

REPORT 954-2

MULTIPLEXING METHODS FOR THE EMISSION OF SEVERAL DIGITAL AUDIO SIGNALS AND ALSO DATA SIGNALS IN BROADCASTING

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

(1982-1986-1990)

1. Introduction

This Report contains the preliminary results of a comparison of the multiplexing methods that can be used for the emission of several channels carrying digital sound and possibly other information, either with or without an associated analogue television picture emission, for new broadcasting applications. In this context this Report deals primarily with the use of the satellite-broadcasting channel, although some of the information is also valid for terrestrial broadcasting channels.

The study of multiplexing methods for several audio signals and data must take due account of the two main methods of multiplexing the complete digital signal with the video, which are:

- “interrupted”, i.e. digits in the line-blanking interval;
- “uninterrupted”, i.e. digits on a sub-carrier.

In the latter case, for sound only in the absence of a video signal, the digits can be on a carrier, but this is only a special case of the sub-carrier. In the case of “interrupted”, the case of digits in the field-blanking interval is not considered in this Report.

Regarding the multiplexing of the various digital sound and data signals, two basic techniques are envisaged here: “continuous” multiplexing and “packet” multiplexing; the advantages and disadvantages inherent in the principle of each system have been investigated for both cases mentioned above, i.e. the interrupted and uninterrupted multiplex. Further studies will be needed to optimise these types of systems.

2. Services to be offered

Sound services envisaged in broadcasting are:

- high quality (stereophonic or monophonic) fully encoded digital sound with an audio bandwidth of 15 kHz and a dynamic range of up to 98 dB [CCIR, 1982-86a, b] for sound broadcasting programmes only;
- high quality (stereophonic or monophonic) sound associated with video (audio bandwidth of 15 kHz);
- high quality (monophonic, stereophonic or even quadraphonic) sound for additional radio sound programmes (audio bandwidth of 15 kHz);
- monophonic high quality or medium quality sound for various purposes (e.g. for multilingual commentaries in association with the international sound, etc.);
- commentary quality signals;
- telephone quality signals.

Additional data services envisaged could include:

- data information (e.g. service information, coded text, sub-titling, computer software and programme labelling);
- special information for pay TV service;
- paging.

This list is not exhaustive: in the future, other applications may be possible in accordance with the evolution of needs and technology. The requirements may vary from country to country and from time to time.

For this reason, some flexibility in the use of the digital bit stream is desirable. At the same time, it is necessary that the multiplexing techniques used should be standard and as simple as possible, in order to minimize receiver complexity and reduce costs while facilitating the reception of the various services even if they are not presently identified.

In the case of satellite broadcasting with a video signal, the sound requirements are for a capacity giving the equivalent of two to eight high quality monophonic sound channels. In the absence of a video signal, this requirement can rise to the equivalent of thirty to forty monophonic sound channels.

3. Multiplexing of the digital signal in the television channel

Two principal methods for the multiplexing of the digital signal in the television channel have been identified. They are:

- “interrupted multiplexing” corresponding, for example, to the inclusion of digital pulses in the line synchronisation, either in baseband (system B, see § 4.3.1 of Report 632), or at radio frequency (system C, see § 4.3.3 of Report 632); this method corresponds to the principle of time-division multiplexing, which is used in a generalised manner if the image signal is based on the coding of time-compressed components (MAC system);
- “uninterrupted multiplexing” corresponding, for example, to digital pulses on a sub-carrier (system A, see § 4.2.2 of Report 632); this method corresponds to the principle of frequency-division multiplexing, which is used in a generalised manner if the image is based on the coding of a composite signal with a colour sub-carrier.

The case of sound with no video is considered to be a special case of the uninterrupted multiplex. Before considering the multiplex of sound with data, it is necessary to consider the implications of the bearer channel on services being carried.

Services fall into four categories:

- (a) those in which the data is generated at a regular rate and which must be recovered at a regular rate and for which propagation time is critical (e.g. digital sound);
- (b) those in which the data is generated at an irregular rate, but for which propagation time is critical (e.g. coded sub-titles for an accompanying television programme);
- (c) those in which data is generated at an irregular rate and for which the propagation time is, within limits, unimportant (e.g. service messages);
- (d) those in which the data is broadcast cyclically at a rate such as to fully occupy the spare capacity of the system (e.g. certain forms of coded text and computer software broadcasting).

It is clear from this that the most important service, sound, is also the most critical as far as the bearer channel is concerned, and will therefore be considered here in more detail.

Sound samples can be considered as being generated regularly (e.g. at 32 kHz rate). At the receiver, it is necessary to recover the regularity of the audio samples in order to avoid distortion or “wowing” of the sound. This operation requires the recovery of the audio clock, which can be achieved by the following two methods:

- when the characteristic frequencies of the bearer channel are asynchronous and completely unrelated to the sampling frequency of the audio, it is necessary to use some form of elastic “first-in, first-out” store and clock rate with sufficient precision, with or without feedback, such that the rate of reading out from the store is made equal to the average rate of filling the store;
- when the characteristic frequencies of the bearer channel are synchronous with or related by a rational fraction to the sampling frequency of the audio, then the bearer channel itself may be used to convey the audio clock frequency. In the case where a relationship between the sampling frequency and the bit rate of the bearer channel is in the form of p/q , the clock recovery can be obtained by a phase locked loop. As an example, for interrupted digital signals, the 32 kHz can be recovered if related to line frequency by the ratio of 256:125. If the relationship is in the form of $1/n$, a simple division is sufficient and reduces the receiver costs. As an example, for uninterrupted digital signals, the sound clock can readily be recovered if the overall bit rate is an integer multiple of 32 kHz (e.g. 2048 or 1792 kbit/s).

The first method places much of the complexity in the domestic receiver design while giving a relatively simple transmitter design. The process of such asynchronous clock recovery can also lead to timing jitter on recovered clocks, particularly in a domestic design where cost must be minimised.

The second method requires that all the sampling frequencies of the audio channels are synchronised to the bearer channel, leading to complexity at the transmitter in requiring sampling rate synchronisers, but results in the simplest and most stable system design for domestic receivers.

The inherently intermittent nature of a TDM channel and the basic principles of the packet multiplexing system prevent the direct signalling of precise time or phase relationships such as those needed for some types of control signals or multi-channel sound. In such or similar cases, coding efficiency may be preserved by taking advantage of the television synchronizing signals to provide a reference timing grid, and using the relationship between the sound coding blocks and the timing reference. Such a method is described in Part 3, Chapter 3, of the CCIR Special Publication "Specification of Transmission Systems for the Broadcasting-Satellite Service".

The timing problems of data services (b), (c) and (d) (as categorized above) are rather less critical than for the sound services and therefore do not present any special problems.

It is clear from this that synchronous operation of the sound channels with the bearer channel is an advantage irrespective of the method by which the sound and data channels are themselves multiplexed to form the digital signal.

4. Multiplexing of the sound and data channels

The single digital channel considered in § 3 above for multiplexing with the video is itself produced by time division multiplex of the various sound and data signals. Two basic multiplexing concepts are possible, which will be described as "continuous" and "packet".

4.1 Continuous multiplexing

In conventional continuous multiplexing, a given fixed number of digits (called a frame) consists of bits, each of which is assigned a specific purpose according to its position in the frame. Thus, particular bits are dedicated to convey the information relating to one input signal. A predetermined pattern of bits (a frame alignment word) enables the receiver to identify and extract the particular bits which convey each signal in the multiplex.

The simplest form of such a multiplex has input signals whose bit rate is a precise sub-multiple of the final serial bit rate. This is known as synchronous multiplexing. When this condition does not apply, a process of asynchronous multiplexing is possible by arranging certain bits within the frame to carry either real or dummy information according to a control signal. This process is known as justification. An alternative method can be used for sound signals in which the sampling rate is adjusted by means of a synchroniser so that synchronous multiplexing can be used. Proposals have been made to use continuous multiplexing in the asynchronous mode, by use of justification, to insert audio signals sampled at 32 kHz in the line-blanking interval, but other methods are also possible. It is clear from § 3 that the synchronous form of multiplexing has many advantages.

Continuous multiplex systems require only a small proportion of the total bit rate for "overheads" such as framing. An example of the efficiency which can be obtained in a regular multiplexing system is given by NICAM 3 [Caine *et al.*, 1980] which only uses 7 kbit/s out of the total 2048 kbit/s for framing. The typical reframing time after loss of synchronization for this system is 2 ms.

In normal service, sound channel re-assignment will be relatively infrequent because of the continuous nature of sound. A continuous multiplexing system is therefore particularly suited to this application with a relatively low overhead required for secure definition of the sound channel structure. A small capacity can be used for signalling to indicate change of use of channel and thus provide flexibility. The capacity of any channel not used for sound can be assigned to data. In the case of data, the detailed structure can be carried in the data channel in a way similar to that used in packet multiplexing with a constant length of packets (see § 4.2). This method does not therefore involve any increase in the multiplex overheads to indicate the presence of data. The aim of such a structure is to exploit to the utmost the channel capacity for the useful information with a reasonable degree of flexibility, as well as simplicity and stability in the receiver design.