An experimental digital consumer HDTV recorder using MC-DCT video compression

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AN EXPERIMENTAL DIGITAL CONSUMER HDTV RECORDER USING MC-DCT VIDEO COMPRESSION

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Abstract — *The concept and real-time implementation of an experimental home-use digital HDTV recorder is presented. An advanced motion-compensated DCT bit-rate reduction system is used to realize high-quality images at 50 Mbit/s only, thereby enabling small recording mechanics for portable applications and featuring a new high-efficiency channel code with embedded tracking information.*

1. Introduction

In the past years, several experimental digital recording systems have been worked out for standard-definition (SDTV) signals [1] at ca. 20 Mbit/s net video bit rate. In this paper¹ we explore the subsequent step, namely digital high-definition video recording (HDTV) for consumer use, and we present an experimental HDTV recorder with a playing time of two hours at a net video bit rate of 50 Mbit/s. The recorder concept, verified by a real-time prototype, is based on small magnetic recording mechanics and advanced bit-rate reduction techniques.

It needs no argument that digital recording is particularly suited for HDTV applications, since it offers intrinsically high time-base stability and flawless reproduction and copying of high-quality images. Bit-rate reduction is a key aspect of a consumer HDTV recorder, since the starting bit rate of HDTV signals, typically 400–500 Mbit/s, is far too high to implement a cost-effective recording mechanism based on a

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video inputs	Y, U, V / 1250, 2:1
intern. sampling Y	54 MHz
sampling U, V	27 MHz
compression unit	2 frames
frame 1 coding	intraframe DCT+VLC
frame 2 coding	motion compensation
HDTV bit rate	50 Mbit/s
audio	2 stereo channels
audio sampling	48 kHz, 16 bits
error correction	R-S product code
channel code	24-25 modulation
recording rate	72 Mbit/s
tracking	embedded tones
scanner	2×2 heads
drum	40 mm
track pitch	10 μm
tape	8 mm, ME

Table 1. Key parameters of HDTV recorder.

small cassette.

Recently, experimental SDTV recorders have been reported using intraframe (or -field) Discrete Cosine Transform (DCT) coding for bit-rate reduction. Borgers *et al.* [2] and Yoneda *et al.* [3] describe DCT coding of 8 × 8 sample blocks with feedback buffer control on 19 and 28 Mbit/s, respectively. The difference in bit rate is explained by the applied sampling standards for YUV signals (3:1:1 versus 4:2:2). Doi *et al.* [4] and Yamamitsu *et al.* [5] reported on a system with feedforward bit-rate control yielding fixed rate on subimage basis, using 19–40 Mbit/s video bit rate. Finally, Onishi *et al.* [6] discussed a more complex system in which 4 consecutive frames of 4:2:2 sampled images were compressed

of the recorder.

The recording channel is based on a special scanner with 40 mm drum and 2×2 heads. In the drum, rotating preamplifiers have been mounted for improving the SNR. The scanner contains no actuator, although the tracking system allows for dynamic track following. The video data are recorded via two parallel channels using 36 Mbit/s recording bit rate for each channel (see Fig. 1, right-hand part). System robustness is improved substantially by using a Reed-Solomon product code for error correction. The code block consists of rows that are equal to sync blocks, while the number of rows is large enough to correct long burst errors. For further channel coding, we have implemented a new high-efficiency DC-free channel code, called 24-25 modulation, to constitute pilot tones for tracking which are embedded in the data.

3. Bit-rate reduction

3.1 System constraints and sampling

The choice for a particular coding algorithm needs to be adjusted to the specific system aspects of a VCR, such as editing, trick-play performance, error robustness and reliability. For the coding architecture, a trade-off should be made between the resulting picture quality and resolution and the hardware complexity. Moreover, the synergy with SDTV cannot be neglected, since this point will be very important in future, both for the consumer, desiring compatibility with existing (SDTV) software, and the manufacturer for reduction of production costs. For this purpose, and to enable the use of small cassettes, a net video bit rate of 50 Mbit/s for HDTV recordings (twice the rate of compressed SDTV) has been adopted. In this way, the channel electronics and scanner can be constructed with acceptable costs for consumer HDTV and an SDTV record-playback mode can be easily realized.

Prior to discussing the compression algorithm, the input sampling will be specified. Table 2 portrays possible sampling standards and indicates the compression ratio needed for a 50 Mbit/s output rate. Two sampling standards –marked with an ‘S’– are particularly interesting, since they enable (theoretically) an HDTV tape format which is very similar to digital SDTV formats.

Sampling frequency	Sampling standard	Comp. ratio	Remark
40.5/13.50	12:4:0 S	6.6	f_s Y $3 \times$ SD
40.5/20.25	12:6:0	7.5	f_s Y $3 \times$ SD
54.0/18.0	16:5.33:0	8.8	f_s Y $4 \times$ SD
54.0/27.0	16:8:0 S	10.0	f_s Y $4 \times$ SD

Table 2. Possible sampling standards for consumer HDTV recording and corresponding compression ratios for 50 Mbit/s net video rate.

The compression ratios indicate that, even if the lowest HD sampling standard of 12:4:0 is adopted, the capabilities of straightforward intraframe coding will be exceeded. Provided that intraframe coding can hardly yield sufficient HDTV quality at 50 Mbit/s, and bearing in mind that there is an improved synergy with current digital video broadcast concepts (e.g. MPEG) when using a more advanced algorithm, the choice for 16:8:0 sampling can be well substantiated, since it yields:

- the best extensions of digital SD formats;
- consistency between the 50- and 60 Hz world;
- sufficient quality gain as compared to SDTV;
- synergy with digital transmission concepts.

3.2 Compression algorithm

The desired output rate of 50 Mbit/s implies a compression factor 10, which can be obtained by a DCT-based coding scheme using interframe coding. Interframe coding using motion-compensated prediction has been proven to be very efficient for transmission applications. However, the special requirements of tape recording restrict the extensive use of interframe coding. Under such conditions, it is a good compromise to apply intra-two-frame coding with predictive coding of every second frame. In addition, the system description shows that our scheme can be extended to an MPEG architecture, thereby enabling flexibility for the future.

Figure 2 portrays an example of a scheme in which only every odd frame is DCT coded directly, while the even frames are skipped and reconstructed by motion-compensated prediction

(a) Static mode.

(b) Motion mode.

	0	1	2	3	4	5	6	7	8
0	11	2	3	4	4	4	5	6	6
1	11	4	5	7	7	8	9	9	10
2	12	5	7	9	10	11	11	12	13
3	13	5	8	10	11	12	13		
4	13	6	9	11	13	13			
5	13	6	10	12					
6	13	7	10	12					
7	13	7	11	13					
8	13	7	11						
9	13	8	12						
10	13	8	13						
11	13	8							
12	14	8							

Table 3: Codeword lengths.

Figure 5: Coefficient scanning patterns in different modes (left and middle) and VLC encoding table (right), showing the word lengths (without sign bit) as a function of the runlength (vertical direction) and the amplitude (horizontal direction).

Figure 2: Encoder block diagram of the HDTV video compression system.

(COMP), using the adjacent frames and the transmitted prediction error. If the switch is in the open position, there is no subtraction and the incoming frame is coded directly with intraframe DCT coding techniques (DCT+Q). Simultaneously at the output, the compressed video is reconstructed by inverse quantization and transformation (DCT+Q⁻¹) and stored in the memory (MEM) of the feedback prediction loop. At the start of the second frame, the switch is closed and the stored reconstructed frame is used to per-

form first motion-estimation (MOTION ESTIM) to calculate the motion vectors, and secondly, apply motion-compensated prediction (COMP) for the new (even) frame. The difference between the prediction and the second frame is then coded, using the same DCT coding scheme. As the compensated prediction is sometimes not reliable due to complex motion (e.g. in the case of scene cuts), the quality of the prediction is analysed (DECIS) to switch to unpredicted (fallback) coding, if required. This decision can be made on block basis. Let us now consider some details of the bit-rate reduction algorithm.

Basically, two prediction schemes can be implemented, namely, either one-sided prediction (previous frame) or two-sided prediction (surrounding frames, see Fig. 3). In both approaches, the motion in frames is estimated by measuring the displacements—denoted by $\vec{D}$ —of objects, so that a good prediction and thus a high coding efficiency is obtained. We have described in [7] that two-sided prediction is more efficient, even if the accuracy is increased to subpixel accuracy in the one-directional case. Considering the importance of a good picture quality for HDTV applications, the two-sided prediction has been implemented, although it requires more processing hardware and larger memory.

The 8×8 sample blocks for DCT coding are constructed within a frame to improve the cod-

Figure 3: Principle of two-sided motion-compensation, using the surrounding frames.

ing efficiency. At moving edges, however, frame-based blocks contain considerable energy on high vertical frequencies. This penalty can be overcome by using a motion detector and subsequently performing an 8×8 DCT in static areas, whereas in the case of motion, a 4×8 field transformation is carried out twice (see [8]). The signal components are merged for subsequent quantization and coding.

The spectral components, i.e. DCT coefficients, are quantized using a 'frequency-dependent' weighting and adaptive uniform quantization. More coarse quantization of the high-frequency components (human eye characteristics) can be accomplished by using a multiplicative weighting factor for each individual coefficient (see e.g. [2]). Additionally, adaptive quantization is employed to exploit the difference in perception between quantization errors in flat areas and structured regions. This adaptivity (4 classes) is implemented by a variable quantizer step size which is controlled by the 'activity' of a block, e.g. the maximum ac coefficient of each individual DCT block. Finally, the quantizer is also controlled by the buffer status (block REG in Figure 2) to obtain a fixed bit cost for every set of two frames (editing).

For further compression of the quantized coefficients, the well-known zig-zag scanning and variable-length coding are performed. As indicated in Figure 5, the scanning of coefficients is different for static (8×8) and 'motion' (two times 4×8) blocks.

Figure 4: VLC output format using a separation of most important and less relevant data. (DC = dc coefficient, AC = ac coefficient, MV = motion vector data, LF = low-frequency coefficients).

The variable-length coding technique is based on zero runlength coding, as considered in the MPEG algorithm. The length of a string of consecutive zero coefficients (the runlength) is counted and is combined with the subsequent non-zero coefficient into one codeword of variable length (see Table 3). Each block is terminated with an EOB codeword. The same technique is applied to coding of the motion-compensated frame.

To increase the error robustness and to enhance VLC signal recovery during the trick modes, a 'self-synchronizing' code (see e.g. [9]) is applied. Taking into account the required throughput rates, the codeword length has been limited to 16 bits.

3.3 Robust output format

When the full capacity of the error-correction system is not available, e.g. during very large burst errors exceeding the correcting capabilities in normal playback or during the trick modes, a robust video format is particularly useful.

The basic features of the VLC output format are best explained by Figure 4. A key aspect of

this format is to separate most important data from less relevant data for pictorial reconstruction. In our case, most important data is always low-frequency (LF) coefficient data, including the dc coefficient (DC) and quantizer control bits in intraframe mode (first frame), and motion vectors in the interframe mode (second frame). Less relevant data contains the remaining high-frequency (HF) coefficient data.

The separation is performed as follows. From every group of DCT blocks we extract the LF AC data as an integer number of bytes, e.g. $N1$ bytes for the first group in the picture. This LF data is then put on fixed positions in the output bit stream. In Figure 4, these positions are at the start of every N bytes in intraframe mode and every M bytes in the interframe mode. The amount of LF bytes is variable, but is uniquely defined by the transmitted quantizer control bits, so that no extra overhead is required.

The fixed positions of the most relevant data enable a high robustness, since this data can always be found, regardless of possible error propagation of the variable-length decoder. Moreover, during trick-mode replay, the LF AC data portions can be used for reconstructing a complete 'low-frequency' image, without the need to resynchronize on the HF AC data stream for variable-length decoding. Basically, a simple byte counter is sufficient to locate the LF AC data. In our implementation, we used two DCT blocks in a group for simplicity, but this format can be extended easily to macroblock basis (6 DCT blocks).

4. Channel coding

4.1 Error-correcting code

The task of the error-correcting code is to protect the compressed digital video signal and the audio signal against random errors, due to the noisy channel, and burst errors resulting from drop-outs, scratches, etc. For this purpose, the well-known Reed-Solomon product code is adopted, consisting of an inner and an outer code [10] [5]. The structure of the data in a track is visualized in Figure 6.

The track is divided into two track parts, each consisting of a video and an audio Error-Correcting Code (ECC) block. The format with the gaps allows for separate editing of the video

Figure 6: Track format of the system, showing the use of product codes for video and audio.

and the audio data. Moreover, it offers the flexibility to use smaller scanners in the future, as the track format can be split into two parts.

The inner ECC, requiring a redundancy of 6 %, is identical for the video and audio signals, and its main task is protection against random errors. The outer code is particularly useful for the correction of burst errors, due to scratches, and requires a redundancy of 8 % for the video part. The outer code for the audio part requires a redundancy of 33 %.

The error protection should not only supply error-free data in normal playback mode, but also deliver as many error-free video blocks as possible in trick modes (i.e. playback speeds that differ from the nominal value). For the trick modes, we distinguish picture search and slow motion. In picture search, the outer ECC cannot be applied, because not all of the inner code blocks related to one product ECC block are received.

In slow motion, however, all data from one product block can be gathered, but the data is not retrieved in a sequential order. This is due

to the multiple scanning of all track parts for such low speeds. As a result, the outer ECC can be activated during slow motion. This is important, because displayed pictures are repeated on the screen, so that errors become more visible.

The performance in trick modes can be investigated and the maximum tracking error for which the quality of the data detection is within the regular boundaries can be determined. An error statistics measurement and evaluation option has been incorporated. One of the results that we have found is that head clogging is a very important source of errors, especially because it seems to be initiated by drop-outs on the tape. This type of error normally leads to burst errors ranging from e.g. 10 % of the track to a few tracks. The kind of head clogging discussed here is self cleaning, but occurs much more often than was expected.

4.2 Digital audio

As is visualized in Figure 6, two separate audio regions are available in the track format, surrounded by edit gaps. This allows for individual editing of the audio information. Each region has sufficient capacity to accommodate a complete stereo channel, having a sampling frequency of 48 kHz and an accuracy of 16 bits. Since they are stored in different sets of 6 tracks, the left and right channel are recorded separately on the tape, thereby allowing for editing on a mono channel basis. Shuffling of the audio data is applied to enhance the error concealment performance. In principle, other sampling standards, such as 44.1 kHz / 16 bits linear or 32 kHz / 12 bits non-linear, can also be recorded. In the latter case, 8 individual audio mono channels are available to the user. This is important for HDTV broadcasts or pre-recorded software with multiple audio channels (multi-lingual). The 48 kHz / 16 bits mode is incorporated in the demonstration model (based on programmable DSPs).

4.3 24-25 modulation

For optimizing the recording channel, we use 24-25 modulation with partial-response detection. The purpose of channel modulation is spectral shaping in order to adapt the bit stream to the

characteristics of the recording channel. This matching refers not only to the detection method, but also to other demands, such as the possibility of Automatic or Dynamic Track Following (ATF and DTF).

For improved bit detection, the modulation imposes constraints on the runlength of bit sequences of equal polarity and it reduces the DC energy of the recording signal. For ATF and DTF, we designed a 24-25 modulation technique with embedded tracking tones, because this is well suited for digital recording [2].

Using the 24-25 modulation, the spectrum of the recording bit stream is shaped such that adequate pilot frequencies are present. During playback, the crosstalk from the pilots in the neighbouring tracks is used as tracking information. The power of the spectral components on and around the pilot frequencies used for both the in-track code signal and the crosstalk from the neighbouring tracks should be minimized for improved tracking detection.

In the playback channel, the well-known Partial-Response Class IV (PR4) equalisation (see [11]) with Viterbi detection is used [12] [13] [14]. In order to prevent error propagation at playback, a technique known as Interleaved NRZI modulation is applied. This is realized by the insertion of a 2T precoder in the record path. As a consequence, a single bit inserted in the total bit stream before precoding, influences the polarity of all the subsequent bits in one of the two interleaved recording bit streams, so that the DC energy can be controlled. If these extra bits are inserted regularly, the spectrum of the recording signal can be adjusted such that it is DC free, contains spectral notches and has pilot tones. In addition, the extra bits from the 24-25 modulation can be used for run-length limiting, due to the interleaved nature of the modulation technique, offering the opportunity to use an almost unique synchronization word.

Because of the byte-based structure of the surrounding error-correcting code, it is advantageous to insert the extra bit between an integer number of bytes, leading to coding schemes like 8-9, 16-17, 24-25, 32-33, etc.. It is important to notice that, with these schemes, the extra code bit occurs alternately in both interleaved streams, so that the polarity of all the bits in the recording bit stream can be influenced.

Figure 7: Code spectrum containing DC-free area, notch and pilot frequency (left to right).

It will be clear that coding tables are not required for the aforementioned mapping. For the choice of the polarity of the extra bit, a 'spectrum analyser' is connected to the output of the 2T precoded bit stream. In this analyser, the desired pilot tone is first subtracted from the signal, and then the spectral power in the recording bit stream at DC and around the pilot frequencies is determined. The extra bit is chosen such that these powers are minimized, resulting in a DC-free spectrum with notches at the pilot frequencies, and a pilot at the desired frequency.

If no pilot is subtracted, a similar spectrum without pilot is obtained. The verification of the runlength constraint (in our implementation maximal 9) is performed separately and it overrules the polarity choice of the analyser, if required.

In the real-time prototype, we opted for three different spectra, indicated by f_0 , f_1 and f_2 , which are applied in subsequent tracks using the order f_0 , f_1 , f_0 , f_2 , f_0 , f_1 , etc.. A spectrum without pilot is indicated by f_0 . The spectra f_1 and f_2 contain a pilot at $f_c/150$ and $f_c/100$, respectively, where f_c stands for the bit clock in the recording bit stream. The spectrum f_2 is given in Figure 7. Because of the adopted two-channel recording system, the pilot f_0 is recorded in channel 1 and the pilots f_1 , f_2 are recorded in channel 2. During playback, only channel 1 is used for

Figure 8: Measurement of track linearity.

the detection of the tracking signal. Because the crosstalk of both pilots is read by the same head, a high robustness against gain variations is obtained.

Decoding of the 24-25 modulation is very simple, since the extra bits only have to be discarded from the detected bit stream. The position of these bits is easily derived from the fixed sync block length, which is a multiple of three bytes (25 bits in the channel). Hence, the extra bits occur in a fixed position in the sync block.

5. Mechanical deck and scanner

The recording system is based on 8mm mechanics with a 40 mm scanner. The analogue video-8 system has an effective track length of roughly 62 mm using 180° wrap angle and a track width of 34 μm (50 Hz environment). To improve the (digital) recording density, the recorded track width is reduced to 10 μm, using guardbandless azimuth recording.

The relatively high recording rate of 72 Mbit/s is processed by a two-channel system, resulting in a scanner with four record-playback heads. Metal-evaporated (ME) tapes with newly developed TSS heads enable a high linear recording density.

5.1 Mechanical deck

The aforementioned density requires an improved mechanical stability and accuracy, so that the deck was modified to obtain sufficient stiff-

ness and robustness. Head window and scanner surface were modified to improve the head-tape contact at the increased scanner speed of 4500 rpm. After carefully checking the tape path and the tape tension, we achieved a typical track linearity as shown in Figure 8.

Writing two 10 μm tracks in parallel without guardband, results in a tape velocity of 3.5 cm/s. The nominal playing time for a 90-min. ME analogue video-8 cassette is roughly 60 minutes. However, as the trend in tape technology is towards thinner tapes, the playing time can gradually increase to 2 hours.

The deck and servo system support a picture search mode and slow motion. By using a programmable system for servo functions (see Section 5.3), the tape speed can be controlled with high accuracy and flexibility. For preliminary experiments, a search mode of 3–3.5 times the normal tape speed has been implemented in both directions, while slow motion is possible at 1/3 of the normal tape speed, both in forward and backward mode.

Although longitudinal tracks can be omitted, due to the ATF information (pilot tones) in the helical tracks, a control track at the bottom tape edge was implemented for experimental purposes at the initial stage when the tones were absent. The ergonomic system design enables the user to control the tape recorder in a convenient way by means of an LCD-display and soft keys.

5.2 Scanner

The scanner assembly consists of the stationary lower drum with a small sensorless and brushless motor and the rotating upper drum. Highly reliable ball bearings enable stable rotation of the upper drum. Between the two drum parts, a two-channel rotary wide-band transformer is placed for data transfer. The active playback circuitry is located inside a shielded cylindrical part of the rotating upper drum. The angle between the heads of the same head pair is 8° . Figure 9 portrays the construction of the scanner system.

The preamplifier circuitry has been mounted in the rotating upper drum to realize short connections between heads and preamplifier. A second advantage is to have no transformer leakage inductance and stray capacitance on the input of

Figure 9: Construction diagram of scanner with rotating preamplifier.

the amplifier. A third benefit, resulting from the shielded and grounded drum, is the protection of the amplifier against RF-interference from outside.

The amplifier has been implemented with SMD technology. The record/playback-switching circuitry is based on a small special relay device. The relays are mounted on the circuit board in such a way that the centrifugal forces from the rotating drum do not influence the relay-contact function. Playback head-switching circuitry on the drum makes it possible to use only one two-channel transformer for 4 video heads. The power supply and head-switch pulse transfer are carried out by an extra transformer on top of the scanner. The power supply is activated during the playback mode. The block diagram of the active playback circuitry is shown in Figure 10.

Figure 10: Block diagram of scanner system.

5.3 Servo system

The servo system has been designed around a high-speed microcomputer (see Fig. 11), which controls all main mechadeck functions, communication, keyboard, display, and it performs the regulation control for the scanner and tape transport systems. The software-based design is perfectly suited for the fast transfer of parameters between related regulation systems; this is especially required in the implemented trick modes, ranging from slow motion up to 8 times search speed in forward and backward direction.

A new algorithm for the dynamic compensation of mechanical and magnetical tolerances inside the scanner speed feedback generator reduces the total scanner jitter with the tape in contact within a margin of $3 \mu s$. The scanner phase control is active on a video frame basis, so that the recorder quickly responds to video source switching. The tape transport system is equipped with a closed frequency-loop system and an ATF-phase control in playback mode, which guarantees fast tracking lock conditions, combined with the lowest possible tracking errors. Although the tape transport system is designed to lock on one particular ATF pattern out of six equal patterns resulting from the multi-track format (24 tracks / 2 frames), the system locks on the first valid ATF block to improve synchronization response time. The variable starting point for video decoding is compensated by the

Figure 11: Architecture of servo system.

use of a large buffer memory.

An RS-232 bus supports communication between the servo system and external electronics, and allows to check and analyze various parameters, such as the deck status, tape counter, or equalizing setting. The system incorporates a monochrome LCD-display with 240×60 dots to monitor status, user commands, icons, and on request, 'hidden' parameters, like head switching delay and motor parameters for technical services.

6. Conclusions

The concept and real-time implementation of an experimental home-use digital HDTV recorder has been presented. The system is based on small (modified 8mm) recording mechanics and cassette and it uses advanced bit-rate reduction. The bit rate is sufficiently low (50 Mbit/s) for a cost-effective implementation of both HDTV and SDTV recording on the same mechanics.

The bit-rate reduction algorithm is based on advanced motion-compensated DCT coding and is organized in such a way that it can be extended to MPEG applications, although a number of measures have been included to support and enhance editing (fixed compression in two

frames) and trick-mode replay (special output formatting).

The error-correcting code is based on two product code blocks on one track to have flexibility for other future tape formats. The newly developed 24-25 channel code, featuring digitally embedded pilot tones for tracking, has a high efficiency (only 4 % overhead) and low complexity (no tables), while it enables sufficient monitoring capability of the tracking tones during playback (and DTF possibility).

The scanner was optimized for the high channel rates, using 2×2 heads, and a rotating preamplifier has contributed to an improved signal-to-noise ratio at the bit-detection side during playback. A software-based servo design allows for high flexibility and improved accuracy in the recording system control.

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