes

In order to take advantage of the similarity of one picture to another, the coding schemes include temporal prediction modes. Typically, systems have two or more modes for predicting the DCT blocks; typical modes are:

- intrafield
- intraframe
- interfield
- interframe (with or without motion compensation).

In the intrafield mode the DCT operates on blocks taken from the current field of the input picture. In the intraframe mode the blocks are made up of picture elements taken from adjacent lines of an interlaced picture rather than a field. In the interfield mode the DCT is applied to blocks obtained by taking the difference between the current intrafield block and a prediction of this block formed from the previous field (by vertical averaging of the appropriate previous field lines without motion compensation). In the interframe mode, a prediction for each block is derived from the previous frame suitably shifted by an amount corresponding to an estimate of the motion for each block from the previous frame to the current frame.

For example, the CMTT/2 34-Mbit/s codec uses intrafield blocks and has three modes which are the intrafield, interfield and motion-compensated interframe modes. A block diagram of a 34 Mbit/s coder is given in *Fig. 1*. In some codecs, such as the IRT/Siemens system, the coding is based on intraframe blocks rather than intrafield blocks.

The coder decides, on a block-by-block basis or on a group-of-blocks basis, which mode should be used to code each block or group of blocks. This information is signalled to the decoder. The mode that gives the lowest bit-rate after variable-length coding should be chosen. In practice, the mean energy of the difference between the input picture and the prediction is compared for each block and for each mode and the mode which gives the least mean square value is chosen. This measure gives a very good comparative measure of the bit-rate after the VLC. Alternatively, the mean modulus of the difference can be used instead of the mean square for nearly equivalent results. Periodically, non-predictive intrafield or intraframe modes must be chosen in order to reset the decoder. Resetting enables the decoder to recover from transmission errors or from initial random access to the transmitted data stream.

2.5. Motion measurement

The most common method for measurement and the one for which ICs are available is "block matching". In this technique the current block is compared with all possible blocks in the previous frame, shifted relative to the current block by an appropriate vector within a given range of values. The previous frame block which gives the best match, in the least mean square sense for example, with the current block is chosen as the prediction for the current block. It has been found that improved performance can be obtained by specifying the motion vector to half-pixel accuracy. For non-integer motion shifts it is necessary to estimate the previous frame information at a position halfway between the sample sites. This is typically achieved by simple linear interpolation between the adjacent sample values.

In order to limit the bit-rate assigned to motion vectors, a single motion vector is usually assigned to a group of blocks, referred to as a "macro block". A macro block typically consists of two luminance blocks and two co-positioned C_R and C_B blocks.

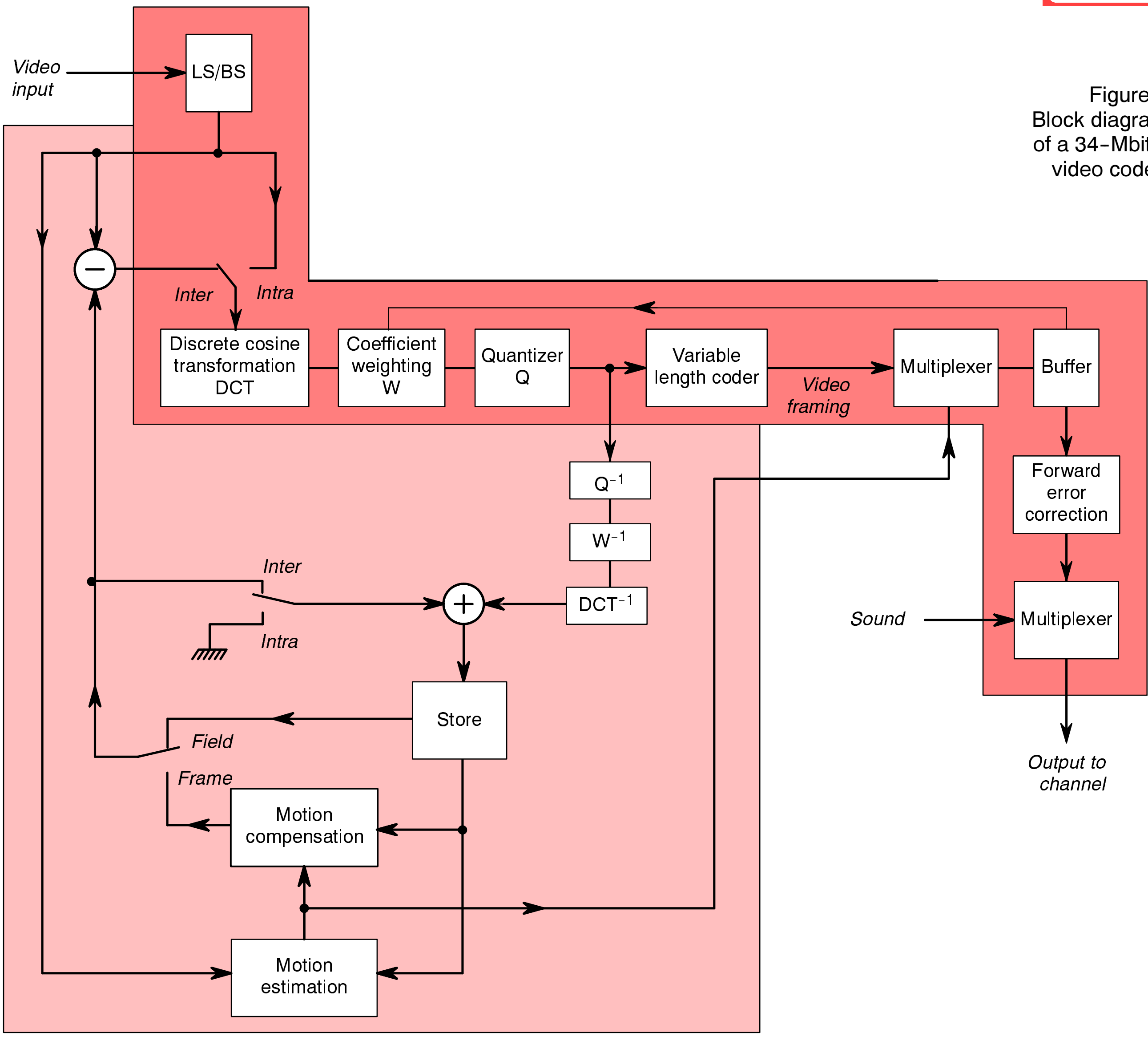


Figure 1
Block diagram of a 34-Mbit/s video coder.

2.6. *Video framing*

All the information relevant to the video signal is multiplexed together before being written into a buffer store.

There are many different solutions for the details of such a multiplex. For example, one solution is to have a stripe-of-blocks-based structure and another is to have a packet structure based around the error correction scheme. However, there are many similarities between the different approaches; for example, certain information is sent once per picture, some is sent once per stripe of blocks or once per packet of data and other information is sent on a macro block or block level.

Information that is sent *once per frame* is typically:

- frame synchronization word or picture start code;
- system type information (e.g. frame rate etc.);
- decoder synchronization information (such as time code or buffer level);

Information that is sent typically *once per stripe of blocks* or for *packets of data*:

- stripe or packet synchronization information;
- further decoder buffer resynchronization information;
- quantization step size information for that stripe;
- cyclic redundancy check information to enable concealment of uncorrected errors;

Information sent *once per macro block*:

- prediction mode for macro block;
- motion vector information;
- additional quantization information (if required, for critical blocks).

Information sent *once per DCT block*:

- transform coefficients;
- end-of-block code (or equivalent).

Variable-length coding is applied to all information where possible, as well as to the transform coefficients. For example, the motion vector is coded as a change in motion vector from one macro block to the next and the change in value is coded using a suitable VLC.

2.7. Buffer storage

Following the video multiplexer, the video information is read into a buffer store. Buffer storage is required at the coder in order to smooth the very irregular data rate coming from the DCT and variable-length coding. The size of the buffer is typically one or two frames at the transmitted bit-rate. At the decoder, there is a complementary buffer which receives the transmitted bit stream and gives out a variable bit-rate as required for the video decoding.

If the bit-rate generated by the video coding is greater than that permitted by the channel then the coder buffer will over-fill. Therefore, in order to limit the input data-rate, the coder buffer occupancy is fed back to coarsen the quantizer parameters. If the buffer is tending to under-fill then the quantizer is made more fine.

It is important that the decoder buffer is properly synchronized with the coder buffer, otherwise data can be lost from the decoder buffer. Decoder buffer synchronization can be achieved either by sending appropriate buffer occupancy information or by sending information which fixes the relative delay between coder input and decoder output.

2.8. Transmission multiplex

Finally, error correction is added to the data, the data is encrypted (if required) and then multiplexed together with accompanying sound data etc. The form of this multiplex for HDTV systems is highly system-dependant and will be described for each of the existing systems in following sections.

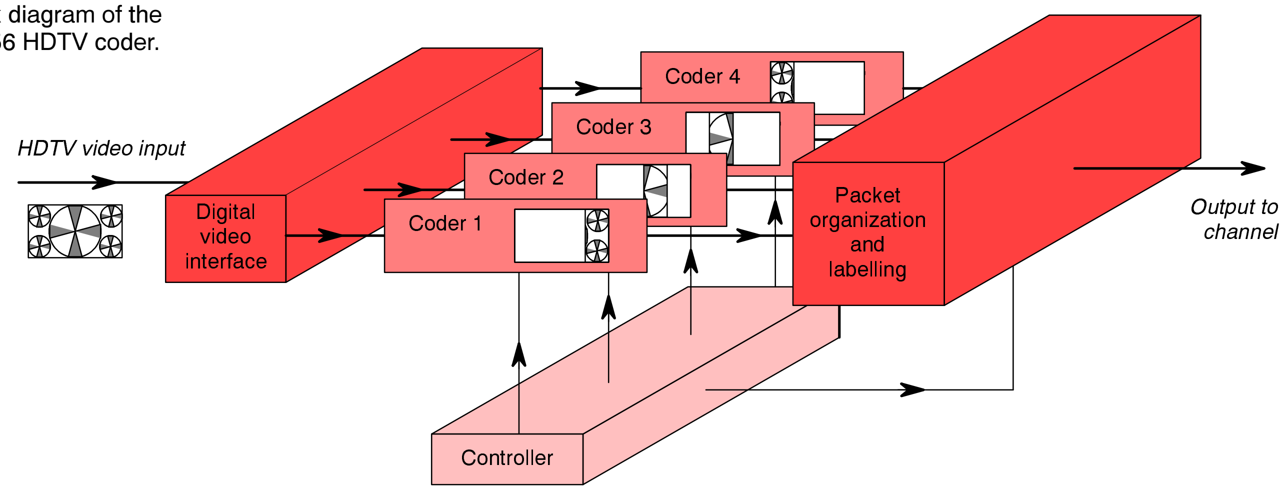
3. Currently implemented systems

Three DCT-based systems for the coding of HDTV at bit-rates between 70 Mbit/s and 140 Mbit/s are being used for the demonstrations of high-quality HDTV transmission. These systems have been developed and built by the Eureka 256 consortium, by IRT/Siemens and by the RACE HIVITS project No. 1018. Descriptions of the individual systems are given below.

3.1. Eureka 256 codec

In 1988, a European project, Eureka EU 256 was launched with the main goal of building codecs for conventional and high-definition television which would be suitable for the transmission of signals of contribution quality. However, the same algorithms and hardware can be tailored successfully for use in other applications, including distribution.

Figure 2
Block diagram of the EU256 HDTV coder.



The system adopted in the EU 256 project is based on the hybrid DCT. The DCT is computed on blocks of 8x8 samples; the adopted prediction modes are intra-field, inter-field and motion compensated inter-frame. A description of the system may be found in [4]. The B2-code has been chosen among the possible VLCs since it has a distribution of word lengths which is a good match to the distribution of the coefficient amplitudes and presents optimum characteristics from the point of view of its hardware implementation, word synchronization recovery and robustness in the presence of errors [5].

A prototype of the conventional-definition television (CDTV) coder operating at 34 Mbit/s was demonstrated during the meeting of Interim Working Party CMTT/2 held in Granada, Spain, in March 1990. This codec is characterised by the packet structure of the transmission multiplex which allows maximum flexibility in the sharing of the total bit-rate among the different services, i.e. video, audio and data, and the adoption of a wide range of transmission rates without any significant modification of the codec itself.

The HDTV codec was demonstrated for the first time in Turin, Italy, in March 1990. It has a modular structure which allows implementation with present-day technology. The HDTV picture is split into four vertical segments,as indicated in *Fig. 2*, and each of these segments is processed by a coding unit which, by itself, is capable of coding a single CDTV signal. Each coding unit is connected to a transmission buffer whose occupancy increases or decreases as a function of the complexity of the processed picture section.

A supervisory unit, known as the buffer controller, manages the four coding units. The weighting function which is applied to the blocks in the four segments to regulate the quantizer is determined by the controller on the basis of the buffer occupancy so that the quality is constant across the whole picture. Each stream is processed by a coding unit and the output data, organized as properly-labelled packets, are protected by a forward-error correction code (FEC) and multiplexed before being transmitted.

The use of packets makes the implementation of this modular architecture straightforward; in fact, each coding unit is identified as a separate data source, so packets relevant to each picture segment can be routed, at the receiving side, to the appropriate decoding unit.

On the occasion of the football matches of the WorldCup Italia'90, a technical demonstration of digital point-to-multipoint transmission of HDTV signal via the Olympus satellite was carried out in Italy by RAI-Radiotelevisione Italiana; Retevisión organized a complementary demonstration in Spain [6]. This experiment highlighted the ability of digital transmission systems to provide high-quality HDTV programmes to a selected audience on the occasion of special events or to distribute, via satellite, movies and information to HDTV theatres and cinemas. A total of more than 50 hours of transmission was carried out and more than 10,000 people had an opportunity to enjoy the matches displayed on wide screens.

The codecs used in the Italia '90 experiments were not equipped with motion estimation and compensation, which is present in the codecs available at the end of 1991. The EU256 HDTV coder and decoder each occupy a single 6U 19-inch rack.

3.2. IRT/Siemens system

3.2.1. General philosophy

The IRT/Siemens system [7] is based on DCT coding with a block size of 16x16 and on temporal DPCM using motion-compensated temporal prediction as shown in *Fig. 3*. Intra/inter-frame coding modes are used, in conjunction with the application of two different types of coding coefficient zones in the transform domain. Mode switching and parameter adjustment is controlled by a combination of information from the local image content and the buffer state. The whole coding algorithm works on a frame basis.

Motion-compensated prediction is applied between frames, on the basis of a full-search block-matching algorithm.

3.2.2. Mode selection

The next step is to switch between the different modes and adjust the parameters for bit-rate control. Accordingly, an "activity measure" and other criteria are applied; these are all computed as sums of differences in the spatial domain.

Intra/interframe mode.

On the basis of the difference criterion, the input block itself, or the prediction error block, is chosen for the DCT coding.

Two DCT zones adapted to the interlaced source format.

According to the local criterion, switching between two different types of coding zones for the DCT coefficients is carried out, taking into account the special situation connected with motion in an interlaced source format. The two zone types are related to static and moving image content, respectively.

Quantization and coefficient selection.

Irrelevancy reduction is applied by an adaptive change of the linear quantizer step size and a threshold operation that varies with the diagonal number which is related to the spatial frequency. In this way, the area of the coefficient set to be transmitted is defined adaptively. Quantization and thresholding parameters are controlled according to both the image block content (activity measure) and the buffer state.

The significant coefficients in a DCT block are entropy coded using different classes with corresponding bit assignment matrices for Huffman tables. In each class, one Huffman table is assigned for each coefficient. The classification of the block is based on the energy content of the coefficients. Also, a class is reserved for blocks that are not coded at all.

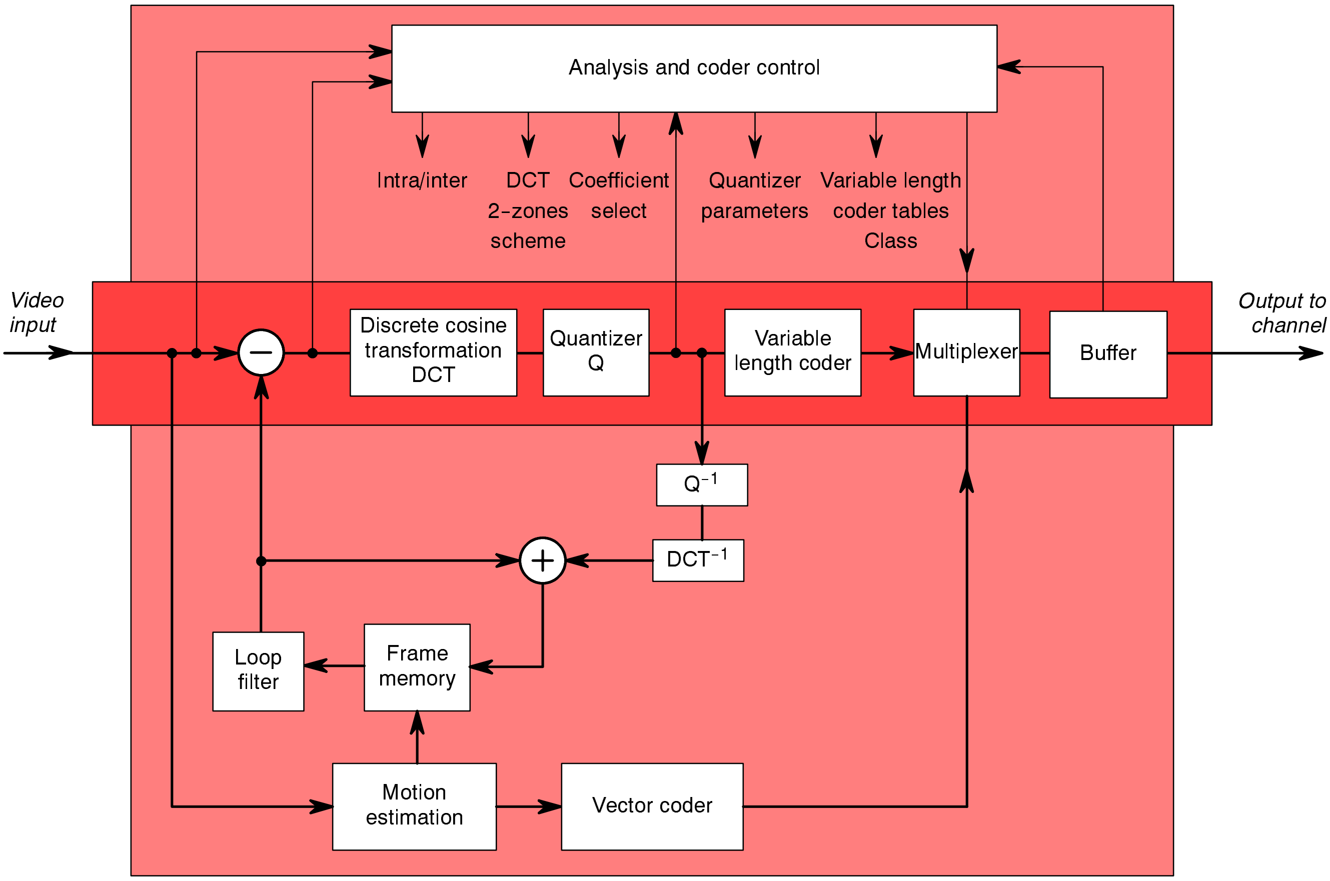
In the design of this coding concept, the idea was to put the most complex operations completely on the coder side and not on the decoder side. The coding algorithm is therefore especially suited to HDTV distribution applications.

3.2.3. *Source and channel coding*

Taking into account the time available for the evaluation of the coding algorithm, the definition and development of hardware components as well as possible changes to improve the coding algorithm, it was decided to build the channel coder and the source coder in software as a first step.

The coding of HDTV source signals, the application of an error-correction algorithm, the multiplexing of picture, sound and data as well as all tasks for the channel coding will be carried out on an HDTV picture processing system (*Fig. 4*).

Figure 3
Block diagram of the IRT/Siemens adaptive DCT coder.



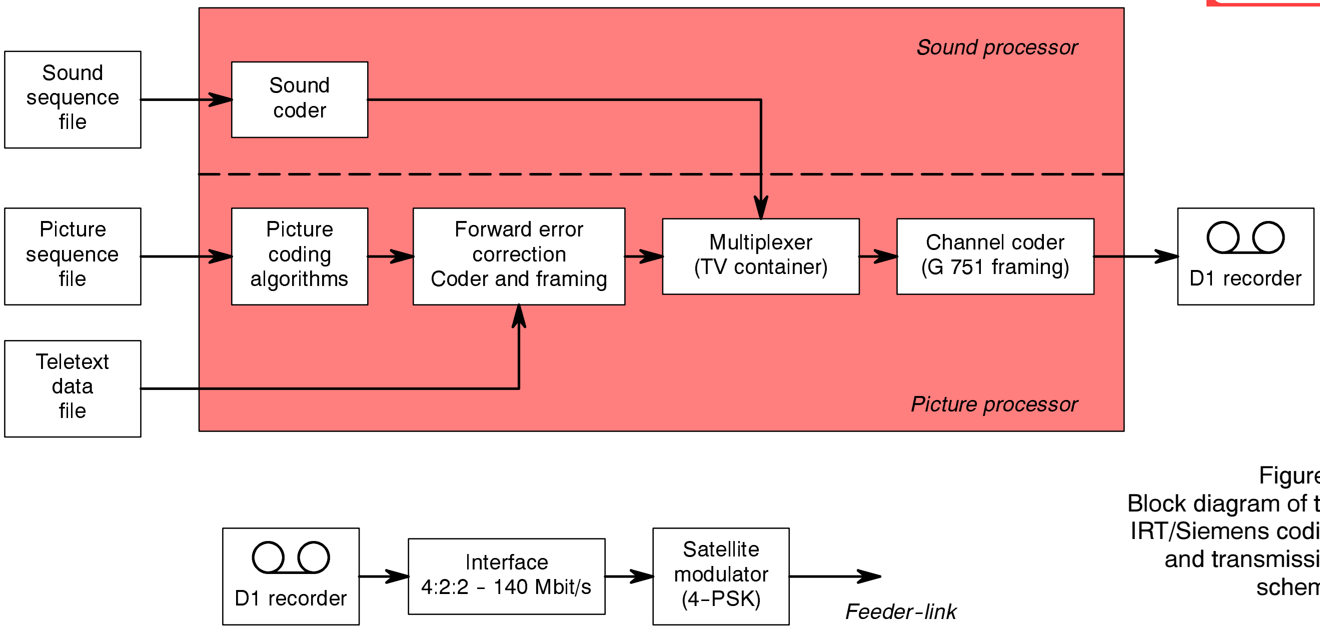


Figure 4
Block diagram of the IRT/Siemens coding and transmission scheme.

The data to be sent to the decoder is combined to form stripes of blocks containing:

- selected coefficients, which are VLC-coded and organized in quadblocks;
- control parameters for each quadblock;
- one motion vector for each luminance block;
- buffer state for each stripe of blocks;
- a unique number for each stripe of blocks;
- a stripe synchronization word.

In order to minimize the disturbances caused by uncorrected transmission errors, each quadblock is allocated its own synchronization word and a unique quadblock number. Computer simulations showed very satisfactory behaviour of these arrangements, working in conjunction with the numbering system for stripes of blocks, in the presence of uncorrected transmission errors.

Video data (124,740 kbit/s) and ancillary data (1,080 kbit/s) are multiplexed together and the multiplexed data is protected by a non-interleaved (233/249) Reed-Solomon error-correcting code.

The data is then combined with the sound data using the "TV container", as described in CCIR Recommendation 721, leaving another 512 kbit/s as a service channel. The TV container output rate is 138 240 kbit/s and fits in a channel frame of the Plesiochronous Digital Hierarchy (PDH) with a transmission rate of 139,264 kbit/s according to CCITT G.751 as well as in a channel frame according to CCITT G.709 for the Synchronous Digital Hierarchy (SDH). Furthermore, direct mapping into SDH-based channels is possible.

Data for transmission is recorded on a D1 digital video recorder. During a satellite transmission, the "pictures" played back from the D1 tape carrying the channel data is converted into a single 140 Mbit/s data stream (CMI) which can be fed to the modulator of the satellite up-link.

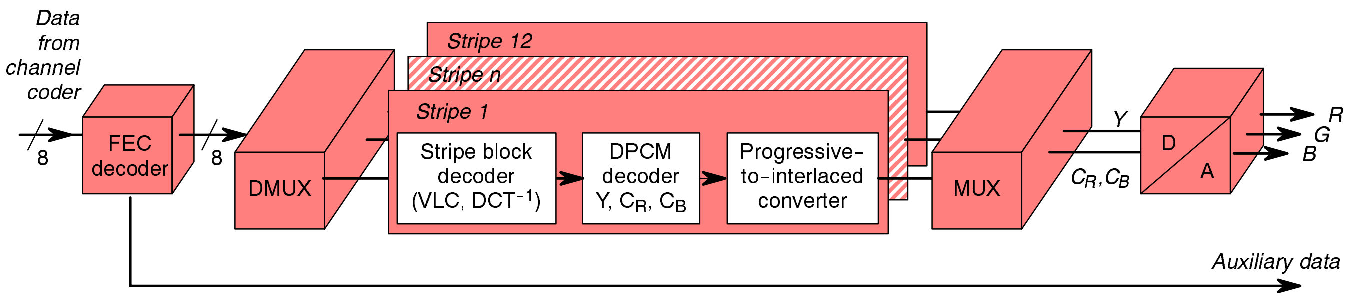


Figure 5
IRT/Siemens picture decoder.

3.2.4. *Source decoder*

In order to reduce the processing speed for the hardware decoder, the decoder is split into twelve parallel decoders working on successive stripes of blocks (*Fig. 5*). Each sub-decoder has its own buffer store. Some form of interworking is necessary between these decoders to carry out motion compensation and filtering in the DPCM loop. The IRT/Siemens decoder occupies a single 10U 19-inch rack.

3.3. HIVITS codec

The HIVITS codec is based on the use of six sub-coders and sub-decoders operating in parallel on adjacent stripes of HDTV blocks as shown in *Fig. 6*. Each sub-codec is based on the 34 Mbit/s DCT system shown in *Fig. 1*. Previous-frame information is passed between adjacent sub-coders (and adjacent sub-decoders) for motion estimation and compensation purposes, within each sub-coder.

Some form of parallelism is necessary in order to reduce the processing speed for individual functional blocks within the codec and this particular form of stripe-based parallelism was chosen in order that the transmitted bit-stream should appear the same as if the coder was implemented with a single DCT loop of the form shown in *Fig. 1*. In this case, the bit-stream is a suitably scaled version of that defined for the European standard 34 Mbit/s DCT system which has a stripe-based multiplex structure. Then, an HDTV coder working with any number of sub-coders in parallel should be compatible with a decoder operating with a different number of sub-decoders in parallel. This gives flexibility in the design of future implementations of the codec and ensures compatibility between different generations of equipment.

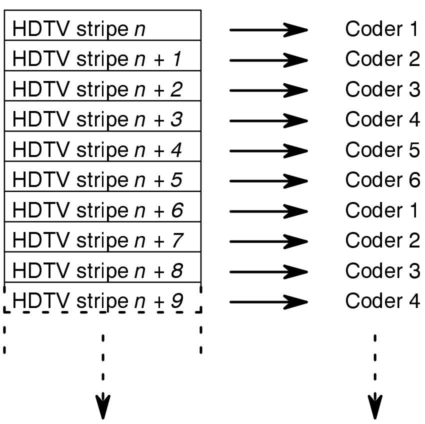


Figure 6
Parallel structure adopted in the RACE HIVITS codec.

Each sub-coder and sub-decoder has its own buffer store and one stripe of HDTV information is read from each coder buffer in turn and transmitted to the corresponding sub-decoder buffer. To ensure that all sub-codecs operate with the same quantization parameters, the individual buffer occupancy values in the coder are monitored and an equivalent total buffer occupancy is calculated. This total buffer occupancy value is used to control the quantization parameters for all the sub-coders. Synchronization of the sub-decoders is maintained by sending information at the start of each stripe, giving the timing of the input video signal. This allows the decoder to construct a video timebase with a relative delay which is appropriate for the effective buffer size and the transmission rate.

The variable-length coding scheme used in the current implementation of the HIVITS HDTV codec is not based on the B2 code (as defined for use in 34 Mbit/s codecs) but on an arithmetically-computed VLC (ACVLC) [2]. In this code the structure and the lengths of the individual code words can be computed readily from a set of eight parameters. This allows the VLC to adapt according to the coordinates of the coefficient being coded, according to the DCT coding mode and according to luminance or chrominance components. 16 different code tables are used and the maximum length of code word is 32 bits.

The format for the information sent for each block of coefficients is as follows:

- number of non-zero coefficients;
- the relative addresses of these non-zero coefficients;
- the VLCs of these coefficients.

All information is coded using one of the 16 ACVLC tables. This code uses the number of coefficients data rather than an “end of block” code to determine the boundary between blocks. Also, the relative addressing of the coefficients is equivalent to run-length coding of the zero coefficients.

The blocks are grouped together in macro blocks consisting of two luminance blocks and the two co-sited chrominance blocks. One motion vector is transmitted per macro block. The range of the motion vector is ± 15 pixels horizontally and ± 8 field lines vertically. The motion vector is measured to half-pixel accuracy.

The information for a stripe of blocks is grouped together and preceded by a unique stripe synchronization word. The stripe synchronization word is followed by quantizer step size information and decoder timebase timing information. A 16-bit cyclic redundancy check word is transmitted at the end of each stripe in order to detect any residual transmission errors. If an error is detected at the decoder then the whole stripe is concealed at the decoder output using the corresponding stripe from the previous frame.

The video data is protected by a double interleaved (239,255) Reed Solomon error correcting code and multiplexed together with a 2048 kbit/s sound channel into a "TV container" which in turn is mapped into a 139,264-Mbit/s G751-compatible bit-stream suitable for a standard plesiochronous transmission channel. The TV container has a total bit-rate of 138,240 Mbit/s and is designed to be compatible with future synchronous "SDH" broadband transmission networks.

The HIVITS HDTV coder and decoder each occupy a single 6U 19-inch rack.

4. Measurement of performance

All three systems described above are based on comparatively similar algorithms and the picture qualities obtained are likely to be similar for all the systems across a range of bit-rates. The picture quality was measured for the EU 256 system and the results are described here.

The main method used to optimise and evaluate the basic picture quality of bit-rate-reduction codecs is the use of subjective assessment. Unfortunately, this is difficult, at present, in the case of HDTV signals because there are technological limits in the present source and display equipment which are such that the codec capabilities cannot be fully exercised. Moreover, it is difficult and time-consuming to set up a series of subjective tests because at least two digital VTRs would be required to edit the test sequences to be presented.

Mr. H. Hofmann *studied communications engineering at the Technical University of Munich. He joined the Institut für Rundfunktechnik in 1969, working initially in the field of holography and optical communication. He has also been involved in studies of bit-rate reduction techniques, teletext and the digital processing and transmission of conventional and high-definition television. In 1986 he was appointed to lead the picture processing and transmission section of the IRT, and since 1990 has been head of the TV systems and psycho-optics section.*

Mr. Hofmann is a member of several national and international study groups working on digital television.

Dr. N.D. Wells *graduated from Cambridge University in 1972 and then studied for his doctorate at Sussex University. He joined the BBC in 1977 and has worked for many years in the field of image coding and bit-rate reduction for television transmission.*

Mr. Marzio Barbero *obtained a degree in electrical engineering at Turin Polytechnic in 1977. In 1978 he joined the Research Centre of RAI-Radiotelevisione Italiana in Turin, where his research interests include digital video and audio systems. He is a member of EBU Specialist Groups V1/RDB and V1/HDTV which are engaged in studies of digital video coding and transmission systems for conventional and high-definition television. He is also Special Rapporteur of CMTT/2, dealing with the secondary distribution of digital television and HDTV signals.*

In 1991 Mr. Barbero was awarded the Gold Medal of the International Television Symposium at Montreux for his contributions to digital television transmission, including HDTV, based on discrete-cosine transform techniques.

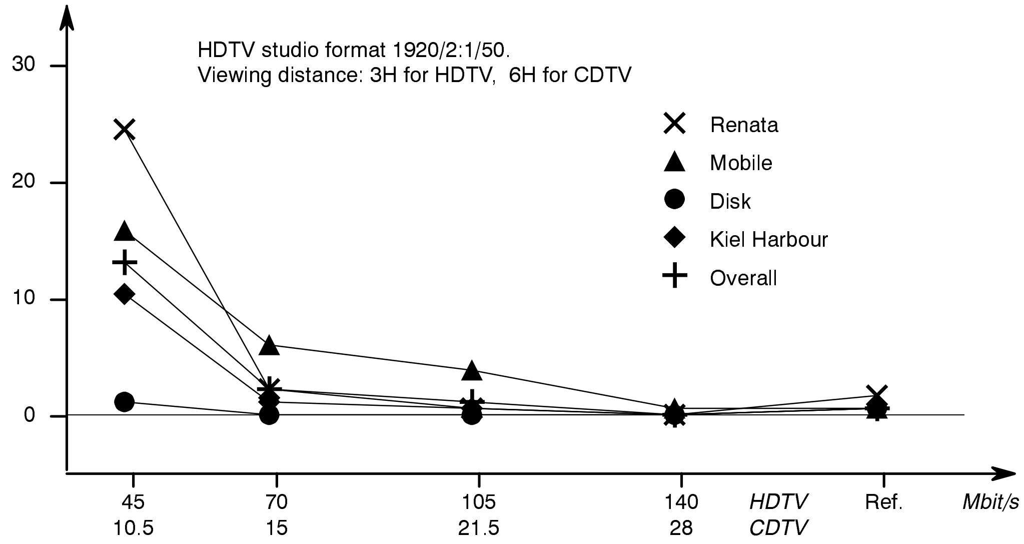


Figure 7
Results of picture quality measurements on the EU256 HDTV codec.

A fallback solution is to carry out the tests in the CDTV domain and to extrapolate the results for HDTV. For this purpose, the CDTV picture is considered as a window of the HDTV picture and the viewing distance is arranged to be three times the height of the virtual HDTV picture, i.e. six times that of the CDTV monitor.

In January 1991, simulations of the final EU256 system, including motion compensation, were carried out by the RAI using CDTV sequences and the resulting quality was evaluated by the RAI and Retevisión [8].

The procedure for the subjective assessments was the double-stimulus continuous-quality scale method described in CCIR Recommendation 500-3. The method is cyclic and the assessor is presented with a series of sequence pairs. Two versions of the same test sequence are presented: one of each pair might or might not contain an impairment. The observers are asked to judge the overall picture quality of the two versions by inserting two marks on a pair of continuous scales on the test report. The scales are 100 mm high and are considered as continuous interval scales, from 0 to 100. Scores are measured for both the target and the reference stimulus of each pair. For each pair, the differences between the reference and the processed sequence are calculated. The difference scores (maximum value 100) indicate the impairment between the co-decoded signal and the reference signal: lower values correspond to higher quality. CCIR Report 1211 indicates 12% as a limit of quality for distribution purposes (25% corresponds to about 1 grade in the CCIR five-grade impairment scale).

The ordinates of *Fig. 7* show the average of the difference scores given by all the observers for each combination of bit-rate and sequence. All the sequences except those for “Renata” were taken from the CCIR library tape described in CCIR Report 1213, which are indicated in Report 1211 as being appropriate for use in the basic quality assessment of codecs for contribution circuits.

The evaluations are for compression ratios ranging from 23 to 7; i.e. from 0.67 to 2.22 bit/pel. The results were obtained for CDTV signals coded with rates from 10.5 to 28 Mbit/s, including the overhead for error protection, audio and auxiliary data at a viewing distance of 6 times the monitor height.

These results can be extended to HDTV if, as noted earlier, the CDTV picture is considered as window of the HDTV screen seen at a distance of 3H. The corresponding bit-rates would range from 45 to 140 Mbit/s for the interlaced format at 50 Hz, with1250 lines/frame and 1920 pels/line.

5. Conclusions

This article has described the methods used in the current generation of European DCT-based HDTV bit-rate-reduction codecs giving high-quality performance at bit-rates between 70 Mbit/s and 140 Mbit/s. Results have been reported giving the picture quality assessment for a range of picture material. These results show that such bit-rate-reduction systems could realistically be used for a high-quality HDTV broadcasting service.

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