

REPORT 953-2*

**DIGITAL CODING FOR THE EMISSION OF HIGH-QUALITY
SOUND SIGNALS IN SATELLITE BROADCASTING
(15 kHz NOMINAL BANDWIDTH)**

(Question 2/10 and 11, Study Programmes 51C/10, 2F/10 and 11)

(1982-1986-1990)

1. Introduction

Technological developments presently allow the emission of digitally encoded audio signals to the public to be envisaged. It would be possible in particular to introduce this technique in the broadcasting-satellite service. This will require the prior definition of characteristics concerning coding, multiplexing and modulation of audio signals. This Report is concerned only with digital sound coding. Multiplexing, bit-error protection strategy and modulation aspects are discussed in Reports 954 and 632 respectively. Information dealing with methods of error correction or concealment can be found in Report 1073.

Study Programme 51C/10 considers that the introduction of digital techniques for emission should allow an improvement in the quality of the signals transmitted. Recommendation 651 deals with digital PCM coding. Results of objective and subjective tests are given in Annex II of the present Report.

Other coding methods leading to a substantial bit rate reduction and at the same time maintaining very high quality are described in Report 1199.

In addition to the improvement of quality, the following should be considered:

- the compromises between quality objectives and bit rate may be different for sound services which may have various quality requirements and planning constraints; they may also vary according to the requirements of individual countries;
- there are clear advantages for broadcasters, receiver manufacturers and the public in using a single standard for each application.

2. Digital PCM coding

2.1 Linear coding

The audio signal is encoded into digital form by a high precision analogue-to-digital converter after being passed through an appropriate anti-aliasing filter and possibly through a pre-emphasis network. The uniform coding resulting from this process leads to a linear representation with a minimum of 14 bits per sample. In that case, a 2's complement coding is used for each sample. A possible scale factor can possibly be associated with blocks of successive samples.

2.2 Floating point or near-instantaneous coding

A 16 bit linearly encoded sound signal can be transmitted on 14 bits per sample using a floating point coding system where the scale factor is based on 64 consecutive samples. Pre-emphasis may or may not be used.

When a lower bit rate per audio channel is required, the near-instantaneous companding allows for a reduction in the number of bits per sample from 14 bits to 10 bits. This companding is applied on blocks of 32 successive samples with a complementary scale factor on five ranges. A 2's complement representation is still used for each sample. In that case, the use of pre-emphasis is recommended to reduce the programme modulated noise.

* This Report should be brought to the attention of the CMTT.

2.3 Differential coding

Differential coding is another technique to achieve a lower bit rate per audio channel by making use of the fact that, in most programme material, correlation between two consecutive samples is high. In this case, the difference between successive samples will generally be smaller than the actual sample values. By coding the differential signal, rather than the original samples, a reduction of bit rate can be achieved for a given requirement of signal-to-noise ratio.

Difference samples can be coded instantaneously, one at a time, as in a linear coder, or by blocks of successive samples (with complementary scale factors) as in near-instantaneous companding. Such a marriage of differential coding with near-instantaneous companding has been proposed [CCIR, 1986-90] to achieve a reduction in the number of bits per sample from 15 bits to 8 bits. The use of pre-emphasis is recommended to reduce the programme modulated noise.

2.4 PCM emphasis

The quality of the transmission system, in particular the noise and distortion characteristics, depends largely on the signal statistics and the coding law used. Also, the audible noise detection response of the receiving system, including the human ear, can be considered to be non-uniform. Taking these factors into account, the characteristic could be improved by adopting appropriate pre-emphasis prior to the coding process and the corresponding de-emphasis at the receiver following the decoding process. For PCM systems two such emphasis methods have been exhaustively studied and evaluated. Annex I contains more detailed information on the two systems, i.e., one based on CCITT Recommendation J.17, the other based on 50/15 μ s.

3. Adaptive delta modulation (ADM)

3.1 Background

The adaptive delta modulation (ADM) system is based on delta modulation (the decoder is mathematically defined in § 3.2). The audio is encoded into digital form by a simple delta modulator after being modified by two processes:

- the audio is passed through a variable pre-emphasis network which alters the audio spectrum; and
- the audio is compressed in level, based on its slope. The compression is “infinite” over a 48 dB input signal level; that is, the audio is compressed to the same level for conversion to digital form. Optimum loading of the digital channel is achieved for most audio signals.

The digital bit stream representing the encoded audio signal is transmitted to the receiver, along with two very low data rate bit streams containing control signal information so that the decoder can precisely reverse the processing performed by the encoder. The decoder recovers audio by integrating the audio bit stream; performing a dynamic range expansion based on the compression (slope) control signal; and performs spectral de-emphasis based on information in the emphasis control signal.

The following points may be noted concerning the use of delta modulation:

- (a) the bit rate of delta modulation may be reduced with a minor effect on audio signal quality (S/N degrades by 9 dB for a halving of bit rate);
- (b) reproduced errors are tolerable;

- (c) a digital-to-analogue converter for delta modulation is very simple, requiring no precision components;
- (d) the de-emphasis provides adequate output filtering and sharp cut-off low pass filters are not required.

Items (c) and (d) affect the required decoding circuitry.

3.2 *ADM decoder definition (Fig. 1)*

3.2.1 *Audio decoder*

The audio decoder consists of a leaky integrator fed with pulses derived from the audio data bit-stream so that data 1's and 0's at a rate in the order of 250 kbit/s cause the output to move positively or negatively by equal steps. The size of the pulses is linearly proportional to an applied control signal over a range of about 50 dB, and the leaky integrator has a frequency response defined by:

$$(1 + sT_0)^{-1}$$

where $T_0 = 0.5$ ms.

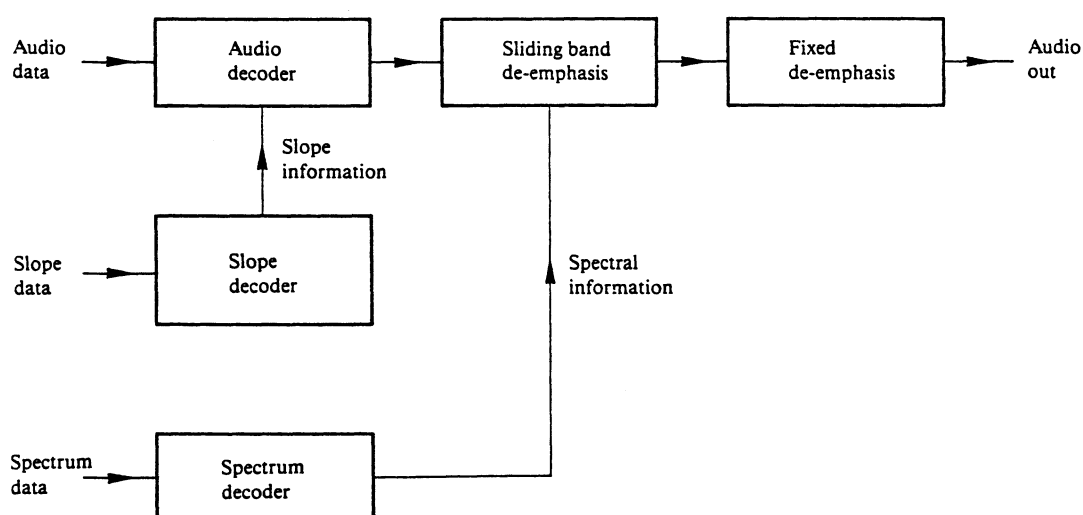


FIGURE 1 – *Adaptive delta modulation decoder*

3.2.2 Sliding band de-emphasis (Fig. 2)

The de-emphasis has the variable frequency response:

$$\left[\frac{10sT_1}{1+sT_1} + \frac{1+sT_2}{1+sT_3} \right]^{-1}$$

where $T_2 = 5 \mu\text{s}$, $T_3 = 50 \mu\text{s}$, and T_1 is variable under the control of the spectrum data.

Note that the first term in the bracket is a high-pass characteristic at frequency f_1 where:

$$f_1 = \frac{1}{2\pi T_1}$$

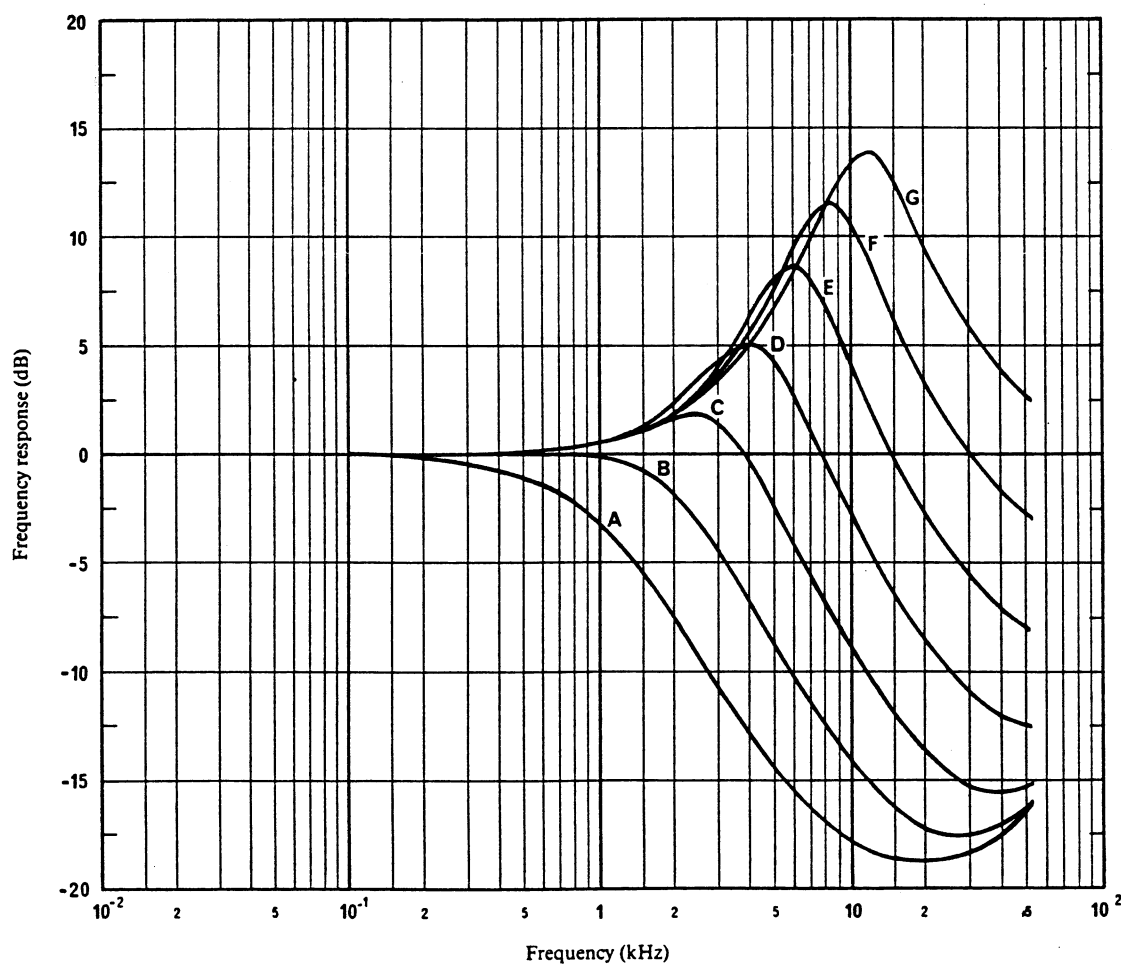


FIGURE 2 – Adaptive delta modulation (ADM) de-emphasis filter

3.2.3 Spectrum decoder

The de-emphasis control information is contained in a data bit stream at about 8 kbit/s, typically but not necessarily an integral sub-multiple of the audio data bit-rate. The mean level V_m of the bit-stream is derived by feeding the data via a three-pole low-pass filter with the characteristic:

$$(1 + sT_4)^{-3}$$

where $T_4 = 2$ ms.

If the data pulses have height V_p , a controlling parameter x is given by:

$$x = \frac{V_m}{V_p}$$

The output of the filter is fed into an exponentiator to produce a control signal to operate on the variable emphasis. The circuit constants are such as to provide the relationship:

$$f_1 = \frac{1}{2\pi T_1} = 4000 (2^{10x}) \quad \text{Hz}$$

With this definition, a change in x of 0.1 moves f_1 by one octave.

3.2.4 Slope decoder

The signal slope information is contained in another data bit stream at about 8 kbit/s. This bit stream is converted to a control signal by means of a low-pass filter and exponentiation exactly as in the spectrum decoder. The height of the pulses integrated in the audio decoder is linearly proportional to this control signal.

As in the spectrum decoder, the mean level V_m of the bit stream is derived using a three-pole low-pass filter with the following characteristic:

$$(1 + sT_4)^{-3}$$

where $T_4 = 2$ ms,

and is then fed into an exponentiator to provide the step or pulse size control signal V_{ss} for use in the audio decoder. This exponentiator has the characteristic:

$$V_{ss} = V_0 (2^{10y})$$

where y is the normalized mean level of the pulses (as x above) and V_0 is a constant scaling factor to suit the audio decoder. With this definition, the pulse height changes 6 dB for each change in y of 0.1.

3.2.5 Fixed de-emphasis

The fixed de-emphasis is a single pole low-pass filter with the characteristic:

$$(1 + sT_5)^{-1}$$

where $T_5 = 25$ μ s.

In consumer decoders no further output filtering is necessary.

REFERENCES

CCIR Documents

[1986-90]: a. 10/52 (Japan)

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ANNEX I

EMPHASIS TECHNIQUES FOR HIGH-QUALITY PCM SOUND SIGNALS

In the technique of digital PCM sound coding, pre- and de-emphasis methods are to be used for several purposes:

- in the case of digital audio signal processing, the emphasis reduces the subjective perceptibility of the quantizing noise, especially for the so-called programme-modulated noise with companded coding;
- in the case of both companded and linearly coded audio signal processing, emphasis may reduce the impairment due to bit errors at low C/N values.

Two different characteristics have been proposed for both companded and linearly coded systems. The first pre-emphasis system is based on CCITT Recommendation J.17 (with an insertion loss of 6.5 dB at 0.8 kHz) and the second on the characteristic of 50/15 μ s time constant. Characteristics of these are depicted in Fig. 1 of Recommendation 651.

Results of investigations into linear coding [CCIR, 1982-86a] and linear and companded coding [CCIR, 1982-86b and c] clearly show the advantages of using pre-emphasis in most cases. It has been observed that both systems have the quality of reducing the quantizing noise, with a slight advantage for 50/15 μ s pre-emphasis in the case of non-companded operation. Concerning the risk of overloading, the documents [CCIR, 1982-86a, b and c] reflect different points of view. No clear difference between these two systems has been noted on impairments due to bit error at low C/N levels.

Results of theoretical assessments [CCIR, 1982-86b] and subjective tests [CCIR, 1982-86d and e] carried out for companded systems, based on the pre-emphasis characteristics shown in Fig. 1 of Recommendation 651, show that a significant reduction of the subjective audibility of the programme modulated noise could be achieved by applying CCITT Recommendation J.17 pre-emphasis. However, results of other subjective tests [CCIR, 1982-86f and g] showed no significant difference. In [CCIR, 1982-86f], insertion loss adjustment in each system was performed in order to avoid overloading.

As far as digital coding of sound programme signals is concerned, CCITT Recommendation J.17 (with an insertion loss of 6.5 dB at 0.8 kHz) is widely used for transmission purposes, as specified by CCITT Recommendation J.41, and the 50/15 μ s pre-emphasis system is widely used for consumer applications.

REFERENCES

CCIR Documents

- [1982-86]: a. 10/21 (EBU); b. 10-11S/201 (EBU); c. 10-11S/139 (Japan); d. 10/269 (Germany (Federal Republic of)); e. 10-11S/206 (France); f. 10-11S/205 (Japan); g. 10-11S/207 (Canada).