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)

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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.

An advanced form of continuous multiplexing studied in Europe involves the transmission, in the digital frame, of coded information known as a "structure map"; this indicates the particular configuration of the multiplex. The addresses and service information are carried in this structure map and they serve to control the demultiplexer which selects the required bits and feeds them to the appropriate decoders. This method enables the provision of the desired flexibility in a continuous multiplex, the configuration of which may thus be revised frequently. A typical useful channel efficiency of 99% is possible. Other forms of continuous multiplexing are described in [CCIR, 1982-86a, c].

The digital subcarrier NTSC system (see Report 1073 (MOD F) has a continuous multiplexing structure for sound and data channels, with control codes in a digital frame indicating the structure of the frame. The system also has a packet multiplexing structure within the data channel area [CCIR, 1986-90a].

4.2 *Packet multiplexing*

In this case, the final bit stream is composed of successive blocks, called packets, with two parts: *heading* and *data*. Each packet conveys data from only one input signal and the selection in the receiver is realised by detection in the heading of the address of the desired service: this method does not impose a predetermined content on the final bit stream [CCIR, 1978-82a].

In the form of the system designed for broadcasting applications, the length of each packet is constant. The packets are transmitted "synchronously" which means that there is continuity of phase of the binary sequence and of the modulated carrier, between packets or groups of packets. The packets are also transmitted with regular periodicity. The checking process then becomes a synchronisation process similar to that used in a continuous multiplex structure. This synchronisation can be done in a conventional manner by the use of a loop locked to a synchronisation flag in the heading of each packet. This restriction of fixed overall packet length does not imply a fixed length of the section containing useful data.

Within the constraints of periodic packet transmission, the rate at which packets are transmitted for a particular service is related directly to the bit rate of the input signal of the associated service. In packet multiplexing, it is possible to operate synchronously or asynchronously. In the latter case, the bit rates of the various input signals do not need to be related directly to the final serial bit rate. Thus, no special provisions are necessary for accommodating asynchronous signals. The content of the multiplex does not need to be pre-determined or fixed and can be changed at any time to be adapted to the needs. In the case of asynchronous operation for sound services (e.g. for insertion of sound signals in the line-blanking interval), packet multiplexing inherently permits the asynchronous insertion into the final bit stream but the recovery of the sampling frequency needs resynchronisation processes as described in § 3.

A packet is formed of two parts:

- a section containing useful data,
- a heading, specific to the transmitter, which serves to synchronise the receiver and to identify the source of the data inserted in the packet and for the transmission of other information. The heading contains a synchronisation flag followed by a prefix.

The synchronisation flag enables the receiver to extract the bytes comprising the packet. It has a role equivalent to that of a locking word in the case of a continuous multiplex. In a periodic transmission, it can also be used for the synchronisation of the loop which serves for the checking process. We should note also the possibility of using this flag to remove the ambiguity due to certain demodulation processes; such an arrangement is advantageous in that it avoids the coding of transitions and, hence, a certain propagation of transmission errors.

The role of the prefix is to characterise and identify the semantic content of the packet and, more especially, the source of the data inserted in the useful data section. Several prefix configurations can be envisaged. In particular, it may be possible to use the prefix configurations considered for data broadcasting [CCIR, 1978-82b].

The increased flexibility to the data of the packet multiplex system is obtained at the price of increased channel overhead for the packet headers which includes, in normal operation, the insertion of programme identifiers. A typical useful channel efficiency of 97% is possible.

4.3 *Sensitivity of continuous and packet multiplex systems to errors*

4.3.1 *Continuous multiplexing*

The continuous multiplexing process is susceptible to two sorts of impairments:

- bit errors,
- frame synchronisation loss.

Errors introduced into multiplex signals can cause loss for a certain number of consecutive frames of synchronism at the demultiplexer, when the framing pattern is not recognised. Careful attention to the design of framing circuits can minimise the risk of losing synchronism; for example, the decoding equipment for the NICAM 3 system mentioned earlier remains synchronised until the bit error ratio approaches 1 in 10. A very rugged synchronisation of framing can be achieved by using special type synchronisation codes (i.e. the class of Barker codes). With this implementation a very rapid, as well as a very stable synchronisation locking at BER of 10^{-1} is feasible [CCIR, 1982-86a].

4.3.2 *Packet multiplexing*

The packet broadcasting process is susceptible to two sorts of impairments:

- bit errors,
- packet loss.

In effect, for each source there is a digital channel identification carried in the packet prefix. In the demultiplexer, the packets are selected by analysis of this identification. Errors in this information, despite its protection, are likely to result in poor recognition of the address of the transmission source and hence, in the loss of the packet, this being apparent at service level as the loss of a certain number of successive bytes (which depends on the format of the data section of the packet(s) lost).

This phenomenon must result in an interruption in the sequence of the service frame and in a loss of synchronisation in the service terminal. Nonetheless, since the length of the *service frame* is a sub-multiple of a *fixed packet* length, there will be a loss of information without a loss of synchronisation.

In this case the packets associated with a given sound source will always have a fixed and pre-determined length which is known to the demultiplexer. It is for this reason that it is preferable to use a prefix configuration without a format byte in these applications.

Several tests of broadcasting digital sound signals with packet multiplexing and with modulation appropriate for satellite broadcasting (with television pictures) have been made in the laboratory as well as with the OTS satellite [EBU, 1981]. Some results are also given in Annex I to Report 632. It has been shown that the packet losses appreciably degrade the sound quality only at levels of the carrier-to-noise ratio that are below the FM threshold, when the television picture is already severely impaired.

4.4 *Experimental results*

Demonstrations of the continuous multiplex with structure map and of the packet multiplex have been carried out by the EBU in association with modulation systems A and C envisaged earlier for satellite broadcasting. More recently, tests have also been made on the packet multiplex in association with the D2 modulation system (see Report 632). The experimental multiplexing systems were designed to provide the greatest possible capacity, taking account of the bit rate imposed for the modulation system. The tests were concerned in particular with the various methods for ensuring protection against errors and with the possibilities for reconfiguring the multiplex with a change in the number and nature of the sound channels and the insertion of data services. The conclusions are summarized in § 4.4.1 and 4.4.2.

It is noted that:

- system capacity is expressed as an equivalent number of high-quality sound channels using near-instantaneous companding (see Report 953) and with a simple error-protection system;
- the available bit rate was 2048 kbit/s with type A modulation (sub-carrier) and about 3 Mbit/s (mean value) with type C modulation (RF time multiplex with an instantaneous bit rate of 20.25 Mbit/s) and about 1.5 Mbit/s (mean value) with type D2 modulation (baseband time multiplex with duobinary coding and instantaneous bit rate of 10.125 Mbit/s).

A continuous multiplex with rigid structure has been employed in a B type modulation system (described in Report 1073, Table III). With this multiplex technique the structure is essentially predefined and fixed by hardware implementation. The capacity is determined by the modulation and coding technique only, as no additional overhead is required to define the multiplex structure. The characteristics of this system are summarized in § 4.4.3.

For type B modulation (baseband time multiplex with four-state coding and an instantaneous bit rate of approximately 14.25 Mbit/s) the available bit rate is 1.57 Mbit/s.

4.4.1 *Continuous multiplex with structure map*

4.4.1.1 *Capacity*

In a type C system, with video, the structure map system offers capacity equivalent to 8 high-quality compressed sound channels.

Any combination of sound channels with linear coding or with companding or other types of sound channel are possible, provided that the total capacity is not exceeded. Any remaining capacity may be used for data.

In a type A system, the multiplexing permits the broadcast transmission of 6 companded audio channels with reduced error-protection (one parity bit for every two samples).

4.4.1.2 *Flexibility of the multiplex*

It is easy to accommodate the sampling frequencies and coding methods recommended for the sound. Other sampling frequencies and other coding methods could also be used. Any form of error protection may be adopted.

The data channels can have a capacity increasing in steps of 100 bit/s and ranging from the lowest values up to the entire available bit rate.

Modifications to the multiplex structure can be made rapidly and with security, so that at any moment the broadcast channel can be used in the optimum fashion for the combined transmission of sound and data. Changes in structure are synchronous and cause neither a variable delay nor an interruption in any channel.

In the case of a system C not carrying a video signal, it is possible to increase the capacity available for the sound and data to about 20 Mbit/s.

4.4.1.3 *Quality and continuity of the audio channels*

The multiplexing technique preserves all the synchronisation and timing information. Phase coherence is therefore assured between all present and future channels. Different error-protection strategies may be used for each of the audio or data channels.

An error in the decoding of the map, which may cause failure of all the channels, cannot occur except well below the failure point of the audio channels.

4.4.1.4 *Simplicity of the demultiplexer*

Construction of the demultiplexer may take the form of a circuit providing all the protection and demultiplexing functions, or an autonomous integrated demultiplexer/decoder for each channel. The logic needed will be the same for all types of channels, including those carrying the structure map.

4.4.1.5 *System ruggedness*

It has been demonstrated that the system synchronisation operates satisfactorily at bit-error ratios slightly lower than 10^{-1}.

4.4.2 *Packet multiplexing*

4.4.2.1 *Capacity*

For system A, the packet multiplex offers a capacity equivalent to 5 companded high-quality monophonic audio channels.

In the case of system C, the specifications of the packet multiplexing system given in Report 1073 offer capacity equivalent to 8 companded high-quality monophonic audio channels. For the D2 system specified in the same Report the offered capacity is equivalent to 4 companded high-quality monophonic audio channels.

Any combination of sound channels with linear coding or with companding or other types of sound channel are possible, provided that the total capacity is not exceeded. Any remaining capacity may be used for data.

4.4.2.2 *Flexibility of the multiplex*

This multiplex system meets the requirements set out in Report 953 on sound coding. As regards the data, it permits the insertion of synchronous or asynchronous services, without any limit on the bit rate (in particular, for low values). Changes in configuration can be assured rapidly and with security. In the case of system C, it is possible, in the absence of a video signal, to increase the capacity available for the sound and data to about 19.5 Mbit/s.

In principle, any form whatsoever of error protection can be used. Two levels of protection are provided in the system specified in Report 1073.

4.4.2.3 *Sound quality and continuity*

Even at high bit error ratios it has been shown that the quality and continuity of the sound are preserved and that modifications in the form of the sound signals to meet operational requirements are effected without impairment to the audio quality. Timing coherence between audio channels is assured.

4.4.2.4 *Simplicity of the demultiplexer*

The selection of a given channel is independent of its content (sound or data) and therefore it can be effected in the same manner for decoders for all types of service.

This feature may be used to implement a simple demultiplexer for existing services and for those yet to be defined.

4.4.2.5 *System ruggedness*

The packet multiplexing system with the specification defined in Report 1073 has been tested. Its capability to broadcast different sound and data channels has been verified for different bit error ratios. Table I gives information concerning the efficiency of the protection against errors on the header field.

TABLE I

Measured bit error ratio (during 30 s)	Packet loss rate	
	Measured	Calculated from the measured bit error ratio
6.6×10^{-5}	0	0
3.2×10^{-4}	0	10^{-10}
1.2×10^{-3}	0	1.6×10^{-8}
3.6×10^{-3}	0	1.4×10^{-6}
9.2×10^{-3}	7.4×10^{-5}	5.5×10^{-5}
2×10^{-2}	1.2×10^{-3}	1.05×10^{-3}
3.8×10^{-2}	10^{-2}	1.03×10^{-2}
6×10^{-2}	5.3×10^{-2}	4.6×10^{-2}
8.8×10^{-2}	1.7×10^{-1}	1.38×10^{-1}

4.4.3 *Continuous multiplex with rigid structure*

4.4.3.1 *Capacity*

In a type B system the rigid structure offers 6 independent audio channels (using ADM coding (see Report 953)). Additionally a data channel of 62 kbit/s is available.

4.4.3.2 *Flexibility of the multiplex*

Any audio channel may be reconfigured into a data channel with a data rate of 204 kbit/s.

4.4.3.3 *Sound quality and continuity*

Timing coherence between audio channels is assured. It has been demonstrated that audio quality is impaired to a quality rating of 4.4 at a BER of 10^{-4}, and impaired to a quality grade of 3.5 at a BER of 10^{-3}.

4.4.3.4 *Simplicity of the demultiplexer*

The rigid structure allows for the simplest possible demultiplexer.

4.4.3.5 *System ruggedness*

It has been demonstrated that system synchronisation occurs satisfactorily at bit error ratios worse than 1×10^{-1}. Both two-state and four-state coding is provided for flexibility in terms of capacity versus ruggedness. (The capacity drops to one-half of that shown in § 4.4.3.1 when two-level coding is being used.) In both cases system synchronization is two-state encoded.

5. Conclusions

This Report has outlined two basic methods of multiplexing digital sound and other signals together for broadcasting.

Summarising, continuous multiplexing offers, in the context of rigid structure, a system which is most efficient for sound transmission and results in a simple receiver. In the context of flexible structure, continuous multiplexing offers a system with adequate flexibility at the price of a greater receiver complexity and some increase of the bit rate overhead. The packet multiplexing system inherently offers a great flexibility with a receiver of medium complexity at the price of bit rate overhead. Both systems require only simple synchronisation circuits at the receiver if the sampling frequencies of the signals to be combined are synchronous or can be synchronised. Each system is in principle vulnerable to the effects of error, but frame loss and/or packet loss are not likely to occur in normal operation.

For the broadcast transmission of a group of digital audio and data signals accompanying the television image, the choice of multiplexing system depends principally on the nature and diversity of services, even if these have not yet been identified. Factors influencing the choice are efficiency, flexibility and receiver complexity. The two multiplexing systems that have been examined in Europe each provides a satisfactory solution to the overall requirements that have been expressed. In view of all the relevant factors, the packet multiplexing has been fully specified for satellite broadcasting at 12 GHz with 625-line television standards. The detailed specifications of the chosen packet multiplex (in association with type C and D2 modulation) are given in Report 1073.

The continuous multiplex with rigid structure has been fully specified for satellite broadcasting at 12 GHz, and is identical for either a 525-line or 625-line system. The detailed specifications for this multiplex (in association with type B modulation) are given in Report 1073, Table III.

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