DIDON, the Experimental Data Packet Broadcasting System of the French Administration

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The development of the experimental data packet broadcasting system DIDON was undertaken by the French Administration in 1975 to offer new services on television networks.

The fundamental choices governing the design of the system had to take into account the existing equipment and the diversity of the services forecast. A data packet multiplexing organization meets all these requirements.

Three kinds of techniques have been implemented : transmission techniques, reception techniques and quality assessment techniques.

The use of these techniques adds a new dimension to the field of data communications.

I INTRODUCTION

Recently emerging teletext and still image broadcasting services have underscored the possibility of using a television channel to transmit services in addition to conventional television service. These services may be classified into two types, depending on whether they are television-program dependent or whether they are useful even without a television program. Two services of the first type have been investigated in France : the first is a program delivery service designed for the telecommand of video recorders for program recording in user's absence; the second is a scrambled television service. With regard to the second type, the French Administration has concentrated on services leading to the display of a non - analogue picture, that is, constructed by the receiving equipment from the codes received.

These services are :

1) - a teletext service leading to the display on a T.V. set screen of a picture composed of alphanumeric or simple graphic symbols,

2) - an audiography service where the T.V. set displays a slow-moving picture accompanied by an audio commentary, and

3) - a broadcast facsimile service using paper to provide a very high resolution picture.

The common feature of all these services is that they are based on broadcasting digital data intended for simultaneous reception by a large number of users. To solve the problem of multiplexing them all into one television channel, in 1975 the French Administration undertook the study and the development of a data broadcasting system which would take advantage of existing equipment used for television signal distribution. This system is the subject of this paper. The fundamental choices taken at the design level are set forth, and the techniques utilized for the implementation of an experimental system are described.

II FUNDAMENTAL CHOICES

The fundamental choices governing the design of the data broadcasting system are based on two considerations :

- the set up of the system take advantage of existing networks,

- the system is to be used by different types of services.

The conclusions to be drawn from these considerations are presented at the end of section II.

2.1. Requirement of compatibility with existing standards and equipment

The use of existing equipment now used for the distribution and reception of television signals allows rapid system growth. In addition, it avoids heavy investment in a new broadcasting network and limits expenditures for installing specific transmission and reception devices. Obviously, the use of an already existing network necessitates compliance with the standards on which this network has been built.

Therefore, the resources available to the data broadcasting system must be investigated from two points of view : time and frequency spectrum.

2.1.1. Resources available in the time scale

The examination of a television signal shows that certain time intervals are not used by either the picture signal or the synchronization signals. These intervals consist of a number of line time slots in the field blanking interval (fig 1 and 2) (although the use of some of these slots for the transmission of measurement or identification signals has increased in the last few years).

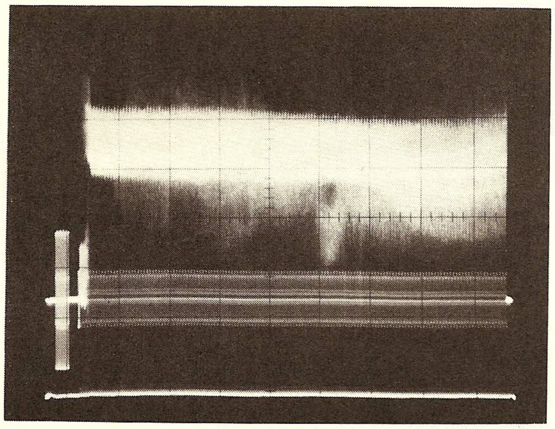


FIG.1 T.V. signal (one Field)

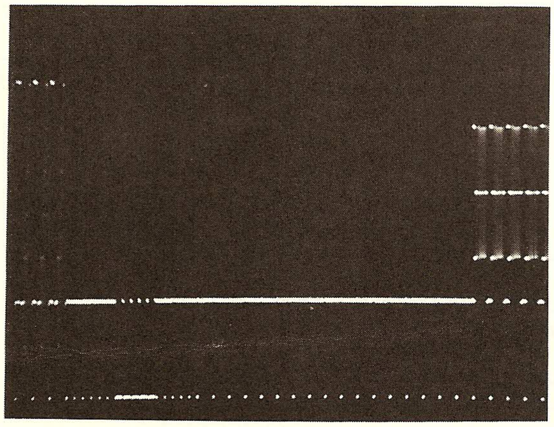


FIG.2 Field blanking interval

Our purpose is to standardize the use of these line time slots and to extend this technique to all television signal lines free while television programs are not being broadcast. A statistical study indicates that television networks are currently utilized only about 50 % of the time for television program broadcasting. Evaluating the marginal working cost of these networks indicates the advantage of more complete use. Therefore, the available resource is discontinuous (composed of free line time slots) and time-dependent (according to whether or not a picture is being broadcast).Thus an asynchronous system was designed.

2.1.2. Resources available in the frequency spectrum

Whereas the active time of the video signal line is about the same in all the commonly used television standards, this is not the case for the spectral bandwidth allocated to the video signal (table 1). Thus, it is essential that the system be adaptable to any value of this parameter. Indeed, the data signal spectrum must be in close relation with the value of the video bandwidth. This means that the bit duration may not be the same for all television standards; the amount of information inserted in a quantum of transmission capacity (a television signal line) may not be constant from one standard to another. Therefore, the system must allow an arbitrary slicing of data flows.

2.2. Use of the system by different services

In the introduction a number of services already under investigation were enumerated. However, it is certainly not an exhaustive list. To preserve the possibility of offering other services, the system design must take into account the following two points :

2.2.1. Differences in the nature of the services

Because of their different aims, the services require the transmission of data of different types. To avoid any limitation, the system must be transparent.

2.2.2. Simultaneity of the services

The substantial resources can be used efficiently only if the system allows several services to be present simultaneously : therefore, the system must provide multiple access.

2.3. The fundamental choices for the design of the system

In conclusion, the system must :
- be asynchronous
- allow arbitrary slicing of the data flows
- be transparent
- provide multiple access.

Data packet multiplexing organization satisfies all these requirements. To understand this concept better, data packet broadcasting can be compared with the well known technique of packet-switching.

In packet switching networks, each packet contains the information needed to route it towards the appropriate receiving equipment. Similarly, each broadcast packet contains the information that makes it recognizable by the receiving equipment it was aimed at. This comparison may be summarized as follows :

point-to-point communication ⟷ switching

broadcasting ⟷ selection

Therefore, the basic sequence of operation is : the assembly of the packet from the data, the labeling of the packet according to data origin, the broadcasting of the packet, and the selection of the packet by the correct receiving equipment.

III DESCRIPTION OF THE TECHNIQUES IMPLEMENTED

3.1. Transmission techniques

The transmission techniques are those used at a given point of the television broadcasting network to insert the numerical data into the video signal. This insertion can be done at the national, regional or local level.

The basic technique is TDMA at two levels :

- multiplexing of packets generated by different sources,

- multiplexing of these packets with the video signal, using the free line time slots.

3.1.1. Resource management

We have seen in § 2.1.1. that the available resource is composed of free line time slots. A few lines per frame are free when a picture is being broadcast; otherwise almost all of them are available. The problem of resource management actually amounts to the problem of video-data multiplex organization. The time-dependent resource is to be shared between users, whose number also varies with time, and whose needs, measured in terms of the quantity of information to be transmitted, may differ. A part of the transmission capacity must be allocated to any active data source : for that purpose, so-called digital channels are defined, one for each active data source. The management system must meet the basic criterion of allowing maximal resource use while efficiently controlling its distribution. This control must match the available resource as closely as possible at a given time with the requirements of the various data sources expressed in terms of throughput. The following method has been adopted : each source outputs data which are accepted by the management system at a certain rate. This rate characterizes the digital channel employed and is defined by parameters specified in the next paragraph. Data blocks are formed, each consisting of data supplied by a particular source. These data blocks are provided with a header containing, among other items, the identifier of the digital channel assigned to the source supplying the block data. The block-plus-header unit constitutes a packet (fig 3 and 4). These packets are described more completely in § 3.2.

The active data sources are scanned and the indication "packet ready" is memorized. The multiplexing device is programmed to select a number of lines in each field usable for data transmission. When such a line appears, the multiplexing device selects one of the data sources in accordance with rules of priority derived from the parameters characterizing the digitals channels. The corresponding packet is then transmitted, i.e. inserted in the free line time slot.

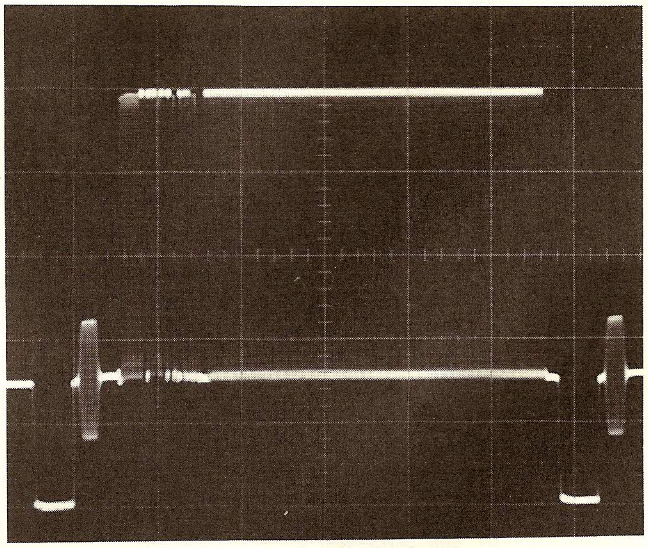


FIG. 3 A packet

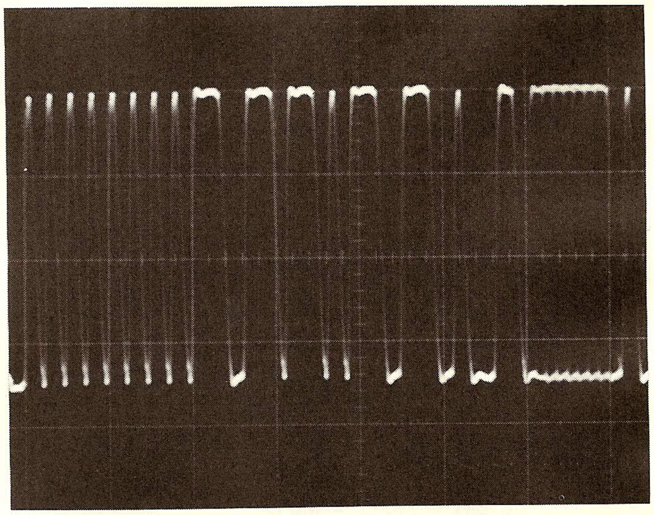


FIG. 4 A packet - header

3.1.2. System access

A given transmission capacity is allocated to each source authorized to use the system. Such a source respects, in exchanges with the system, a certain number of constraints. The definition of these constraints can be done through the characterization of digital channel. A digital channel is characterized by some parameters. For the definition of these parameters, we need to consider the digital channel as a whole, emission end and reception end; it gives us the opportunity for a transition towards the examination of the reception techniques.

a) Characteristics of a digital channel

A digital channel is the link established by the packet broadcasting network between a data source and a terminal using the data supplied by this source. It is delimited at both ends by a digital interface :

- between the source and the network on the emission side;

- between the network and the terminal on the receiving side.

The following paragraph describes these digital interfaces. The total transmission resource has a discontinuous character. The transmission resource provided by a digital channel presents a fortiori this same character due to the use of time division multiplexing. The use of buffers for emission and reception is therefore pratically indispensable.

The parameter characterizing a digital channel is the throughput of the channel. The discontinuous character of the transmission implies that the period of time for which the throughput is computed must be defined. The following two parameters can be used (fig 5) :

- the size of the emission buffer (EB) or reception buffer (RB);

- the maximum time allowed for filling the emission buffer (tf) or the maximum time allowed for emptying the reception buffer (te).

If the throughput of a digital channel is known, two types of constraints can be handled :

- emission constraints linked to resource sharing;

- reception constraints linked to terminal operation.

The loss of data due to overloading of the receiving terminals must be avoided. This typical teleprocessing problem becomes particularly acute here due to the one-way character of the transmission medium. An adequate condition for smooth operation is :

$$EB = RB \text{ and } tf = te$$

The maximum throughput D max can therefore be defined as follows :

$$D\ max = EB/tf = RB/te$$

Non-compliance with the maximum times tf and te has different implications according to whether tf or te is involved :

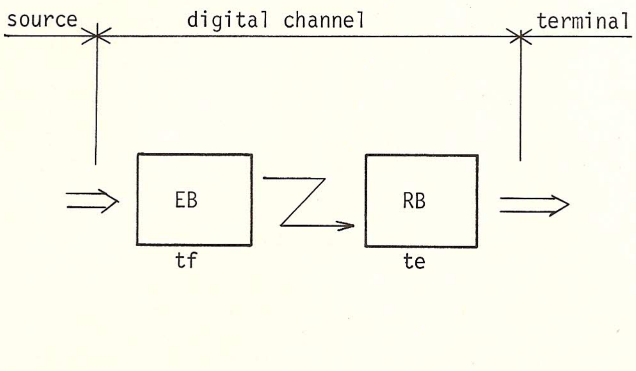


FIG. 5 A digital channel

- If the emission buffer filling time is greater than the maximum allowed (tf), the data source loses part of the transmission capacity which might have been allocated to it. The maximum throughput of the channel will therefore not be obtained.

- If the receiving buffer emptying mean time is greater than the maximum allowed (te), the terminal loses part of the data transmitted by the source. The parameters defining the maximum throughput should be taken into account in designing terminals using the data transmitted on a digital channel characterized by these parameters.

The throughput of a channel may, however, depend on the total system load. To complete the characterization of a digital channel, it is therefore necessary to define the minimum throughput which reflects the limit fixed to this possible reduction of throughput.

A data channel has been defined in a general way. But some services are more demanding than others as far as data transmission rate is concerned : for example, fixed and regular data rate. The adoption of special priority rules for resource sharing allows meeting these particular needs.

b) Digital interfaces

Resource sharing by different users involves the establishment of digital channels, each assigned to a particular user, namely, a source-terminal family. These digital channels are characterized by their throughput but must be general from the standpoint of their ability to carry any type of digital information : they should be transparent to users.

Generally speaking, communication between a transmitter and a receiver operating at different speeds is possible in all cases only if the receiver can control the transmitter via their interface. Since the digital channels have to be general purpose, this problem occurs at both ends of a channel.

Receiving end : The receiving terminal equipment is the same regardless of the terminal connected to it. It is therefore the terminal which must take the initiative for data output in order to accept data supplied by the terminal receiving equipment. This transfer must be subject to speed control, with slower equipment slowing down the faster equipment.

Transmitting end : The above arguments apply to exchanges between a source and a transmitting terminal. The transmitting terminal indicates the network's capacity to accept the data supplied by the source : it must therefore control the source. In addition, if data loss is to be avoided, it must also indicate the target terminal's capacity to accept the data transmitted : the problem is therefore the transfer of control between the terminal and the terminal receiving equipment. The specific characteristics of broadcasting (one-way transmission, large number of terminals) prevent the application of the only strict solution to this problem, namely the transfer through the network of the digital interface from transmission to reception. The problem can only be solved by informing the source of the terminal operating speed. This constraint must be part of resource management.

A simplified version of a low-level point-to-point interface being standardized by ISO can serve as the digital interface since it meets the above requirements.

It is a parallel interface in which the exchange of an octet between a transmitter and a receiver is controlled by two regulating wires. The digital transmitting and receiving interfaces are similar, facilitating the design of source-terminal families by allowing local connections. The receiver mentioned above is either the terminal transmitting equipment or the terminal. The transmitter is either the source or the terminal receiving equipment. However, the transmitting and receiving interfaces differ in one respect : in addition to data exchange, it is desirable that the interface should permit, at the data source and at the terminal, the initializing of the terminal equipment of the corresponding transmitting device. This function is performed by an initializing demand wire operating in the data direction for the transmitting interface and in the opposite direction for the receiving interface.

This digital interface is designed for local data exchange, and does not solve the problem of system access by a remote data source. For that purpose, an existing data transmission link must be used. If this link allows speed matching, it can supply an ordinary data channel with data. If the link has a constant throughput , the definition of the corresponding data channel must take into account the requirements imposed by the fixed throughput. In any case, the digital interface may be physically nonexistent, although each system access must obey its functional characteristics.

3.2. Reception techniques

Because a digital channel can only be described as a whole, reception problems have been discussed in the preceding two paragraphs. The digital aspectsof the reception have been described. Therefore the analogue to digital conversion must now be considered. The video-data multiplex may be carried on each support provided for transmission, broadcasting or recording of a television signal. The reception consists of the extraction from the possibly impaired video-data multiplex of the data corresponding to a data channel. This operation may be described by a three-level structure, based on the use of the packet header (fig 4).

3.2.1. The bit level :

This level consists in resynchronizing the receiver bit frequency oscillator with the data signal. A clock run-in sequence composed of alternating logical "1" and "0" and located in the packet header is used for this purpose.

3.2.2. The octet level :

This level allows, after bit synchronization, the reassembly of received information into octets for further processing. The third octet of the packet header is used for this purpose and sought out by bit shifting until a reference configuration of 8 bits is matched. Careful choice of this framing code allows synchronization to take place even if a transmission error occurs, and consequently, the loss of the corresponding data block is avoided.

3.2.3. The packet level :

Following octet synchronization, the packet level carries out the processing of the information contained in the header section of each packet to select the data transmitted by one of the sources on a given data channel.

Three octets are used for data channel identification. The first two are able to generate 2^{16} = 65536 simultaneous data channels, the third protects the first+second permitting the recognition of an idenfier affected by one bit error, thus avoiding packet loss.

Once an identifier has been checked and a packet accepted, the next octet after the identification section verifies that no data packet has been lost between this one and the preceding one. The continuity octet has a value which increases by 1 for each packet transmitted on the data channel.

Since the size of the data blocks is variable, the last octet of the packet header, the format octet, is used to specify size. These last two octets can be organized to provide a single error correcting code for the continuity and format bits.

IV QUALITY OF A DATA CHANNEL DETERMINATION TECHNIQUES

The video-data multiplex signal can be impaired, at emission, during broadcasting, or at reception. A number of techniques are now in use to evaluate television service quality.However, the difference in the nature of a picture signal and a data signal requires special techniques to evaluate the digital transmission quality. Indeed, field trials have shown that imperfect correlation exists between picture quality and performance of an associated digital data signal.

4.1. Specification of the quality of a data channel

Parameters which can be directly interpreted and easily measured to specify digital channel quality should be defined. One such parameter is the bit error rate, the ratio of the number of incorrectly received bits to the number of received bits. A second parameter is the bit loss rate, the ratio of the number of lost bits to the number of transmitted bits. This parameter refers primarily to the failure in data stream acquisition.

The two parameters can easily be measured by inserting pseudo-random sequences into the normal data format of the digital channel, or they can be calculated using a comprehensive mathematical model of a broadcasting television channel.

4.2. Measurement technique

The two parameters defined above may be measured with very simple equipment using pseudo-random sequences. Furthermore, the properties of these sequences provide, through an analysis of the error schemes, more complete information : it is possible to ascertain the exact time-distribution of the errors. For that purpose, the simple measurement equipment mentionned above is connected to a minicomputer which accomplishes the analysis. In the case of field trials, error schemes are recorded for later analysis.

4.3. Simulation

There is a double interest in using a theoretical model for computing the two parameters specifying the quality of a digital channel. First, this method permits predicting the ability of a transmission chain, whose analogue parameters are known, to carry the digital information. Secondly, this method allows isolating the analogue parameters vital to digital transmission and suggesting changes or adjustments to improve data channel quality.

4 .4. Performance

Due to the nature of the transmission, system performance vary widely depending on the reception site. In addition, the services using this system require the transmission of data of different nature, which are affected by errors transmission in different ways. It is thus advisable that any detecting or correcting code on the data be implemented by the users. On the other hand, it is essential to minimize information losses. Thus single error correction procedures are used in the header.

The first field trials conducted within the television service area give the bit error probability shown in figure 6.

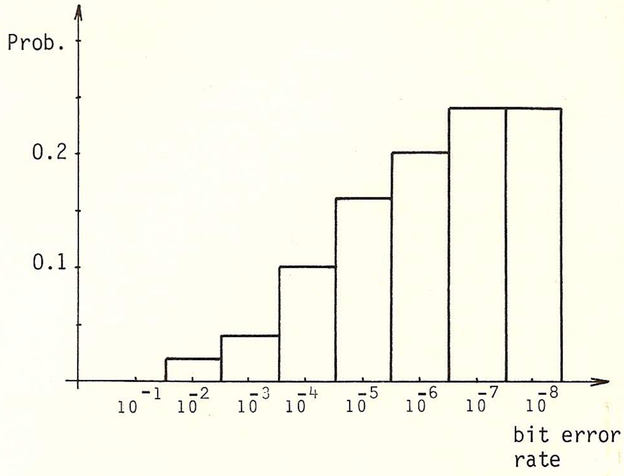


FIG. 6 Bit error rate probability

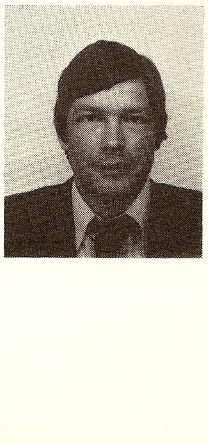
V CONCLUSION

The fundamental techniques used to design a packet broadcasting network have been presented. The use of existing television networks means that network growth can easily adapt to the increasing needs of its users. The substantial available data rates (up to about 4 M bit/second -see Table 1) and coverage on a nationwide basis add a new dimension to data communication.

TABLE 1

Comparison between L-SECAM and M-NTSC characteristics.
Examples of bit frequency and data packet size.

Characteristics	L-SECAM	M-NTSC
Number of fields and frames per second	50/25	60/30
Number of lines per frame	625	525
Channel bandwidth (MHZ)	8	6
Video bandwidth (MHZ)	6	4,2
Video-sound interval (MHZ)	+ 6,5	+ 4,5
Vestigial side band (MHZ)	1,25	0,75
Video modulation	positive	negative
Sound modulation	amplitude	frequency
Bit frequency (MHZ)	6,203125	4,358391
Packet size (header+data;octets)	8 + 32	8 + 20
Data rate with 1 line per field (bps)	12800	9600



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