IEEE Transactions on Consumer Electronics, Vol. 41, No. 1, FEBRUARY 1995

CHANNEL CODE WITH EMBEDDED PILOT TRACKING TONES FOR DVCR

Joost A. H. Kahlman and Kees A. Schouhamer Immink
Philips Research Laboratories
5656 AA Eindhoven, The Netherlands

Summary - *The proper functioning of a digital video recorder is largely governed by the selection of a suitable channel code in conjunction with both the detection method and the tracking system. The channel code for the Digital Video Cassette Recorder (DVCR) satisfies a variety of design requirements. The paper provides an overview of those requirements. A detailed description is given of the construction of the new channel code, called 24→25 code, that complies with the given constraints and involves only a minor drawback in terms of the overhead needed. The servo position information is recorded as low-frequency components, pilot tracking tones, which are embedded in the recorded stream of binary digits.*

Key Words: digital video recorder, channel code, equalization, pilot tracking tones, spectral null codes

I. INTRODUCTION

Dynamic track-following mechanisms are at present used in consumer-type video or digital audio tape recorders aiming to alleviate the mechanical inaccuracies of the recorder. There are numerous techniques with which the read head can measure its position with respect to the written data-track. In general terms, the basic approaches currently in use for the allocation of servo information can be divided into two main categories, which depend on the allowable complexity of the system:

- time-multiplex technique,
- frequency-multiplex technique.

In the time-multiplex technique, the signals representing the user information and the signals required for the servo systems are entirely separated in their respective places on the track. A name commonly used for this technique is *sectorized servo tracking.* The servo position information is a sampled signal and therefore affects the maximum attainable servo bandwidth. A compromise must be sought between the amount of servo position information and the performance of the servo system. As an example, we mention the format employed in the R-DAT digital audio recorder [1]. The frequency-multiplex technique, on the other hand, partitions, as its name suggests, the user and servo position information into separate frequency bands. The latter technique has the obvious advantage over the time-multiplex technique that it provides continuous information rather than sampled information. Often, the servo position information is recorded as low-frequency components, usually called *pilot tracking tones* [2]. The principle of operation is as follows. On even tracks the pilot tone has a frequency f_1 and on odd tracks the pilot tone frequency is f_2. Servo position information is developed from the read-back signals by subtracting the amplitude of the f_2 component from the f_1 component. These two components can be separated with band filters since they differ in frequency. As the head moves off track in one direction, the amplitude of one component decreases while the amplitude of the other increases. On the basis of the difference error signal and the control technique employed, this then instructs the control mechanism. In other words, by observing the difference between the pilot tone amplitudes of adjacent tracks we can tell whether we are moving off-track to the right or to the left. To circumvent interaction between the pilot tones and the user information, user information has to be encoded in such a way that the power spectral density function of the encoded stream vanishes at the pilot tone frequencies [3].

Pilot tracking tones are used to derive position reference information employed by a head-positioning servo system to position and maintain the head precisely over a selected track in the DVCR. Experiments have shown that adding analog pilot tones to the write current will result in serious interference between the digital data and the tones. Furthermore, the precise amplitude of an analog pilot tone depends on the characteristics of the head-tape combination, which means careful adjustment of the write current is necessary for each of the heads. This fact precludes the technique, of e.g. simply adding a sinusoidal waveform to the binary data The only option available is to embed the tones in the recorded binary sequence. A major challenge encountered during the system conception was the design and evaluation of the channel code. In a prior art video recorder [4] a channel code of rate 8/10 was used. It featured, for example, embedded pilot tracking tones, unique sync words, and freedom of low-frequency components. However, the redundancy of 25 percent of the above rate 8/10 code was considered to be a serious drawback. We investigated the feasibility of

Contributed Paper
Manuscript received December 27, 1994

an alternative code with similar features as the rate 8/10 code while imposing a much smaller redundancy. We succeeded in finding a new channel code of rate 24/25, called 24 →25 code. Naturally, in a number of aspects the new code does not perform as well as the prior art rate 8/10 code, but ample experimental evidence has shown that it performs sufficiently in the environment of the digital video recorder at hand [5].

The paper outlines the details of the design considerations of this highly efficient channel code and its implementation.

II. CHANNEL CODE DESIGN CONSIDERATIONS

A critical part of a high-density digital recorder is the channel code in conjunction with both the detection method and the tracking system. In order to achieve a robust recording system, much emphasis has been put on this subject. The following considerations formed the framework of the channel code design process:

- 1. Digitally-embedded pilot tones in order to make Automatic Track Following (ATF) or Dynamic Track Following (DTF) possible.
- 2. Limitation of the maximum runlength at the detection point and low-frequency content of the channel code. Both clock recovery and amplitude detection will improve with an increasing number of write-current transitions. Poor low-frequency response of the rotary transformer will result in baseline wander, which may disturb the recording process, especially when metal-powder tapes are used. Overwriting (e.g. perfect insert, audio dubbing) is easier when the ratio of maximum and minimum runlength is decreased.
- 3. The loss of signal-to-noise (SNR) ratio at the detection point due to channel coding must be as small as possible. Computations [6] and experiments have shown that the rate R of the code should be chosen to be as large as possible. When a runlength-limited code [6] is used, relatively complex receiver structures (decision feedback equalizer in conjunction with Viterbi detection and noise prediction) may be needed. Such a receiver, however, with its mutually dependent compensation methods for both noise and Inter-Symbol-Interference, is not sufficiently robust due to the very narrow operation area in which the SNR is maximal. Investigations into several channel codes and detection methods revealed that the consumer-type magnetic helical scan recording channel, with its specific demands with respect to receiver acquisition time (head switch over, audio dubbing and trick modes) operating at low SNR value operates best with high-rate codes.
- 4. The detection method has to be maximally insensitive to low-frequency interference arising from for example crosstalk. This item becomes more and more critical when narrower tracks are used. For reasons of tracking margin and power consumption (no erase head), it may be advantageous to use heads with over-width. Then the track pitch is smaller than the width of the playback head, which will result in additional (low-frequency) crosstalk. When trick (or shuttle) modes are done without a head actuator, the playback head can only read correctly in very short periods of time. This means that the number of bits received correctly depends mainly on how the receiver copes with crosstalk.
- 5. Special attention must be given to the design of the synchronization word, as the system performance at poor signal-to-noise ratios (e.g. during trick modes) depends critically on it. The synchronization word should be (sufficiently) unique, so that the sync word detector can discriminate between regular data bits and the sync word.

In the next section, we will discuss the specification of the new code that complies with the requirements.

III. SPECIFICATIONS OF THE 24 →25 CHANNEL CODE

The system requirements mentioned above and massive recording experiments resulted in the 24 →25 channel code. The main parameters of this code can be summarized as follows:

- Rate = 24/25.
- Minimum runlength = 1.
- Maximum runlength = 9. This maximum runlength feature is not guaranteed, but the probability of occurrence of runlengths larger than 9 is smaller than 5 times 10^{-5}.
- Precoding is done to make Partial Response (PR4) detection possible.
- Limited low-frequency response (perfectly DC-free, -3 dB at $f \leq f_b/500$).
- Provisions for tracking: (channel bit rate 21 Mbit/s)
  - Pilot tone $f_1 = f_b/90$ (230 kHz), SNR $\geq$ 20 dB in 300 Hz.
  - Pilot tone $f_2 = f_b/60$ (350 kHz), SNR $\geq$ 20 dB in 300 Hz.
  - Notches around the pilot tones f_1 and f_2 stabilize the amplitude.
  - Notch depth $\geq$ 5 dB (resolution bandwidth 300 Hz). Amplitude stability within 0.3 dB peak-to-peak. Notches at frequencies f_1 and f_2 improve the tracking SNR by reducing the in-track code noise.

TABLE I

Definition of types of heads

type	*notches*	*pilot*
F0	DC, f_1, f_2	No pilot tone
F1	DC, f_1, f_2	f_1
F2	DC, f_1, f_2	f_2

TABLE II

Configuration of heads and spectra

tracks	F0	F1	F0	F2	F0	F1
head number	0	1	2	3	0	1

- Almost unique sync word. The probability of generating a false sync word in the normal data stream is lower than the probability of occurrence of channel drop outs (1×10^{-6}).
- No coding/decoding tables are needed.

IV. TRACKING FORMAT

The recorder system uses two pairs of heads that are diametrically positioned with respect to one another on the drum. The tracking format requires three types of tracks with different frequency spectra. The track types, F0, F1, and F2, are listed in Table I. The footprint for the proposed scanner configuration is shown in Table II. It can be seen that each head pair has one head of type F0 (no pilot tone), while the second head is alternately of type F1 or F2. During the reading of the F0-type of tracks, the crosstalk is measured from the neighboring tracks which are of type F1 or F2. As explained in the Introduction, servo position information is developed by subtracting the amplitude of the f_1 component from the f_2 component. These two components can be separated with band filters since they differ in frequency. As the head moves off track in one direction, the amplitude of one component decreases while the amplitude of the other increases. The difference error signal is used for steering the actuator on which the head pairs are mounted. The spectral notches are required to cancel any interference between the written data and the servo detection system.

V. DETECTION METHOD

The detection method used is Partial Response Class4 (PR4). The transfer function of PR4 detection can be modelled by

$$1 - D^2, \tag{1}$$

where D denotes the time interval of a channel bit cell. Prior to recording, the original bit stream is transformed by a filter with a response having the inverse of (1), that

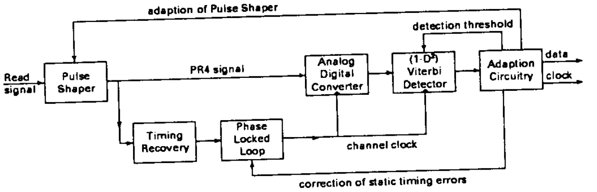


Fig. 1. Block diagram of the receiver. The read signal is, after pulse shaping, forwarded to both the clock recovery and an AD-converter. The AD-converter digitizes the input signal. The digital signal is applied to the $1-D^2$ Viterbi detector whose thresholds are governed by an adaptive circuit.

is,

$$\frac{1}{1 \oplus D^2},$$

where $\oplus$ denotes modulo-2 addition. The filter is called a *2T-precoder*. Figure 1 shows a schematic block diagram of the receiver. The PR4 signal, obtained at the output of the pulse shaper, is synchronously sampled and detected according to the Viterbi algorithm. From the Viterbi detector, control signals are derived for adaptation of the detection threshold and the transfer function of the pulse shaper. For reasons of acquisition time and stability, the bit detection and the timing recovery are separated (off-line timing recovery).

VI. DESCRIPTION OF THE BASIC 24 →25 ENCODING PROCESS

A control bit is stuffed in each sequence of 24 successive input data bits to create the freedom to shape the spectrum of the encoded bit stream. By setting this control bit to '0' or '1', the polarity of 25 interleaved bits at the output of the 2T-precoder can be chosen. The sender opts for that value of the control bit that minimizes the accumulated signal power of the transmitted code words at three predefined frequencies, namely DC, f_1, and f_2. Furthermore, subtraction of the desired pilot tone before measurement of the power at the pilot tone frequencies, the pilot tone will appear at the output of the 2T-precoder. Should the maximum runlength on the channel become more than the predefined maximum, i.e. nine, the above spectral optimization is overruled.

The encoding process is detailed by focusing on the situation in type-F1 tracks. Type-F1 tracks, see Table I, have spectral notches at f_0, (=DC), f_1, and f_2 plus a pilot tone at frequency f_1. It is assumed that the input data

$$(\ldots, x_{-24}, \ldots, x_{-1}, x_{-1}, \ldots x_{24}, x_{26}, \ldots, x_{49}, \ldots)$$

are translated into channel bits

$$(\ldots, z_{-25}, \ldots, z_{-1}, z_0, \ldots, z_{24}, z_{25}, \ldots, z_{49}, \ldots)$$

that are written serially on the tape. The encoding process is performed in six steps:

1. Partition the input data into groups of 24 bits (3 bytes):

$$\mathbf{x} = (x_1, x_2, \ldots, x_{24}), x_i \in \{0,1\},$$

and define $\mathbf{x}_0$ and $\mathbf{x}_1$ by

$$\begin{aligned} \mathbf{x}_0 &= (0, x_1, x_2, \ldots, x_{24}) \\ \mathbf{x}_1 &= (1, x_1, x_2, \ldots, x_{24}). \end{aligned}$$

2. The candidate input words, $\mathbf{x}_0$ and $\mathbf{x}_1$, are 2T-precoded to obtain the candidate code words $\mathbf{y}_0$ and $\mathbf{y}_1$:

$$\begin{aligned} \mathbf{y}_0 &= (y_{0,0}, y_{0,1}, \ldots, y_{0,24}) \\ \mathbf{y}_1 &= (y_{1,0}, y_{1,1}, \ldots, y_{1,24}), y_{h,i} \in \{0,1\}. \end{aligned}$$

The 2T-precoder is initialized by (z_{-2}, z_{-1}), that is, the last two bits of the previously transmitted code word $\mathbf{z} = (z_{-25}, z_{-24}, \ldots, z_{-2}, z_{-1})$. The following recursive equations describe the 2T-precoder response:

$$\begin{aligned} y_{0,0} &= z_{-2} \\ y_{0,1} &= z_{-1} \oplus x_1 \\ y_{0,i} &= y_{0,i-2} \oplus x_i, 2 \le i \le 24 \\ y_{1,0} &= \bar{z}_{-2} \\ y_{1,1} &= z_{-1} \oplus x_1 \\ y_{1,i} &= y_{1,i-2} \oplus x_i, 2 \le i \le 24. \end{aligned}$$

where $\oplus$ denotes modulo-2 addition.

3. In our experiments we opted for pilot tones with a square-wave shape of frequency f_x and amplitude A, or

$$\begin{aligned} \mathrm{Tone}_{x,i} &= A, \quad i \text{ Mod } 1/\mathrm{f_x} < 1/2f_x \\ &= -A, \quad i \text{ Mod } 1/\mathrm{f_x} \ge 1/2f_x. \end{aligned}$$

4. Calculate $P_{0,s}$ and $P_{1,s}$, where $P_{0,s}$ and $P_{1,s}$ are defined as the accumulated signal power of the output signal at frequencies f_0, f_1, and f_2 when it is assumed that $\mathbf{y}_0$ or $\mathbf{y}_1$ is transmitted. $P_{0,s}$ can be expressed as

$$\sum_{k=0}^{2} \left| \sum_{i=-\infty}^{s-1} (\tilde{z}_i - Tone_{1,i}) e^{j2\pi i f_k} + \sum_{i=s}^{s+24} (\tilde{y}_{0,i} - Tone_{1,i}) e^{j2\pi i f_k} \right|^2,$$

where $j = \sqrt{-1}$, $f_0 = 0, f_1 = 1/90$, and $f_2 = 1/60$. The parameters $\tilde{y}_{h,i} = 2y_{h,i} - 1 \in \{-1,1\}$ and $\tilde{z}_i = 2z_i - 1 \in \{-1,1\}$ denote the bipolar versions of $y_{h,i}$ and z_i. Similarly, the accumulated power for candidate y_1, $P_{1,s}$, is computed.

5. Determine the maximum runlength that occurs as a result of the concatenation of the candidate words with the previously transmitted information. The maximum runlengths are denoted by RL_0 and RL_1. If both RL_0 and RL_1 are less than the predefined maximum, then transmit that candidate code word that minimizes the power.

6. The following two rules are applied to choose between candidate code words when RL_0 or RL_1 violate the predefined maximum

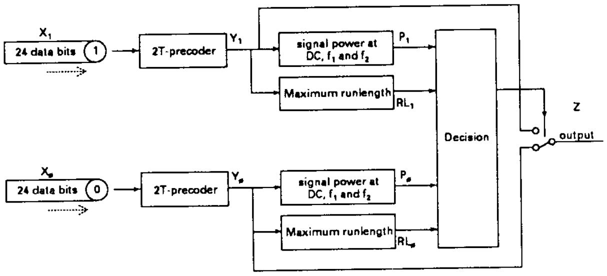


Fig. 2. Block diagram of the encoder. The two candidate input words $\mathbf{x}_0$ and $\mathbf{x}_1$ are forwarded to a 2T-precoder. For each candidate the signal power is computed at Dc, f_1, and f_2. The decision circuitry opts for the best candidate.

- if ($RL_0 \le 9$ and $RL_1 \le 9$) or ($RL_0 > 9$ and $RL_1 > 9$) then transmit that candidate code word that minimizes the power.
- if $RL_0 > 9$ then transmit $\mathbf{y}_1$. If $RL_1 > 9$ then transmit $\mathbf{y}_0$.

This concludes the encoding process.

A. Sync word generation

The syncword consists of 15 bits preceded by two control bits. The control bits are chosen to be such that the maximum runlength constraint is violated. This property prevents the risk the presence of false sync words in regular data. The encoding process is changed as follows:

1. Take 8 bits from the data source and define source word

$$\mathbf{x} = (011000000001101, x_{17}, x_{18}, \ldots, x_{24}).$$

2. Compose candidate input words $\mathbf{x}_0$ and $\mathbf{x}_1$

$$\begin{aligned} \mathbf{x}_0 &= (z_{-2}, z_{-1}, 011000000001101, x_{17}, \ldots, x_{24}) \\ \mathbf{x}_1 &= (\bar{z}_{-2}, \bar{z}_{-1}, 011000000001101, x_{17}, \ldots, x_{24}). \end{aligned}$$

3. 2T-precode candidate input words $\mathbf{x}_0$ and $\mathbf{x}_1$ to obtain candidate code words $\mathbf{y}_0$ and $\mathbf{y}_1$

$$\begin{aligned} \mathbf{y}_0 &= (00011111111110001, y_{0,17}, \ldots, y_{0,24}) \\ \mathbf{y}_1 &= (11100000000001110, y_{1,17}, \ldots, y_{1,24}). \end{aligned}$$

B. Realization of the basic 24 →25 encoding process

Figure 2 shows a block diagram of the encoder. The two candidate input words $\mathbf{x}_0$ and $\mathbf{x}_1$ are processed in parallel. The circuitry computes the corresponding candidate code words $\mathbf{y}_0$ and $\mathbf{y}_1$, the accumulated signal power P_0 and P_1, and the maximum runlengths RL_0 and RL_1. Subsequently, the decision circuit opts for the optimum alternative for code word $\mathbf{z}$. Figure 3 shows the set-up of the "spectrum analyzer" $\mathrm{P}_{0,s}$. Tone_1 is subtracted from

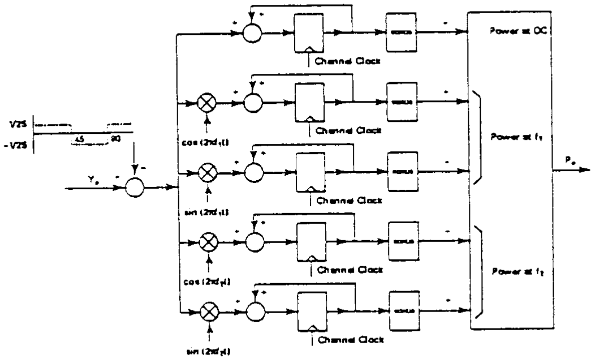


Fig. 3. The set-up of the "spectrum analyzers". The circuitry computes the quadratic sum of the powers at DC, f_1 , and f_2.

candidate code word $\mathbf{y}_0$, before five multiply/accumulate operations take place, namely two for each of the pilot tones plus one for DC. $\mathrm{P}_{0,s}$ is obtained after quadratic summing of the results of 25 multiply/accumulate steps performed on candidate code word $\mathbf{y}_0$. Before the calculation is started the integrators are updated. The 2T-precoders are preset to (z_{-2}, z_{-1}), that is, the two last bits of the previous transmitted code word. When in the previous encoding step, for instance, candidate codeword $\mathbf{y}_1$ has been found to be the optimum code word, the contents of the integrators in the analyzer with output $\mathrm{P}_{1,s}$ are copied to the corresponding integrators in analyzer $\mathrm{P}_{0,s}$.

C. Look-ahead encoding

The interleaving character of the 2T-precoder and the oddly numbered distance between the polarity bits increase the influence of the polarity bits to a range of 50 channel bits. The first bit in the candidate input words only affects 13 bits in the corresponding candidate code words $\mathbf{y}_0$ and $\mathbf{y}_1$. The remaining 12 bits have already been determined by the previous transmitted code word $\mathbf{z}$. More precisely, the oddly numbered bits $(y_{h,1}, y_{h,3}, \ldots, y_{h,23})$ are only determined by (z_{-1}) and $(x_1, x_3, \ldots, x_{23})$. Because one half of the channel bits is not controlled by the encoding process, the SNR of the pilot tones and the notches is not maximal. This drawback can be alleviated by taking into account the complete effect of the first bit in the candidate input words during the evaluation. A look-ahead method optimizes the effect of four different candidate input words

$$\begin{aligned}
\mathbf{x}_0 &= (0, x_1, \ldots, x_{24}, 0, x_{26}, \ldots, x_{49}),\\
\mathbf{x}_1 &= (0, x_1, \ldots, x_{24}, 0, x_{26}, \ldots, x_{49}),\\
\mathbf{x}_2 &= (0, x_1, \ldots, x_{24}, 0, x_{26}, \ldots, x_{49}),\\
\mathbf{x}_3 &= (0, x_1, \ldots, x_{24}, 0, x_{26}, \ldots, x_{49}).
\end{aligned}$$

The candidate code words $\mathbf{y}_0, \ldots, \mathbf{y}_3$ are obtained by 2T-precoding the four candidate input words

$$\mathbf{y}_h = (y_{h,0}, y_{h,1}, \ldots, y_{h,49}).$$

For a type-F1 track, the accumulated signal power $\mathrm{P}_{h,s}$ is given by

$$\sum_{k=0}^{2}\left|\sum_{i=-\infty}^{s-1}(\tilde{z}_i\text{-}Tone_{1,i})e^{j2\pi i f_k} + \sum_{i=s}^{s+49}(\tilde{y}_{h,i}\text{-}Tone_{1,i})e^{j2\pi i f_k}\right|^2$$

The method described above is used to compute $\min\{P_{0,s}, P_{1,s}, P_{2,s}, P_{3,s}\}$, with which it should be appreciated that only the first 25 bits of the candidate code word $\mathbf{y}_h = (y_{h,0}, \ldots, y_{h,24})$ are transmitted. In other words, only a tentative decision has been taken regarding $\mathbf{y}_h = (y_{h,25}, \ldots, y_{h,49})$. The spectra plotted in Figure 4 were obtained in computer simulations of the above algorithm. As a final remark, it should be noted that the encoder can be programmed to bar sequences of the type '1010...1010', which could be vexatious for a PR4 receiver. This can only be done by sacrificing a longer maximum runlength.

VII. CONCLUSIONS

We have given an overview of the system requirements of a consumer-type digital video recorder. A detailed description has been given of the construction of the new channel code, called 24 →25 code, that complies with the given constraints and involves only a minor drawback in terms of overhead needed. The servo position information is recorded as low-frequency components, pilot tracking tones, which are embedded in the recorded stream of binary digits.

REFERENCES

[1] A. Hitomi and T. Taki, 'Servo Technology of R-DAT', *IEEE Trans. Consumer Electr.*, vol. CE-32, pp. 425-432, Aug. 1986.

[2] K.A.S. Immink, 'Signal-to-Noise Ratio of Pilot Tracking Tones Embedded in Binary Coded Signals', *IEEE Trans. Magn.*, vol. MAG-24, no. 2, pp. 2004-2009, March 1988.

[3] B.H. Marcus and P.H. Siegel, 'On Codes with Spectral Nulls at Rational Submultiples of the Symbol Frequency', *IEEE Trans. Inform. Theory*, vol. IT-33, no. 4, pp. 557-568, July 1987.

[4] S.M.C. Borgers, E. de Niet, and P.H.N de With, 'An Experimental Digital VCR with 40mm Drum, Single Actuator, and DCT-based Bit-Rate Reduction', *IEEE Trans. Consumer Electr.*, vol. CE-34, pp. 597-605, Aug. 1988.

[5] P.H.N. de With, A.M.A. Rijckaert, H.-W. Keesen, J. Kaaden, and C. Opelt, 'An Experimental Digital Consumer HDTV Recorder Using MC-DCT Compression', *IEEE Trans. Consumer Electr.*, vol. 39, pp. 711-722, Nov. 1993.

[6] K.A.S. Immink, 'Coding Techniques for the Noisy Magnetic Recording Channel: A State-of-the-Art Report', *IEEE Trans. Commun.*, vol. COM-37, pp. 413-419, no. 5, May 1989.

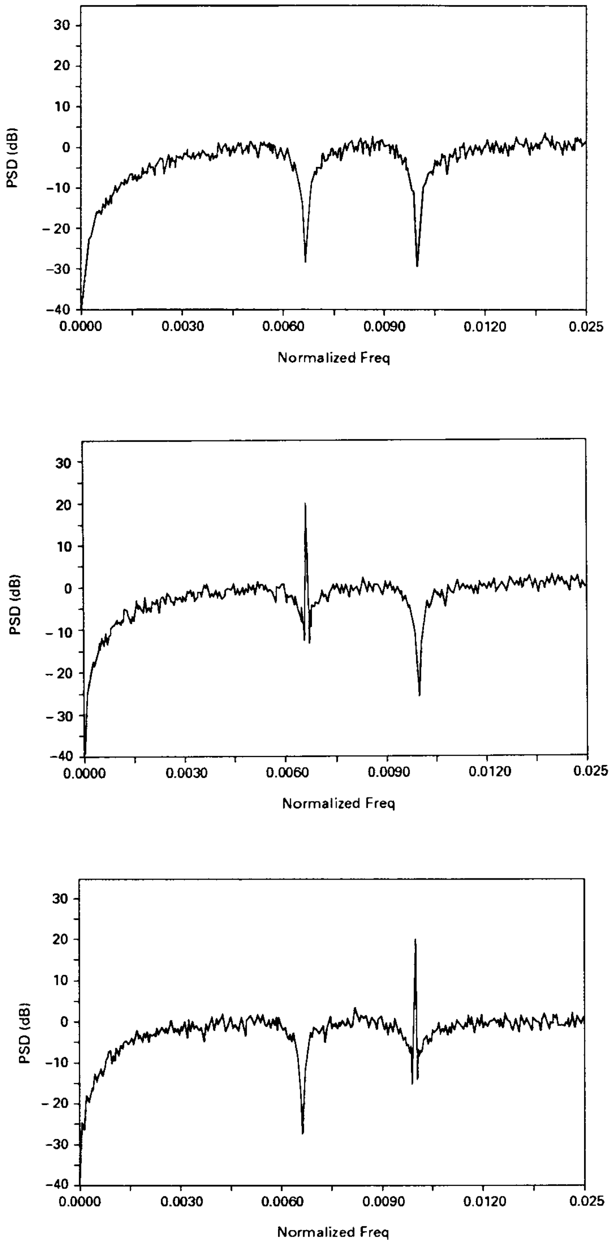
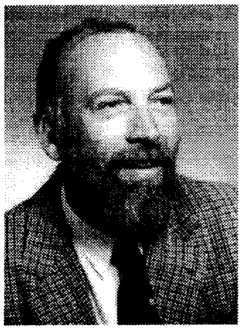


Fig. 4. Power spectral density (PSD) of recorded patterns. The upper curve shows the F0-type spectrum, the middle plot shows the F1-type spectrum, and the lower curve shows the F2-type spectrum

BIOGRAPHIES

Joost Kahlman was born in Tilburg, The Netherlands, in 1956. He was awarded the B.Sc. degree in electronics from the Eindhoven Polytechnic in 1979. After serving the Royal Dutch Army, he joined the Optics group of the Philips Research Laboratories in 1980. He contributed among others to Compact Disc and Compact Disc Video. From 1985 to 1992 he was involved in digital magnetic recording, such as DVCR. In 1992 he returned to the optics group, where he is currently involved in optical tape storage. He holds various patents in the recording field.



Kees A. Schouhamer Immink (M'81-SM'86-F'90) received the M.S. and Ph.D degrees from the Eindhoven University of Technology in 1974 and 1985. Immink joined the Philips' Research Laboratories in Eindhoven in 1968. He contributed to the design and development of coding techniques for the Compact Disc, Compact Disc Video, R-DAT, and DCC recorders. Immink holds more than thirty patents and has written numerous papers in the field of coding techniques for optical and magnetic recorders. He is author of the monograph *Coding Techniques for Digital Recorders* and co-author of *Principles of Optical Disc Systems* and *Reed Solomon Codes: Theory and Application.* He was elected a Fellow of the AES, IEE, and IEEE; furthermore he received the AES Silver Medal in 1992, the IEE Sir J.J. Thomson Medal in 1993, and the SMPTE Poniatoff Gold Medal for technical excellence in 1994.