REPORT 632-4

BROADCASTING-SATELLITE SERVICE (SOUND AND TELEVISION)

Technically suitable methods of modulation

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

(1974-1978-1982-1986-1990)

1. Introduction

There are a large number of possible methods of modulation for the broadcasting-satellite service, the choice depending on several factors. This Report discusses different technically suitable methods for the broadcasting-satellite service in all the frequency bands allocated to this service.

As regards the broadcasting-satellite service in the 12 GHz band, it should be borne in mind that the planning at the WARC-BS-77 was carried out on the assumption that a television programme would have only one sound channel, transmitted using a frequency-modulation sub-carrier positioned at the inter-carrier spacing of the television system used in the service area under consideration. Nevertheless, the WARC-BS-77 did not exclude the use of modulation signals having different characteristics, provided that the use of those different characteristics did not result in greater interference than that caused by the system considered in the Plan.

2. Sound broadcasting

This section concerns the methods of modulation suitable for sound broadcasting when this is the main service on the carrier(s) considered.

2.1 *Analogue methods of modulation*

Among analogue methods of modulation, it seems preferable to use frequency modulation with the same standards as those used for terrestrial sound broadcasting (see Recommendations 412 and 450); but they could be different in certain cases. In particular, it may be desirable to use a higher deviation, to reduce the necessary satellite transmitter power, especially in the frequency bands where new receivers or additional equipment for existing receivers, would in any case be required.

For stereophonic broadcasting using a frequency-modulation multiplex system (see Recommendation 450), it is necessary to increase by about 20 dB the values of field-strength, power flux-density, and satellite e.i.r.p. or the figure of merit of the receiving earth station. Stereophony could also use two identical channels, carrying the left and right signals, but there may be some problems for compatible monophonic reception.

2.2 *Digital methods of modulation*

For the broadcasting of a large number of sound channels, it may be advantageous to use TDM digital techniques. In this case, the choice of the modulation does not depend upon the nature of the signals to be broadcast, but only on the characteristics of the radio-frequency channel. In § 6 of this Report the choice of these digital modulation techniques for any kind of signal (e.g. sound, data, picture, etc.) is discussed.

Concerning the organization of the digital multiplex, several procedures may be used; they are described in Report 954. Details of digital coding methods are contained in Report 953.

Digital sound coding is described in Report 953 and subjective results related to C/N or bit error ratio are reported in the Annexes to this Report. It should be noted that sound quality is dependent upon the overall transmission process (multiplexing procedure, modulation type, bit interleaving, error protection strategy, demodulator characteristics, etc.) and upon the nature of the errors arising therein. Account must be taken of this when comparing results.

3. Analogue television with an associated sound channel

This section relates to the broadcasting of an analogue picture signal associated with a sub-carrier for one analogue sound channel.

The two types of analogue modulation best suited to satellite television broadcasting seem to be vestigial side-band amplitude modulation and frequency modulation.

For a given quality of service and a given figure of merit of the receiving installation, frequency modulation permits a much lower satellite transmitter power than amplitude modulation. However, in frequency bands for which there are existing terrestrial television receivers, amplitude modulation would allow these receivers to be used without modification. From the point of view of planning, frequency modulation requires wider channels, but the protection ratios are lower than for amplitude modulation, so either type of modulation may be advantageous, depending on the circumstances.

When frequency modulation is used, it is desirable that, after demodulation, the composite vision and sound signals should be the same as in the terrestrial service in the given geographical area; this would simplify the design of compatible receivers. This implies the use of a sub-carrier for the sound signal at a frequency equal to the spacing between the vision and sound carriers used for the terrestrial service. However, a sub-carrier of high amplitude can cause a visible beat pattern with the colour sub-carrier, and a buzz on the sound. Experiments by EBU members have shown that the receiver bandwidth need not be wider than is necessary to achieve a good quality of the picture alone, when the sound sub-carrier has an amplitude giving about 30% of the total peak-to-peak deviation of the carrier. Nevertheless, in some of these experiments the best signal-to-weighted-noise ratio which was achieved for the sound was 50 dB, as a result of buzz caused by variations in group delay of the receiver filter characteristics. If better sound quality is required (for example, with a signal-to-weighted ratio of 60 dB), it may be necessary to abandon the analogue sub-carrier principle, and to transmit the sound by other methods. One suitable method could consist of using a separate RF carrier with the same modulation characteristics as those which may be used for sound broadcasting from satellites.

In frequency-modulation television, the signal bandwidth limitation arising from radio-frequency and intermediate-frequency filtering, causes distortion which may significantly impair the picture quality. The most critical part of the system in this respect is the receiver; this must have cheap and simple filters, which may not be phase-corrected. In the absence of sub-carriers for the sound signals, the most critical distortions for a colour picture are the differential phase and gain of the colour sub-carrier. These distortions should be taken into account when deriving the relationship between the frequency deviation and the equivalent rectangular bandwidth of the receiver. Studies made by EBU members have shown that it is possible to obtain reasonable values of the distortions, as mentioned in Table III of Report 215, with a peak-to-peak frequency deviation of approximately 14 MHz/V at the reference frequency of the pre-emphasis characteristic, and a receiver bandwidth of 27 MHz. Studies carried out in Japan [CCIR, 1978-82a] with the 525-line M/NTSC system have shown that there are suitable combinations of carrier frequency deviations due to a video and a sound sub-carrier signal and the sub-carrier frequency deviation due to a sound signal which make it possible to obtain the required signal-to-noise ratios without producing a visible beat pattern and truncation noise.

4. Analogue television with several sound channels

This section describes the modulation methods which enable several sound channels and an analogue picture signal to be broadcast in the same radio frequency channel.

4.1 *Objectives*

There will probably be a need in the future for a capability within the satellite broadcasting channel for a number of sound channels beside the picture and, if possible, for using that capability flexibly for the emission of high quality sound (including stereophony), multilingual commentaries and even data or sound not directly related to the picture.

The systems used for this purpose in the 12 GHz band in Regions 1 and 3 will have to meet the requirements laid down by WARC-BS-77 relating, *inter alia*, to the occupied bandwidth and interference with other services; similarly, the decisions of the 12 GHz broadcasting satellite planning conference for Region 2 (RARC SAT-83) have to be followed. Furthermore, it is desirable that the standards for the sound accompaniment in broadcasting by satellite and terrestrial transmitters in the long term be brought into line (see Study Programme 47A/10).

The sound quality should be at least as good as that attained under the single FM sub-carrier system mentioned in § 3 above. The following objectives may be considered for high quality monophonic sound:

- audio bandwidth: 15 kHz;
- for analogue systems, a signal-to-noise ratio for 99% of the time of at least 50 dB, and, if possible, 60 dB as quasi-peak value with the weighting network described in Recommendation 468;
- for digital systems using companding to reduce the total bit rate, a dynamic range equivalent, for example, to that provided by a basic analogue to digital conversion at 14 bits per sample.

It would also be useful to envisage replacing one high quality sound channel by two or three commentary channels having the same signal-to-noise ratio but a smaller bandwidth.

Another important objective in designing broadcasting systems for the public is the cost of receiving equipment, which should be kept as low as possible. The same applies to any changes that may be required in cable distribution networks.

4.2 *Frequency multiplexing*

4.2.1 *Use of several analogue sub-carriers*

Frequency-multiplexing of several sub-carriers modulated by the sound signals results in a particularly economical arrangement for the receiver, at the same time needing only an insignificant or no increase in the width of the radio-frequency channel. However, intermodulation between the picture and sound signals, and between the various sound signals, may occur if the sub-carrier levels are not set properly within the common channel.

Regarding the technical basis of the 12 GHz plan for Regions 1 and 3, studies carried out in France and the Federal Republic of Germany [CCIR, 1978-82b and c] using the video characteristics of L/SECAM and G/PAL systems have shown that if the frequency separation between the sub-carriers is subject to a tight tolerance, the intermodulation products fall between the spectral lines of the picture signal. Tests carried out in the Federal Republic of Germany showed that the sub-carrier deviation may be increased to ± 63 kHz, thus improving the signal-to-noise ratio while retaining compatibility with existing receivers. In those countries where there is no compatibility problem with existing systems, the deviation may even be increased to ± 100 kHz. In this last case, measurements made in France demonstrated that the audio signal-to-noise ratio was 50 dB (CCIR quasi-peak weighting) in the presence of video and transmission noise. For these systems, the results of the measurements carried out with the OTS satellite are given in Annex I.

In Japan a two sub-carrier sound system for the 525-line M/NTSC system, has been studied by use of home receivers, with the sub-carrier frequencies at 5 and 5.5 MHz [CCIR, 1978-82d]. When the carrier frequency deviations by video signal and by each sound sub-carrier signal were set to be 17 MHz peak-to-peak and ± 1.3 MHz, respectively, and the sub-carrier deviation by sound signal was set to be ± 75 kHz, the signal-to-noise ratios achieved were 38 dB for video (unweighted), 59 dB for the first sub-carrier sound and 58 dB for the second sub-carrier sound (r.m.s., unweighted), without producing visible beat pattern and truncation noise at the carrier-to-noise of 14 dB.

If a two-carrier sound system were adopted in terrestrial television, it would be desirable that the sub-carrier frequencies in the satellite television system should be equal to the spacings between the vision carrier and the sound carriers in terrestrial television.

Studies carried out in Japan have shown that a second frequency modulated sub-carrier and pulse time multiplexing may be used to transmit up to six additional sound channels without increasing the bandwidth of the receiver [CCIR, 1974-78a].

4.2.2 *Use of a digitally modulated sub-carrier*

This section deals with the use of only one digitally modulated sub-carrier. The organization of the digital multiplex applicable to this case is described in Report 954. From the many possibilities, one that is of special interest is that of a single sub-carrier modulated by a digital multiplex with a bit rate of either approximately 700 kbit/s, 1400 kbit/s or 2100 kbit/s (i.e. the equivalent of two, four or possibly six high-quality monophonic sound channels). The feasibility of this type of system has been studied and experiments have been conducted by the BBC [Gilchrist, 1976; Kalloway, 1976], with a sub-carrier at 6.5 or 7 MHz, a bit rate of 700 kbit/s and four-phase PSK modulation. An analogous experiment was conducted in Italy, with four-phase PSK modulation of a sub-carrier at 7.5 MHz; the bit rate was 2.048 Mbit/s and continuous stream multiplexing was used [CCIR, 1978-82e]. Similar studies and tests were carried out in France and Sweden [CCIR, 1978-82f and g] with a sub-carrier at 6.656 MHz, a bit rate of 2 Mbit/s and continuous phase half-index frequency-shift keying modulation, obtained by residual sideband filtering of a two-phase modulation named "simplified MSK" [Amoroso and Kivett, 1977; Pommier and Veillard, 1979]. Theoretical and experimental studies have been undertaken by the EBU with a view to optimizing the modulation parameters and to provide complete specifications for this type of system. On the occasion of the tests carried out in France, a packet multiplexing procedure was tested.

In the United States a system is used in the FSS with a sub-carrier at 5.5 MHz and a 1.79 Mbit/s capacity, i.e., four 15 kHz high quality audio channels, or two stereophonic pairs, or one quadraphonic transmission. Somewhat similar approaches are used in video/audio encryption equipment for applications of satellites utilizing frequency modulation with an RF bandwidth of 36 MHz.

It should be noted, however, that the bandwidth of the baseband signal (video plus digitally modulated sub-carrier) may be, in certain countries, larger than the channel width provided in the television distribution network. This may lead to some difficulties when the signal is to be distributed in such networks.

Whatever may be the modulation process adopted, the sending-end and receiving-end digital sub-carrier filters are essential elements of the system. In the case of a single digital sub-carrier, the function of the sending-end filter is to limit the spectrum of the digital sub-carrier, in order to obviate any disturbance of the television signal. At the receiving end, the digital-demodulator filter must limit the noise band without introducing inter-symbol interference. In order to obtain the optimum performance, the sending-end and receiving-end filters cannot be designed separately. The introduction of a system having the optimum characteristics will therefore necessitate relatively precise characteristics for those filters, while seeking a simple and economical solution for the demodulator filter.

The search for a compromise between efficiency and simplicity of the digital demodulator has led to a preliminary conclusion that the simplified MSK modulation seems attractive for broadcasting with digital sub-carriers. The characteristics of the system judged to be optimum following the EBU studies are given in Annex I, for a service objective corresponding to a bit-error ratio of 10^{-3} at a carrier-to-noise ratio of 8 dB, allowing a 2 dB margin for implementation of the receiver [CCIR, 1982-86a].

A digital sound transmission system by means of four-phase DPSK modulation of a sub-carrier at 5.727272 MHz was adopted in Japan for its planned operational television broadcasting by the satellite BS-2. A bit rate of 2.048 Mbit/s and a frequency deviation of ± 3.25 MHz of the main carrier by the sub-carrier was selected. The specifications of this system are given in Report 1073. Experimental results obtained are given in Annex II of the present Report [CCIR, 1982-86b].

The United States commercial equipment referred to above uses QPSK modulation on the sub-carrier with 14-bit PCM with instantaneous companding, thus resulting in 11 bits plus sign. After adding parity bits at word lengths of 13 bits, the sub-carrier is injected -19 dB relative to video level and produces a bit error ratio less than 1×10^{-6} at $C/N = 13.5$ dB. Transmission quality is excellent for both terrestrial microwave (RF bandwidth = 20 MHz) and satellite (RF bandwidth = 36 MHz and main carrier peak deviation = 12 MHz) channels. Additional information is given in Report 488 and Report 215.

4.2.3 *Use of a digitally modulated sub-carrier plus a sub-carrier with analogue FM modulation*

The possibility of broadcasting simultaneously an analogue sub-carrier and a digitally modulated sub-carrier would make it possible, depending upon the nature of the receiver used, to offer two possibilities, namely:

- the possibility, which would lead to very economical receivers, of the reception of a television programme having a good quality monophonic sound channel, and
- the possibility of access, in addition, to several complementary services, such as high-quality stereophonic or monophonic sound channels, commentaries, sub-titles, teletext and additional sound programmes. The bit rate would have to be chosen so as to make possible at least the equivalent of four high-quality sound channels.

In order to attain those objectives, the analogue sound sub-carrier must make it possible to obtain good sound quality, while the bit rate of the digital sub-carrier must be at least 1.4 Mbit/s with a bit error ratio of less than 10^{-4} for a C/N ratio greater than 10 dB.

Studies in this field carried out in France and the United Kingdom with a bandwidth of 27 MHz and 625-line systems, have shown that all the foregoing conditions can be met, provided that the design of the receiver is such as to obviate any risk of disturbance of the picture by intermodulation products due to the presence of the two sub-carriers. It is, however, to be noted that spreading the spectrum of the digital sub-carrier renders such beats, when they occur, less visible than those generated by two analogue sub-carriers.

In the case of an analogue sub-carrier and a digital sub-carrier, supplementary constraints become evident at the level of the digital sub-carrier filters at the sending and receiving ends. In effect, the sending-end filter has, in this case, to ensure the adequate protection of the analogue sub-carrier, which necessitates a greater reduction of the spectrum of the digital sub-carrier. Similarly, the digital demodulator filter has to be narrow enough to ensure good separation between the digital sub-carrier and the analogue sub-carrier. For a given bit rate, these constraints render the search for a satisfactory compromise between the characteristics of the sending-end and receiving-end filters more difficult than in the case of a single digital sub-carrier.

4.3 *Time-division multiplexing*

The procedures for setting up a digital multiplex may be similar to those foreseen for a digitally modulated carrier of sub-carrier systems (see Report 954).

4.3.1 *Baseband insertion of digital audio signals in the line-blanking interval using full-response coding*

The insertion of digital audio signals in the line-blanking interval is an attractive technique because it enables high-quality signals to be transmitted without increasing the width of the baseband or the RF channel.

The use of this technique in terrestrial television is the subject of Report 958. Certain systems are also described in Report 488 and in [CCIR, 1970-74a]. Report 958 states in particular in the case of B/PAL and M/NTSC (Japan) television systems, that there is little hope that any system employing digital signals in the line-blanking interval will be compatible with existing receivers, even if only half the total capacity of the line-blanking interval is used. Moreover, new studies are necessary in the case of other television systems. On account of this compatibility problem, Report 958 concludes that it will be necessary to introduce new types of terrestrial television receivers if it is desired to use the line-blanking interval to transmit up to four high-quality sound channels.

This technique is more appropriate for satellite broadcasting where compatibility with existing receivers is not as critical as in the case of terrestrial television. This is because the signal processing, including the regeneration of the synchronizing pulses, could be performed in the converter which must in any case be added to television receivers. Furthermore, multipath propagation, which might cause impairment to the received picture when this system is used with terrestrial television, will not occur in the case of satellite broadcasting.

The insertion at baseband of a digital audio and data multiplex signal during the line blanking interval has been fully developed in the B-MAC system described in Report 1073. The instantaneous bit rate for the 525-line B-MAC signal is either 14.3 Mbit/s using a 4-state coding technique or 7.15 Mbit/s using a 2-state coding technique. Such 4-state and 2-state FSK modulation allows signal detection through a simple FM demodulator. The corresponding data transmission capacity is 1.57 Mbit/s or 0.785 Mbit/s providing for up to 3 or 6 high quality audio channels respectively and a utility data channel of 9600 bit/s or 4800 bit/s protected with 5:1 majority logic. This sound and data multiplex does not require more bandwidth than the accompanying vision MAC signal. Australia has adopted the 625-line B-MAC system providing for a capacity of either 1.594 Mbit/s or 0.797 Mbit/s using the 4-state or 2-state coding respectively. This provides for up to 6 or 3 high quality audio channels respectively.

In both versions of the B-MAC format, adaptive delta modulation (ADM) with a sampling rate of 204 kHz (13 times line frequency) is used for high quality sound coding to be able to fit the required number of channels. Report 953 gives a detalied description of this adaptive delta modulation coding, and gives results of subjective measurements conducted in Australia, Canada and the United States. Results show that with appropriate error concealment as described in Report 1073, the service failure ($Q = 1.5$) occurs at a bit error ratio of 15×10^{-2}. Results also show that in a non-impaired transmission channel, ADM will give similar performance to the 14-10 semi-instantaneous companding scheme described in Report 953 except in the case of critical programme material. It is also found that an increase of the sampling rate for ADM will bring it closer to the performance of the semi-instantaneous companding scheme for all types of programme material.

This time multiplexing at baseband avoids intermodulation with the vision signal and degradation in threshold performance caused by the usual sound sub-carriers. It also allows for wider carrier deviation within the RF channel and permits the use of a simple vision FM demodulator to recover vision, sound and data channels.

Tests have been carried out within the framework of the EBU on a system of this type in association with a PAL television signal, using an instantaneous bit rate equal to twice the colour sub-carrier frequency and with digital synchronization inserted in the field-blanking interval. The available capacity is then 1.625 Mbit/s, which is equivalent to four high-quality sound channels. Taken alone, this system satisfies the service continuity criterion requiring a bit error ratio of 10^{-3} at a carrier-to-noise ratio of 8 dB. However, this criterion is not satisfied if a digital sub-carrier system of the type described in Annex I is added in the hope of increasing the capacity to a value close to 3.5 Mbit/s.

4.3.2 *Baseband insertion of digital audio signals in the line-blanking interval using partial response coding (duobinary)*

Two systems combining MAC image coding with TDM baseband multiplexing of the digital signals have been studied in Europe. These systems, Known as "D-MAC/packet and D2-MAC/packet", employ baseband multiplexing (type B) with duobinary coding. For the D-MAC/packet system a gross bit rate of 3.28 Mbit/s is obtained from 10 μs bursts, and an instantaneous bit rate of 20.25 Mbit/s. For the D2-MAC/packet system a gross bit rate of 1.64 Mbit/s is obtained from 10 μs digital bursts, and an instantaneous bit rate of 10.125 Mbit/s. For both systems the other coding and multiplexing characteristics are common to the systems of the MAC/packet family.

The D-MAC/packet and the D2-MAC/packet system possesses the original feature of being able to adapt to the physical characteristics of the transmission media. This type of flexibility results from the TDM baseband multiplex concept associated with the use of a duobinary code for the transmission of the digital signal. The D-MAC/packet system combines optimum use of the broadcasting-satellite channel and compatibility with channels used in cable networks of at least 12 MHz bandwidth. The D2-MAC/packet system appears as one of the best compromises between the optimum use of a broadcasting-satellite channel and direct compatibility with the 7 or 8 MHz bandwidth channels used in cable networks.

For the D2-MAC/packet system in satellite broadcasting, the baseband signal is transmitted with a bandwidth of at least 8.4 MHz. Thus a picture signal complying with Report 601 does not undergo any passband reduction and the digital signal can meet the continuity of service criterion, corresponding to a bit error ratio of 10^{-3} for a carrier-to-noise ratio of 8 dB.

For the D-MAC/packet system in satellite broadcasting, the baseband signal is transmitted with a nominal bandwidth of 10 MHz. Thus a picture signal complying with Report 601 does not undergo any passband reduction and the digital signal can meet a continuity of service criterion, corresponding to a bit-error-ratio of 10^{-3} for a carrier-to-noise ratio of 8 dB when the appropriate passband filtering and error-reduction techniques are used in the receiver.

Both systems have the following other main advantages:

- possibility of increasing the video bandwidth in the case of an adaptive filter demodulator for C/N ratios greater than the frequency demodulation threshold (about 11 dB);
- no intermodulation between baseband signal components;
- demodulation of the entire signal by the same demodulator;
- the possibility of reducing the bit error ratio and picture impulsive noise by optimized filter bandwidths;
- the possibility of using digital processing of the duobinary coded signal to reduce the effect of bit errors;
- flexible hence evolutive organization of the multiplex as a whole.

Further advantages of the D-MAC/packet system are as follows:

- direct compatibility with all transmission channels with a baseband equal to or greater than 9 MHz;
- highest practical data transmission rate of 20.25 Mbit/s;
- continuous broadcasting of data at a rate of 20.25 Mbit/s by elementary D-MAC/packet multiplex.

Further advantages of the D2-MAC/packet system are as follows:

- direct compatibility with all transmission channels with a baseband equal to or greater than 4.5 MHz;
- continuous broadcasting of data at a rate of 10.125 Mbit/s by elementary D2-MAC/packet multiplex.
- with the use of a demodulator with adaptive filtering, a bit error ratio of 10^{-3} can be reached for a carrier-to-noise ratio of 6.5 dB.

In view of these advantages, the full specifications of the D-MAC/packet and D2-MAC/packet systems have been established for satellite broadcasting at 12 GHz with a 625-line standard (see Report 1073). Report 634 gives the noise sensitivity measurement data [CCIR, 1982-86c and CCIR, 1986-90a].

U.K. industry and broadcasting laboratories have carried out comprehensive joint experiments on the D-MAC/packet system. These experiments used the full range of media; satellite, cable and terrestrial radio-relay links and have confirmed the practicability and economics of receiver implementation. The results for satellite broadcasting are summarized in Annex IV to the present Report.

European industry and the CCETT Laboratories have carried out numerous joint experiments on the D2-MAC/packet system. These experiments were performed using numerous media: satellite, cable and land radio-relay networks. The results for satellite broadcasting are summarized in Annex III to the present Report.

4.3.3 *Radio-frequency time-division multiplex using the line-blanking interval*

The modulation is analogue (frequency modulation) during the active line period and digital during the line-blanking interval. This type of system offers a capacity close to 3 Mbit/s (equivalent to eight high-quality sound channels) and it is naturally compatible with the coding of time-compressed and time-multiplexed image components (MAC) [Lucas and Windram, 1981]. Additional equipment is needed at the receiver if transcoding to existing receivers is required, but taken as a whole the receiver is not more complex than those needed for other systems using digital sound modulation. It is not possible to use this system in terrestrial television, but simple transcoding can be effected to a type B system (see § 4.3.1) of lower capacity for which most of the circuitry of a type C receiver is re-used.

The main properties of the radio-frequency time-multiplex system are the following:

- high capacity for digital sound and data signals with good error performance; in particular, it is possible to satisfy the service continuity criterion corresponding to a bit error ratio of 10^{-3} for a carrier-to-noise ratio of 8 dB;
- potential for increased video bandwidth;
- simple video filtering;
- no degradation of the frequency demodulation threshold;
- no intermodulation problems;
- one-step demodulation of the digital signal;
- in principle, easy conversion to suit cable distribution*; and
- can be made fully compatible with continuous data transmission.

In view of these advantages, system C has been fully specified in association with a MAC picture and a packet multiplexing system for satellite broadcasting at 12 GHz with 625-line standards (see Report 1073). Measurements of sensitivity of interference are given in Report 634.

The EBU has conducted numerous tests with a type C system associated with a MAC television picture signal (see Annex V).

For this system, preference is accorded to 2-4 PSK modulation [Duponteil, 1981] in which the phase shift is instantaneous and equal to 90° for each bit (before filtering), as this offers the following advantages:

- suitable spectrum shape and low spectrum spreading after passage through a non-linear device such as a satellite repeater;

* It should be noted, however, that if the full capacity is required, the bandwidth may be, in certain countries, larger than the channel width provided by the channelling arrangement in the cable network. This may lead to some difficulties when the signal is to be distributed in such networks.

- simple receiving equipment with good performance for differential demodulation* ; and
- possibility of using the same FM demodulator for vision and sound in areas of high signal strength.

5. Digitally modulated radio-frequency carrier

This section relates to a modulation system in which the radio-frequency carrier is directly modulated by a digital bit stream. It concerns the broadcasting of all signal types: sound, pictures, data, etc.

Digital encoding of television signals, as well as data-compression techniques for picture information redundancy reduction are currently under intensive study and investigation (see Report 629). A broadcasting link using direct carrier modulation by the digitized video represents another alternative to analogue/FM modulation.

Digital modulation has potential advantages over analogue/FM, including the possibility of lower satellite transmitter power and narrower channel bandwidth requirements if a sufficiently low bit rate can be achieved.

While this approach would be currently too expensive to implement for individual reception, the cost may not be prohibitive for community reception [CCIR, 1974-78b]. It is also likely that decoder hardware, once standardized, will show the same dramatic decrease in cost as has occurred with other digital hardware such as computers and calculators.

5.1 Modulation techniques

The type of modulation must be selected as a function of criteria such as spectrum congestion, noise and interference immunity. In the case of individual reception, the choice must also take into account the simplicity and cost of the demodulator. The modulation methods which appear to be suitable include two- or four-phase PSK modulation, continuous phase half-index frequency shift-keying and frequency shift-keying using the principles of partial response coding. The last-named offers the advantage of a narrow power spectrum associated with a constant envelope [CCIR, 1978-82h and i]. Among the modulation processes of this type are included, notably, the half-index duobinary FSK and tamed FM processes [de Jager and Dekker, 1978], which may be considered as derivatives of MSK modulation, wherein the phase transitions are rendered interdependent by the nature of the code employed. These processes also have many features in common with the four state phase-shift keying with offset-streams modulation processes, such as the offset QPSK [Gronemeyer and McBride, 1976]. In fact, the half-index duobinary FSK and offset QPSK processes are both four-state phase-shift and offset-stream processes; the so-called tamed FM process, on the other hand, may be considered as half-index duobinary FSK modulation in which there is a higher level of correlation between the phase-transitions of the emitted signal. All these similarities are important, because they suggest that, at the demodulation level, there is considerable compatibility among these forms of modulation. This is particularly interesting in the case of broadcasting, wherein the effect of the existence of large numbers of terminals, when new systems involving different radio-electric constraints are being defined, is well known.

5.2 Interference

A constructive study using digital technique [CCIR 1978-82j; Pommier and Siohan, 1981] gives some results based on QPSK and MSK modulation, when digital modulation is used on a carrier of the frequency plan of the 12 GHz band (Regions 1 and 3). One of the points studied concerns the protection of adjacent channels, which has to be effective even in the presence of spread spectrum phenomena due to the non-linearities of the satellite power tube used in near-saturation conditions. Indeed for MSK and QPSK, the adjacent channel interference seems to be the main reason for the limitation of the usable bit rate.

Two distinct phenomena affecting the analogue FM channel suffering interference must be considered separately:

- the first phenomenon is the impairment of the signal received above the demodulator threshold;
- the second phenomenon is the apparent shift of the demodulator threshold, at a given thermal noise power N, to higher carrier-to-noise (C/N) values.

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* The theoretical degradation of differential demodulation as compared to coherent demodulation corresponds to an increase in carrier-to-noise ratio of 1.1 dB for a given bit error ratio of 10^{-3}.

This study has shown that interference into the FM adjacent channels may lead to an impairment manifested mainly by a shift of the threshold to higher values of the carrier-to-thermal-noise ratio. The limitation of the bit rate due to this phenomenon depends on the tolerated increase in threshold. For a threshold increase kept down to about 0.15 dB, usable bit rates with QPSK and MSK are 26 Mbit/s and 20 Mbit/s respectively. If a greater threshold increase is accepted, for example of the order of 0.3 dB, bit rates of 34 and 27 Mbit/s may be used with QPSK and MSK modulation respectively, these values were obtained with a receiving filter of width 27 MHz and conventional frequency demodulator, for the demodulation of the FM carriers suffering interference. Other conditions such as for example the use of demodulators with threshold extension would no doubt give very different results.

For both MSK and QPSK, with coherent demodulators and an equipment margin of 2 dB, a bit error ratio of 10^{-4} may be obtained for a power flux-density of -107 dB(W/m^2) with a bit rate around 21 Mbit/s, at a receiving station figure of merit (G/T) of 6 dB(K^{-1}).

A recent study [Newland, 1988] has established that when digitally modulated signals mutually interfere, the degree of mutual interference which may be tolerated is substantially greater than is the case for analogue signals carrying broadcast-quality pictures or sound. The permissible carrier-to-interference ratio (C/I) depends on the method of modulation and error-correction coding (if any). There is also scope for trade-off between C/N and C/I in the overall link budget. Further information is given in Report 634.

5.3 Advanced modulation and coding methods

Modern trends in digital communication by satellite use forward error-correction (FEC) by concatenation of block codes and convolutional codes or trellis modulation. Soft decision Viterbi decoding is used in the receiver. High coding gains are achievable, allowing a reduction of satellite power and the service outage times. The hardware implementation of these techniques at high bit-rates (e.g. 140 Mbit/s) has provided recently an adequate solutions for professional applications. A comparison of the performance of some digital systems for satellite transmission is given in Table [I]. The Eb/No values have been obtained by computer simulations and laboratory tests [Cominetti and Morello, 1989] and from the existing literature [Seo and Feher, 1988]. Convolutional and trellis coding schemes offer high coding gains. However, to overcome the effect of error bursts at the output of the Viterbi decoder, interleaved BCH code or Reed Solomon code may be used.

TABLE I - Performance of some modulation and coding systems via satellite

System	Modulation	FEC 1	FEC 2	Eb/No AT B.E.R. of 10^{-8}	RELATIVE SPECTRAL EFFICIENCY (%)
1	QPSK	-	-	15.7	100
2	QPSK	BCH(255,239,2)	-	11.2	94
3	QPSK	CONVOL. 3/4	BCH(255,239,2)	7.4	70
4	QPSK	CONVOL. 1/2	BCH(255,239,2)	6.2	47
5	8PSK	BCH(255,239,2)	-	16.4	141
6	8PSK	TRELLIS 2/3	BCH(255,239,2)	8.2	94
7	16SQAM	BCH(255,239,2)	-	16.0	187
8	16SQAM	TRELLIS 3/4	BCH(255,239,2)	12.1	141

6. Other consideration

6.1 Formulae governing modulation performance for analogue TV and audio signals

6.1.1 Video modulation only

In a frequency-modulation system:

$$S/N = C/N + F_{dB} + k_w$$

where:

S/N: ratio of peak-to-peak luminance amplitude to weighted r.m.s. noise (dB)

C/N: pre-detection carrier-to-noise ratio in the radio-frequency bandwidth (dB)

F: $3(D_{p\text{-}p}/f_v)^2 \cdot (b/2f_v)$ (power ratio which equals F_{dB}, when expressed in dB)

$D_{p\text{-}p}$: peak-to-peak deviation by video signal (including synchronization pulses)

f_v: highest video frequency; (e.g. 4.2 MHz in the case of System M)

b: radio-frequency bandwidth (usually taken as $D_{p\text{-}p} + 2f_v$)

k_w: combined de-emphasis and weighting improvement factor in frequency modulation systems (dB) (see Table II).

6.1.2 Audio modulation only (low level audio FM sub-carriers)

The unweighted signal-to-noise ratio of accompanying audio channels, which consist of FM sub-carriers located above the video baseband, is determined by the following equation:

$$S/N_a = 10 \log \left[\frac{3}{4} \left(\frac{b}{f_a}\right) \left(\frac{D_s}{f_s}\right)^2 \left(\frac{D_a}{f_a}\right)^2\right] + \left(\frac{C}{N}\right) + k_a$$

where:

S/N_a : audio channel r.m.s. signal to r.m.s. noise ratio (dB);

D_s : peak deviation of the main carrier by the sub-carrier (MHz);

D_a : peak deviation of the sub-carrier by the audio (MHz);

f_s : frequency of the sub-carrier (MHz);

f_a : highest audio frequency (MHz);

C/N : pre-detection carrier-to-noise ratio (dB);

k_a : combined improvement factor due to pre- and de-emphasis for the audio channel (dB). (See CMTT Report 496, Table II, for improvement factors corresponding to various audio channel baseband bandwidths);

b : pre-detection RF bandwidth (MHz) defined by equation (1) of § 6.1.3.

6.1.3 Combined video and audio (low level FM sub-carrier) modulation

Combined video and audio FM sub-carrier signals form a composite baseband signal as illustrated in Fig. 5. The required RF bandwidth is approximated by the following equation:

$$b = D_{b_{p\text{-}p}} + 2f_b \qquad (1)$$

where $D_{b_{p\text{-}p}}$ is the peak-to-peak deviation of the carrier by the composite baseband signal.

It is not certain how the deviation of a high level video signal should be combined with the individual deviations of a multiplicity of low-level audio signals. Further study and measurements are required on this subject especially on the effect of additional deviation caused by the multiple sound sub-carriers on the video performance.

However it can be assumed for practical purposes that the overall peak-to-peak deviation can be approximated by the peak-to-peak deviation due to the video signal only, provided the individual deviations of the few audio channels are small in comparison, thus:

$$D_{b_{p\text{-}p}} \approx D_{p\text{-}p} \qquad (2)$$

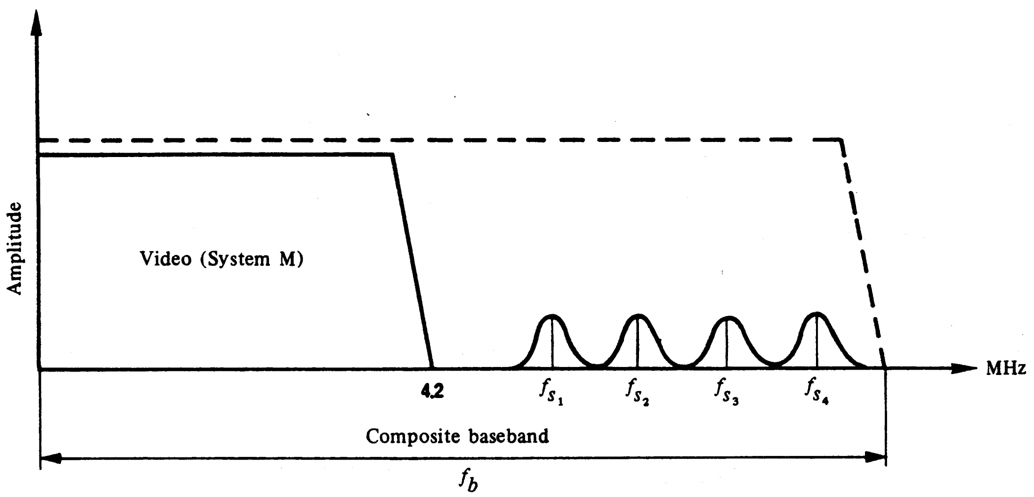

FIGURE 5 – *Composite video and multiple FM sub-carrier baseband*

Substitution of equation (1) and approximation (2) into the equation in § 6.1.1, and combining constants, results in the following equation for the weighted video signal-to-noise ratio:

$$\frac{S}{N_v} = 10 \log \left[\left(\frac{D_{p\text{-}p}}{f_v}\right)^2 \left(\frac{D_{p\text{-}p} + 2f_b}{f_v}\right)\right] + \frac{C}{N} + k_e + 1.8$$

where:

k_e: combined noise weighting and pre-emphasis advantage (e.g. 12.8 dB for System M/Canada, United States).

This equation can be used to estimate the effect of multiple audio FM sub-carriers on the signal-to-noise ratio of the video signal in FM transmission systems operating above threshold.

TABLE II – *Video-frequency noise weighting-network reduction factor for monochrome television*

System	Weighting (dB)		Weighting including de-emphasis, k_W (dB)
	White noise	Triangular noise	Triangular noise
B, C, E, F, G, H and M (Japan)	8.5	16.3	16.3
D, K, L	9.3	17.8	18.1
I	6.5	12.3	12.9
M (Canada, USA) (1)	6.8	10.2	13.8

(1) Weighting factors for 525-lines System M (Canada, USA) are based on Recommendation 567. (Values according to Report 637).

Note – When using pre-emphasis according to Recommendation 405, the combined effect of weighting and de-emphasis for triangular noise is approximately the same as that of weighting alone. More details are given in Report 637.

6.2 Analogue component TV signals

Future television receivers are expected to have an input socket for component signals (Y, U, V or R, G, B) and it may be possible to exploit this feature by transmitting the signal in component form. This could have important advantages in the future development of systems.

Analogue component TV signals are the subject of Report 1073.

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[1974-78]: **a.** 11/41 (Japan); **b.** 11/90 (USA).

[1978-82]: **a.** 11/81 (Japan); **b.** 10-11S/17 (France); **c.** 10-11S/7 (EBU); **d.** 10-11S/112 + Corr.1 (Japan); **e.** 10-11S/161 (Italy); **f.** 10-11S/124 (EBU); **g.** 10-11S/126 (Sweden); **h.** 10-11S/18 (France); **i.** 10-11S/30 (USA); **j.** 10-11S/137 (France).

[1982-86]: **a.** 10-11S/42 (EBU); **b.** 10-11S/6 (Japan); **c.** 10-11S/165 + Corr.1 (EBU).

[1986-90] : a. 10-11S/14 (UK).

ANNEX I

STUDIES AND TESTS CARRIED OUT BY THE EBU ON BROADCASTING OF SEVERAL SOUND CHANNELS WITH A 625-LINE ANALOGUE TELEVISION SIGNAL USING ONE OR TWO SUB-CARRIERS

1. Television with two analogue sub-carriers

Tests carried out in France and Germany using the video characteristics of L/SECAM and G/PAL television systems (625 lines) with the following parameters:

- Sub-carrier frequencies: 5.5 MHz and 5.75 MHz
- Frequency deviation of sub-carrier: ± 63 kHz,
- Carrier deviation by the picture signal: 14 MHz/V,
- Deviation of the carrier by each of the sub-carriers: ± 2.8 MHz.

TABLE III – *Typical measurement results*

C/N (dB)	Weighted *S/N* picture channel (dB)	Weighted quasi-peak *S/N* 1st sound channel (5.5 MHz) (dB)	Weighted quasi-peak *S/N* 2nd sound channel (5.75 MHz) (dB)
14	44	45.8	47.8

Similar measurements have shown that, with a ± 100 kHz peak-to-peak deviation of each of the sub-carriers, the weighted quasi-peak signal-to-noise ratio may attain 50 dB in each of the sound channels, with a carrier-to-noise ratio of 14 dB.

The conclusion of the tests carried out in France is that the satellite broadcasting of a TV signal accompanied by two high quality analogue sound signals in a 27 MHz channel is possible provided that the receivers' IF filters are within certain group-delay range and have a peak-to-peak tolerance of the order of 16 ns. Under these conditions, the picture is very slightly disturbed by the presence of two sound-sub-carriers and the CCIR-weighted quasi-peak sound signal-to-noise ratio may attain the value of 50 dB.

2. Television with a digitally-modulated sub-carrier (Type A System)

The EBU studies on this type of system have led to consideration of the following characteristics as being optimum in the case of 625-line television standards:

2.1 *Basic assumptions*

2.1.1 *Modulation characteristics applicable to 625-line television systems in a 27 MHz satellite channel*

– transmitted video bandwidth	6 MHz*
– carrier deviation produced by 1 V of video signal**:	13.5 MHz
– pre-emphasis for PAL and SECAM	according to Recommendation 405
– energy dispersal	600 kHz (related to field rate)

2.1.2 *Bit error ratio corresponding to the continuity limit of the sound services:* 10^{-3}

2.1.3 *C/N ratio corresponding to the continuity limit of the sound services:* 8 dB (27 MHz)

2.1.4 *Bit-rate:* a multiple of the sampling frequency of 32 kHz (in accordance with the recommendation proposed by the EBU and given in Report 953)

2.2 *Characteristics of the digitally-modulated sub-carrier*

2.2.1 *Type of modulation:* vestigial sideband two-state phase-shift keying (VSB-2-PSK) with transmission of the upper sideband and coherent demodulation

2.2.2 *Bit rate:* 2.048 Mbit/s***

The stability of the broadcast binary signal should ensure:

- long-term clock-frequency stability of 10^{-6},
- maximum clock jitter of 5 ns r.m.s.

2.2.3 *Sub-carrier frequency*****

The frequency of the original carrier of the 2-PSK modulation (f_1) is 6.5 MHz (6.5 MHz = 416 times the television line frequency).

The long-term stability of this carrier frequency f_1 should be at least 10^{-6}.

The centre frequency of the transmitted spectrum of the sub-carrier signal (f_0) is 7.012 MHz, i.e.:

$$7.012 \text{ MHz} = 6.5 \text{ MHz} + \frac{2.048}{4} \text{ MHz}$$

---

* This value permits the use of a version of the image component coding system using time-compression and time-multiplexing.

** The deviation quoted is for a 1 V peak-to-peak sinusoidal signal at a frequency of 1.52 MHz.

*** This value corresponds to five or six high-quality sound channels, depending on the type of multiplexing used.

**** The description of the modulated sub-carrier requires that two parameters be defined: the frequency of the original carrier of the 2-PSK modulation f_1, and the centre frequency f_0 of the spectrum obtained after vestigial sideband filtering.

2.2.4 *Deviation of the main carrier by the digitally-modulated sub-carrier:* 2.5 MHz r.m.s.

2.2.5 *Coding between the binary signal and the digitally-modulated sub-carrier*

An absolute code is used between the binary signal and the two-state phase modulation (this code requires the use of an ambiguity resolution device in the demodulator).

2.3 *Experimental results*

The following is a summary of the results of numerous tests that have been done with the type A system described above, used in conjunction with a SECAM vision signal and a packet multiplex of 5 high-quality audio channels (coded with near-instantaneous companding and with error-protection using one parity bit covering the 5 most significant bits of each sample).

2.3.1 *Bit error ratio:* see Fig. 2 below.

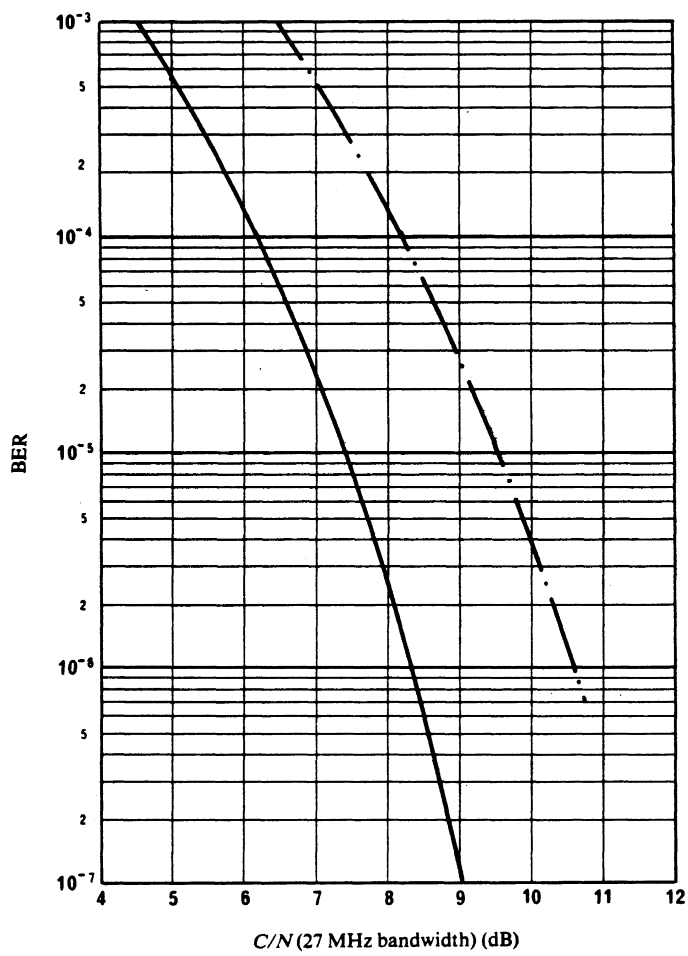


FIGURE 2 – *Typical values of bit error ratio (BER) with a video signal carrying 75% colour bars*

———— theoretical characteristic (Gaussian white noise)

.——.——. typical measured characteristic (from laboratory experiments and experiments with OTS)

2.3.2 *Vision and sound quality*

Subjective tests have been conducted with a group of 37 observers; they used the single stimulus method for the vision and conformed to the provisions of Recommendation 562 for the sound. The laboratory measuring apparatus included a satellite simulator with a non-linear element. The pictures were derived from EBU test slides. The sound was evaluated independently of the vision, using two musical extracts considered to be sensitive to faults and errors in digital systems (a Haydn trumpet solo and Japanese theatre music). The results are shown in Fig. 3 on the 5-grade quality scale. It will be seen that the best vision quality was 0.65 grade below the quality provided by the 4:2:2 digital television standard of Recommendation 601.

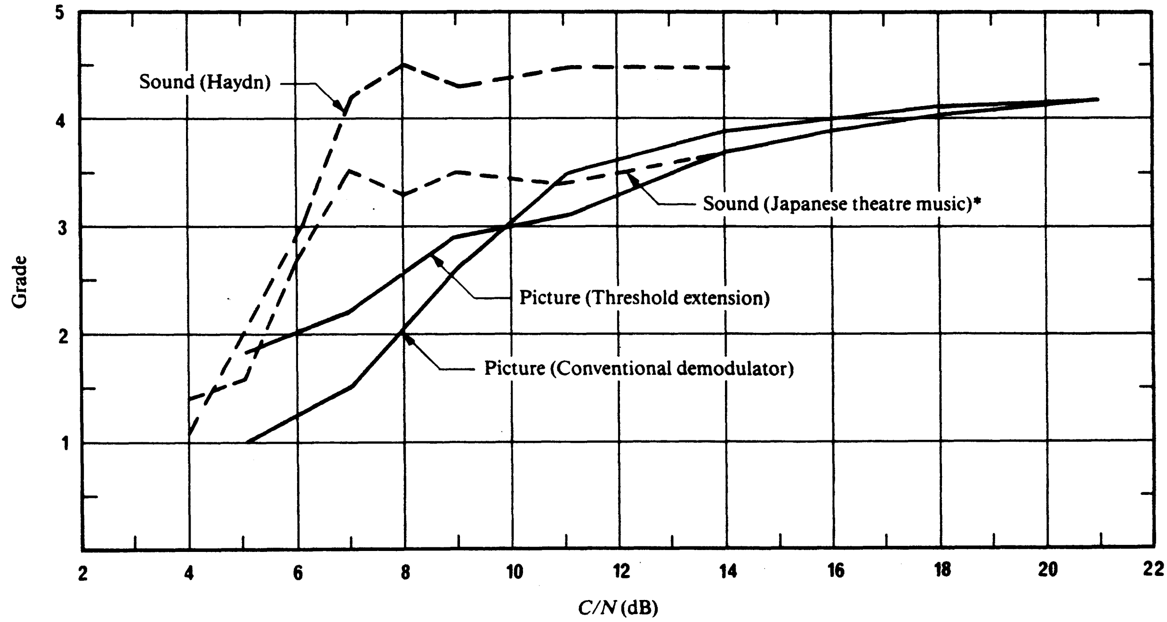


FIGURE 3 – *Quality as a function of C/N ratio for the A/SECAM system (37 observers)*

* The EBU experts involved in the relevant subjective tests are of the opinion that the degradation of Japanese music at carrier-to-noise ratios above 8 dB with reference to Haydn music was due to an overload of the A/D converter at the modulation input.

2.3.3 *Vision and sound failure points*

Taking the failure point to correspond to quality grade 1.5, subjective tests in which the vision and sound programmes were presented simultaneously, and were obtained using guitarists in the studio gave the following results in terms of carrier-to-noise ratio.

Vision

Normal demodulator: 5 dB

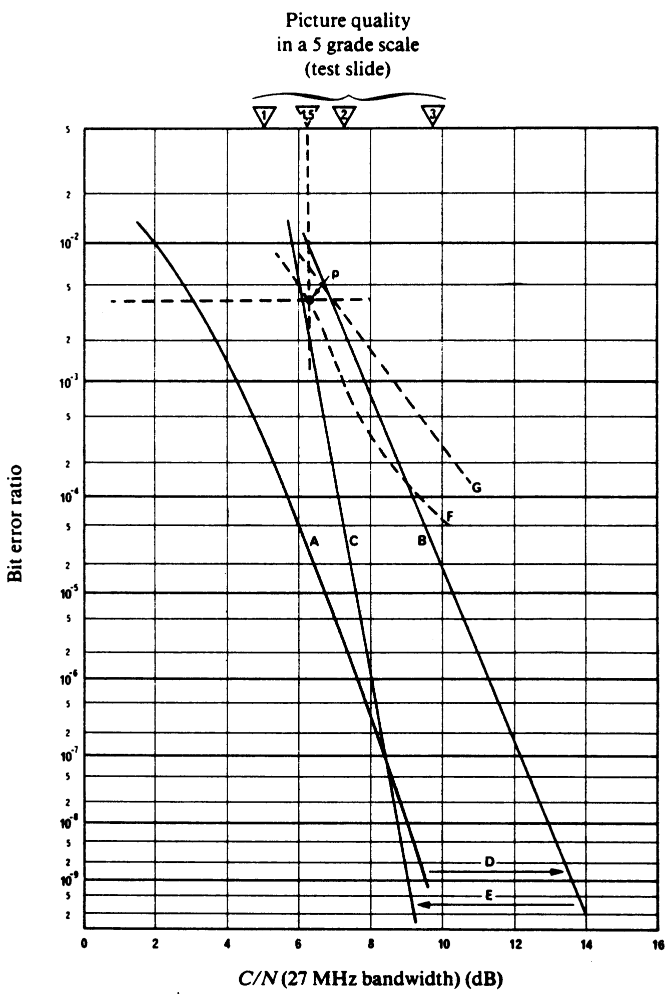
Threshold-extension demodulator: 4 dB

Sound: 4 dB

It should be noted that differences from results given in Fig. 3 are due to the test methodology and evaluation criteria.

ANNEX II

EXPERIMENTAL RESULTS FOR THE DIGITAL SUB-CARRIER SYSTEM INTENDED FOR USE WITH TELEVISION SYSTEM M (JAPAN)



Relation between bit error ratio and C/N for the digitally-modulated sound sub-carrier system

Maximum video signal frequency: 4.5 MHz
Frequency deviation of main carrier by video signal: 17 MHz peak-to-peak
Sub-carrier frequency: 5.727272 MHz
Frequency deviation of main carrier by sub-carrier: ±3.25 MHz
Transmission bit rate : 2.048 Mbit/s

——————— *C/N* versus bit error ratio
A theoretical value (Nyquist)
B experimental value, without error correction and interpolation
C experimental value, with error correction and interpolation

— — — — bit error ratio when the subjective quality of sound is equal to that of picture (with 14/10 bits near-instantaneous companding)
F test slide (woman/sea-side)
G colour bar

D degradation from theoretical value to experimental value
E improvement of bit error ratio by error correction and interpolation technique
p bit error ratio of 4×10^{-3} corresponding to grade 1.5 on a 5 grade scale without error correction and interpolation

ANNEX III

EXPERIMENTAL RESULTS FOR THE TDM BASEBAND MULTIPLEXING SYSTEM

"D2-MAC/packet" (type B system using duobinary coding)

This Annex summarizes the results of numerous laboratory tests carried out in France with broadcasting-satellite simulation equipment having the following characteristics:

- input filter (including the earth-station transmitting filter, assuming linear mode operation):
  - -3 dB bandwidth 34 MHz
  - parabolic distortion of group-delay time in 27 MHz bandwidth 20 ns
- TWT amplitude-phase conversion 5°/dB

  The output filter, with a width of 50 MHz, introduces no measurable distortion in the signal.
- the frequency demodulator is of a conventional or threshold-extension type.

1. Bit error ratio

In view of the precoding effected on transmission (modulo 2 division of the binary signal by the duobinary code $(1+2)$ generating polynomial), the bit errors are independent.

The curves of Fig. 4 show the error rate measured by means of the simulation model, where:

Curve A: conventional frequency modulator,

Curve B: adaptive filter threshold-extension frequency demodulator.

The principle of the threshold-extension frequency demodulator used for these tests is shown in Fig. 5. With the device as described, the IF filter retains a bandwidth of 27 MHz for carrier-to-noise ratios higher than 11 dB, a value which corresponds to the frequency demodulation threshold. Below 11 dB, the IF filter bandwidth is gradually reduced to about 14 MHz.

Tests performed in similar conditions in the United Kingdom, i.e., with two IF bandwidths, provide results which corroborate those of Fig. 4.

Further improvements of about 0.5 dB may be obtained by the use of the Viterbi decoder [Alard, 1986 a and b; Jankowiak, Lamnabhi and Arragon, 1985].

2. Sound and picture quality with the D2-MAC/packet system

Subjective tests have been carried out in France by observer groups consisting mostly of experts. These tests, together with the common features of the C-MAC/packet and D2-MAC/packet systems described below, were used to plot the sound and picture quality curves. To avoid a proliferation of data, the results given are confined, for the D2 system, to the quality curves corresponding to those of Annex V for the C-MAC system.

2.1 *Description of features common to the C-MAC/packet and D2-MAC/packet systems*

- *Picture:* The two systems are identical with regard to picture transmission:
  - same coding,
  - same pre-emphasis,
  - same frequency deviation,
  - same energy dispersion.

  Hence any quality differences in the picture signal can derive only from differences in the behaviour of the frequency demodulator.
- *Sound:* The two systems differ only with regard to bit rate, digital signal coding and modulation characteristics. However, in view of the interlacing of bits and error distribution, the test results established that the same error rate for the two systems provides the same sound quality for a given coding law and a given level of protection.

2.2 *Conventional demodulator*

The curves of picture and sound quality, quasi-instantaneous coding and parity bit protection with a conventional frequency demodulator are given in Fig. 6.

2.3 *Threshold-extension demodulator*

The picture and sound quality is given in Fig. 7 for an adaptive filter threshold-extension frequency demodulator. The sound is quasi-instantaneously coded with a parity bit for error protection.

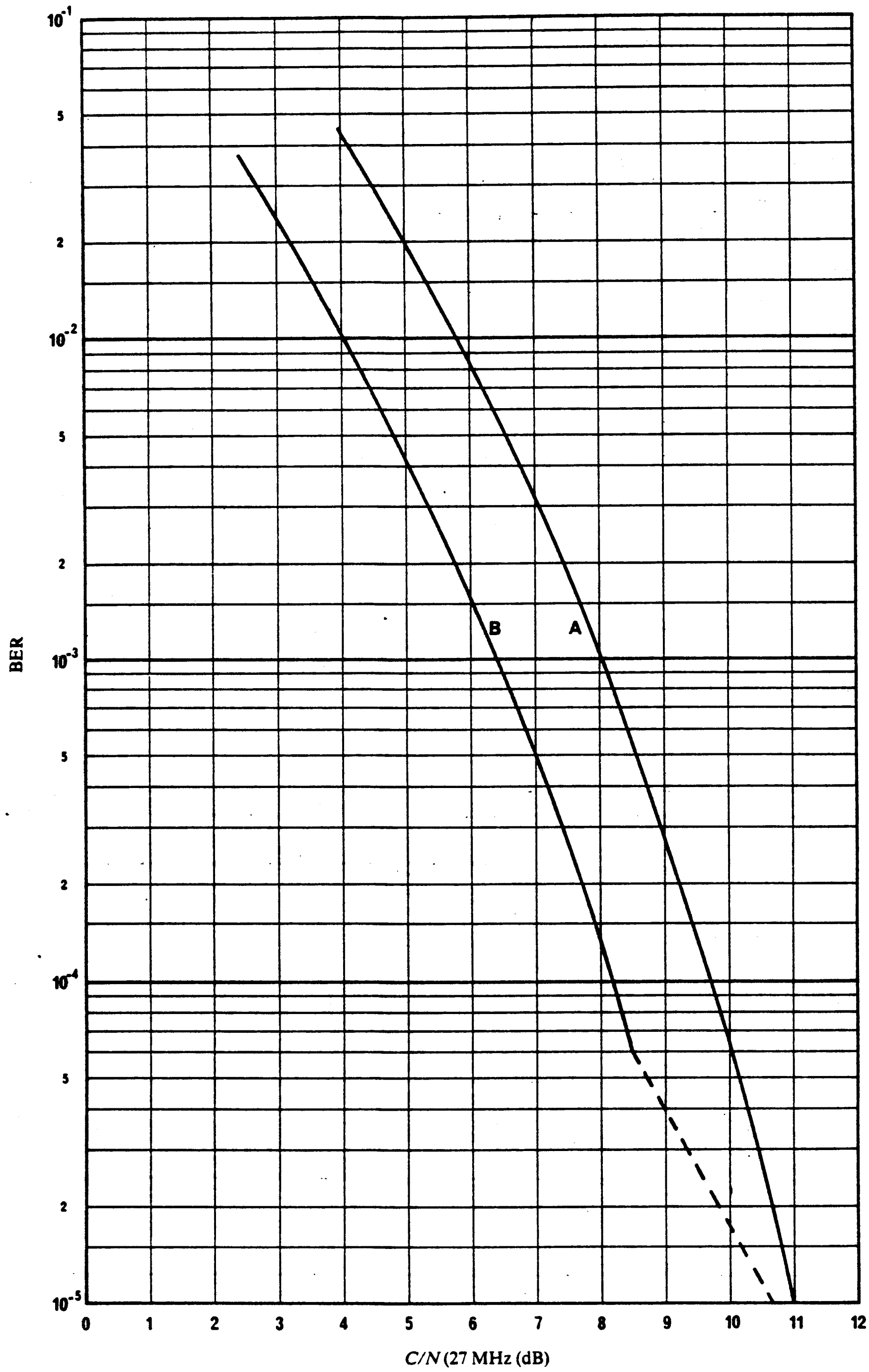


FIGURE 4 – *Bit error ratio measured using the satellite simulator*
– bit rate: 10.125 Mbit/s
– modulation: duobinary FM

Curves A: conventional demodulation
B: adaptive filter threshold-extension demodulator

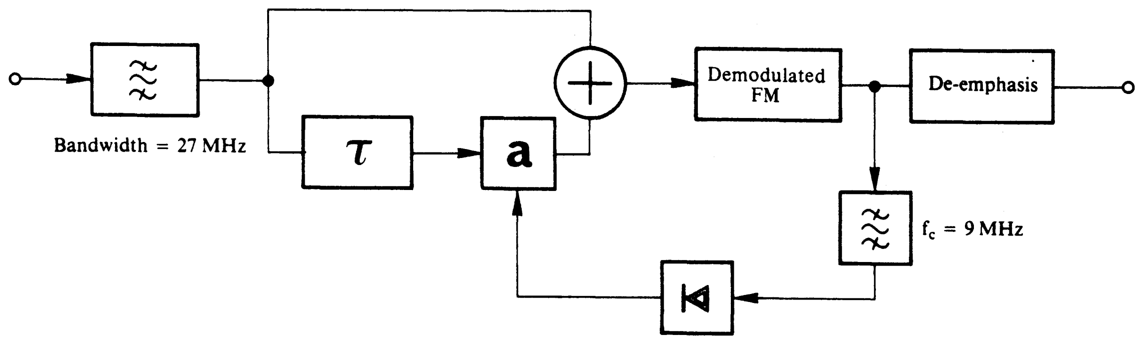


FIGURE 5 – *Principle of a threshold-extension demodulator based on progressive bandwidth reduction of the IF filter when the carrier-to-noise ratio reaches the frequency demodulation threshold (11 dB)*

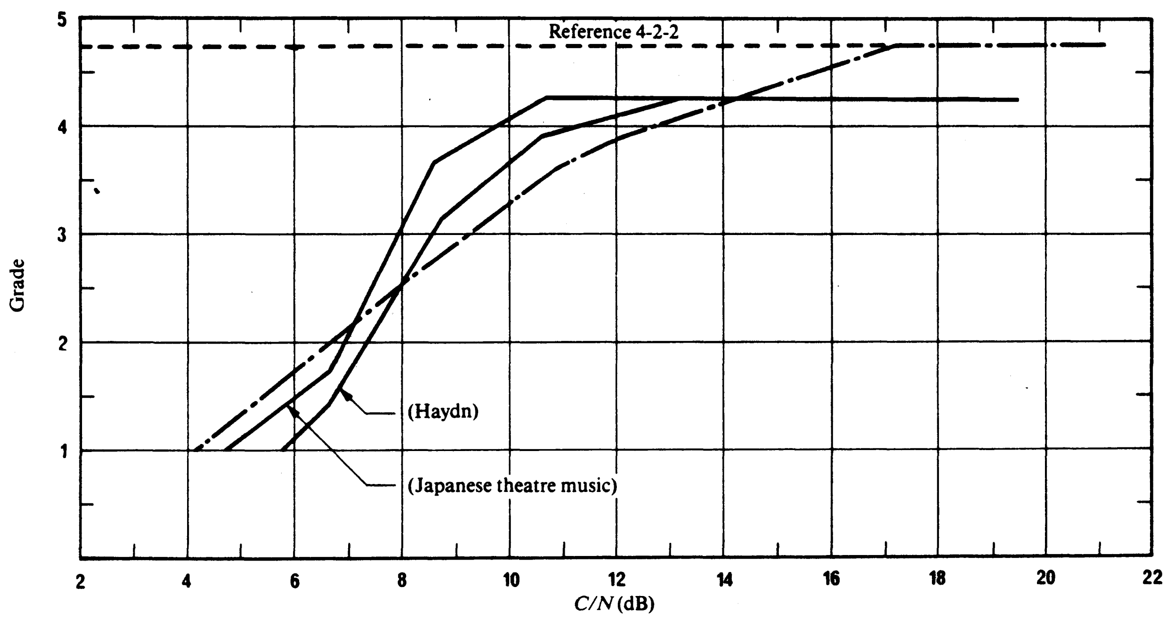


FIGURE 6 – *Picture and sound quality as a function of the carrier-to-noise ratio for a conventional demodulator*

——— sound (quasi-instantaneous coding and protection by parity bit)

—·— picture

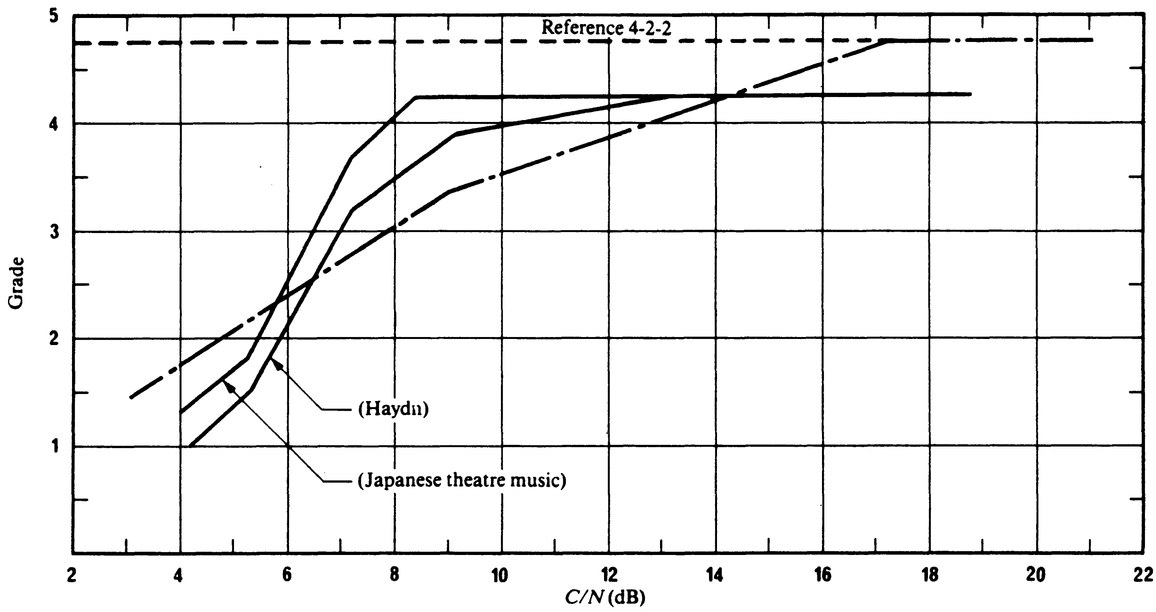


FIGURE 7 – *Picture and sound quality as a function of the carrier-to-noise ratio for an adaptive filter improved threshold demodulator (see Fig. 5)*

——— sound (quasi-instantaneous coding and protection by parity bit)

—·— picture

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ANNEX IV

EXPERIMENTAL RESULTS FOR THE TDM BASEBAND MULTIPLEXING SYSTEM

"D-MAC/packet" (type B system using duobinary modulation)

1. Introduction

This Annex summarizes the results of laboratory tests carried out in the U.K. with satellite channel simulation equipment (Priestman and O'Neill, 1987 and Beech 1987) and in a linear channel (Clark, 1987).

The characteristics of the satellite channel simulation equipment were as follows (see Stickland and Barber 1984):

Satellite Input Filter
3 dB bandwidth : 36 MHz
20 dB bandwidth : 44 MHz
Group delay : 5 ns at centre frequency ± 13.5 MHz

Satellite Output Filter
3 dB bandwidth : 38 MHz
20 dB bandwidth : 52 MHz
Group delay : 8 ns at centre frequency ± 13.5 MHz

Satellite TWT Amplifier

AM to PM factor: 5.5 degrees/dB. This is the measured figure and is typical of TWTs used in high power satellite broadcasting applications. The TWT was operated at saturation. (The specified figure is 5 degrees/dB).

The AM to PM factor is not significant for the data component of the D-MAC/packet system. The transmitted frequency modulated data component has an essentially constant envelope and there is only a very small amount of energy outside the passband of the typical direct broadcasting satellite input multiplexing filter. Hence the signal at the input of the travelling wave tube has essentially constant amplitude.

The feeder link transmitter was a nominally 1.5kW klystron operated at about 1kW output power.

No thermal noise was added on the feeder link.

2. Bit error ratio

The curves of Figure 8 show the bit error ratios obtained for various transmission conditions.

Curves C and B apply for a linear channel and with 27 MHz and 21 MHz bandwidth IF filters, respectively. It can be seen that the 21 MHz bandwidth IF filter provides about 1 dB performance improvement at 10^{-3} bit error ratio. Such a filter bandwidth is compatible with low distortion MAC vision transmission and also provides about 1 dB frequency modulation threshold extension improvement for the vision signal compared to a 27 MHz bandwidth IF filter. Curve A applies to the simulated satellite channel and a 27 MHz bandwidth IF filter.

Further improvements in performance can be obtained using maximum likelihood decoding technques (the Viterbi algorithm). Figure 8, curves D and E, relate to a receiver employing such techniques with a 21 MHz IF filter in a linear channel and via the satellite simulator respectively. The carrier to noise ratio for 10^{-3} bit error ratio is 8 dB. Therefore, using such techniques, the continuity of service criterion of 10^{-3} bit error ratio for 8 dB carrier to noise ratio is satisfied. The degradation due to the satellite channel is negligible.

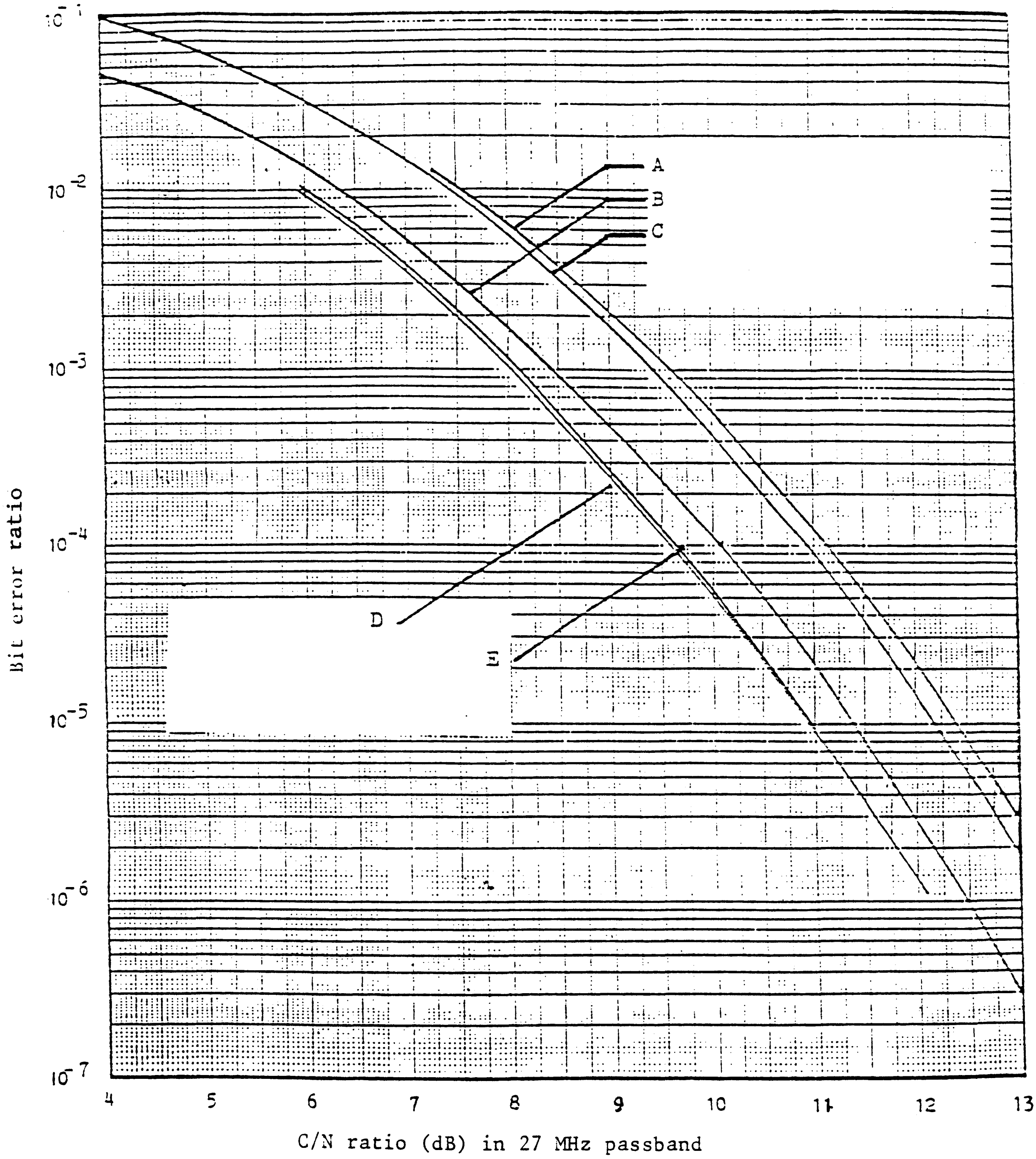

FIGURE 8

Bit error ratio performance of the D-MAC/packet system using frequency modulation

A: Simulated satellite channel 27 MHz)
B: Linear channel 21 MHz) normal decoding
C: Linear channel 27 MHz)

D: Linear channel 21 MHz) Viterbi coding
E: Simulated satellite channel 21 MHz)

Table IV indicates the results of measurements of multiple bit errors for the D-MAC/packet system using normal decoding in a linear channel (Clark, 1987). The double-, triple- and higher-order bit error ratios are small compared to the total bit error ratio. The result of de-interleaving these errors in the D-MAC/packet receiver is to distribute the individual bit errors through the data multiplex such that the output data contains predominately isolated single bit errors.

TABLE IV - Multiple bit error parameters for the D-MAC/packet system using frequency modulation

Receiver IF filter bandwidth	R(1)=BER	R(2)	R(3)	R(4)
27 MHz	3.2×10^{-3}	2.9×10^{-4}	1.2×10^{-5}	4.8×10^{-7}
20 MHz	1.3×10^{-3}	5.3×10^{-5}	1.8×10^{-6}	4.9×10^{-8}

Carrier-to-noise power ratio : 8.5 dB in 27 MHz bandwidth.
(Measurement performed in a linear channel.)

$$R(n) = \frac{\text{rate of occurrence of a group of n consecutive bit errors (events/s)}}{\text{data rate (bits/s)}}$$

3. Sound and picture quality of the D-MAC/packet system

3.1 Sound

Resulting from the de-interleaving process used in the MAC/packet systems,the sound channel performance is dependent only on the channel bit error ratio. Therefore it is possible to use the results applying to the C-MAC/packet system for the sound quality of the different MAC/packet sound coding options, scaled appropriately for the carrier-to-noise ratio differences between the C-MAC/packet system and the D-MAC/packet system.

Using the Viterbi decoding algorithm, the sound quality of the D-MAC/packet system is equal to that of the C-MAC/packet system at carrier-to-noise ratios equal to or greater than 6 dB.

Thus,receivers for the D-MAC/packet system using the Viterbi decoding algorithm can satisfy the service criterion for the MAC/packet family of systems.

3.2 Picture

The picture coding and modulation for the D-MAC/packet system are identical to those of the other members of the MAC/packet family. Hence the picture quality is identical to that of the C-MAC/packet and D2-MAC/packet systems described in the Annexes to this report. The baseband frequency response of the D-MAC/packet system is shown in

Figure 9 for the case of a receiver baseband filter of 8.4 MHz. It will be noted that, in the case of a 21 MHz filter, there is a small degradation of the high frequency response. However, this is limited to less than 0.5 dB reduction in the worst case encountered with typical vision signals corresponding to -3 dB relative to 1 V p.-p. The use of a 27 MHz filter will permit an increased system bandwidth.

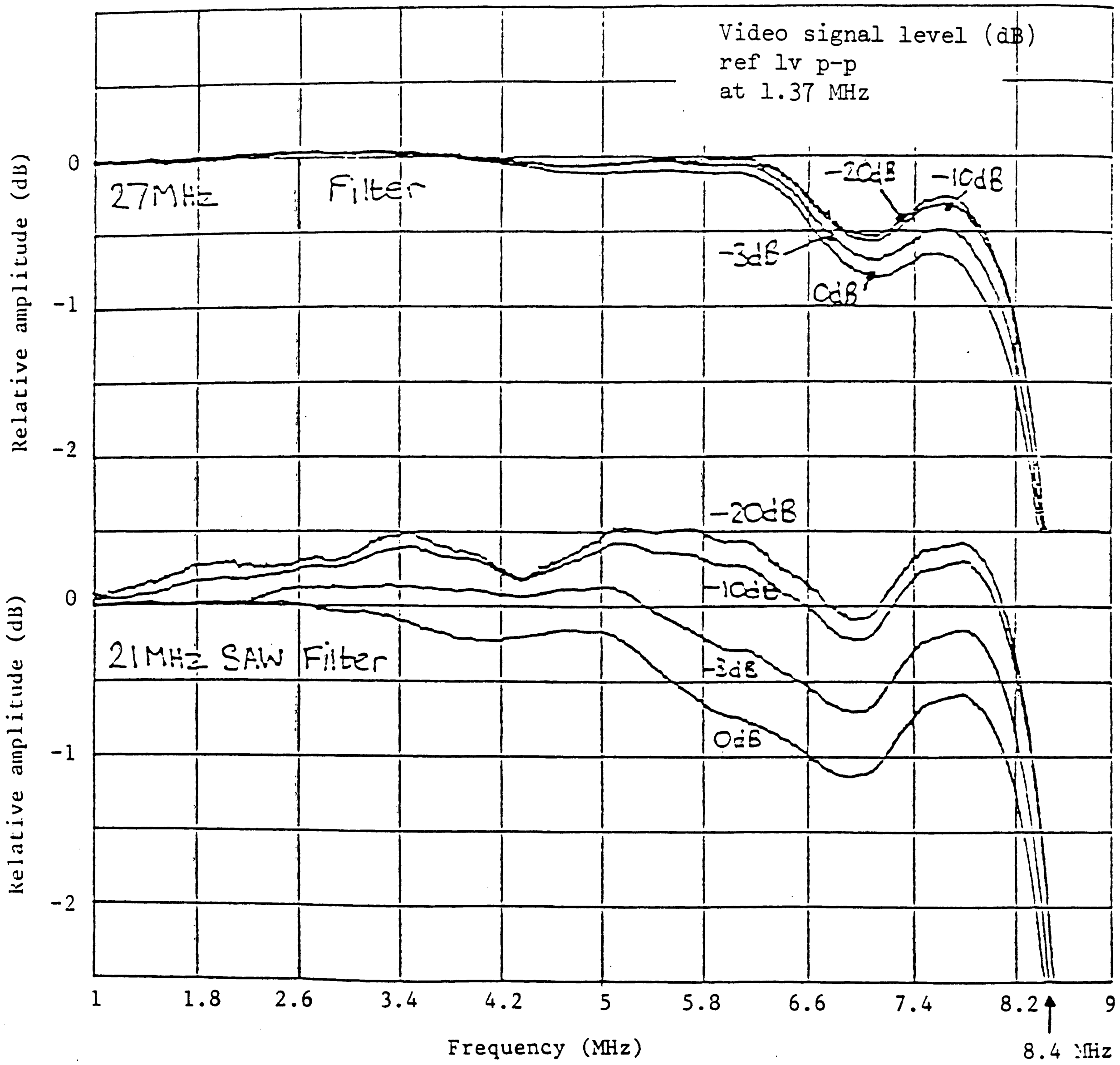

FIGURE 9

Baseband frequency response of the D-MAC/packet equipment using two different channel filters and an 8.4 MHz baseband filter in the receiver

REFERENCES

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ANNEX V

EXPERIMENTAL RESULTS CONCERNING THE RF TIME-MULTIPLEX SYSTEM (TYPE C SYSTEM)

The following summarizes the results of numerous tests conducted by the EBU, both in the laboratory and with the use of the OTS satellite, on the type C system, operated in conjunction with a MAC vision signal and in the case of either a continuous structure-map multiplex or a packet multiplex.

1. Bit error ratio

See Figs. 10 and 11 below and Table V.

TABLE V – *Typical measured BER versus C/N for different demodulation methods of 2-4-PSK*

	C/N in 27 MHz for BER of 10^{-3} (dB)
T demodulation i.e. conventional differential demodulator	7.9
T + 2*T* demodulation i.e. "Masamura" demodulator (1)	7.3
Coherent demodulation (continuous data only)	7.1

(1) The "Masamura" demodulator [Masamura *et al.*, 1979] gives a result which approaches that of a coherent demodulator and can achieve this performance in burst mode.

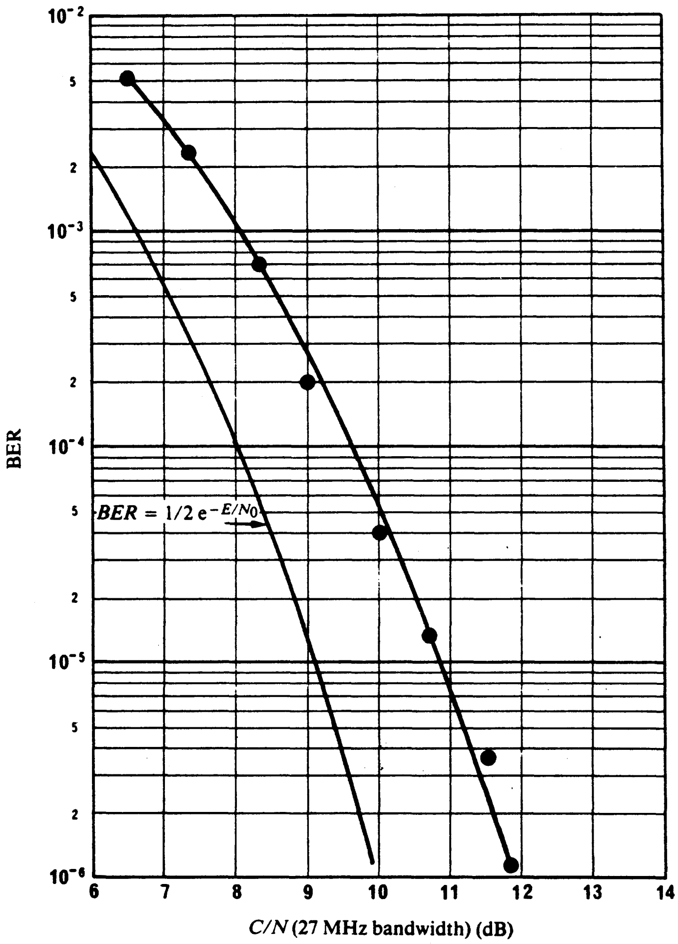

FIGURE 10 – *Typical BER as measured in the case of transmission via the OTS satellite and various satellite simulators*

Bit rate : 20.25 Mbit/s
2-4 PSK differential demodulation

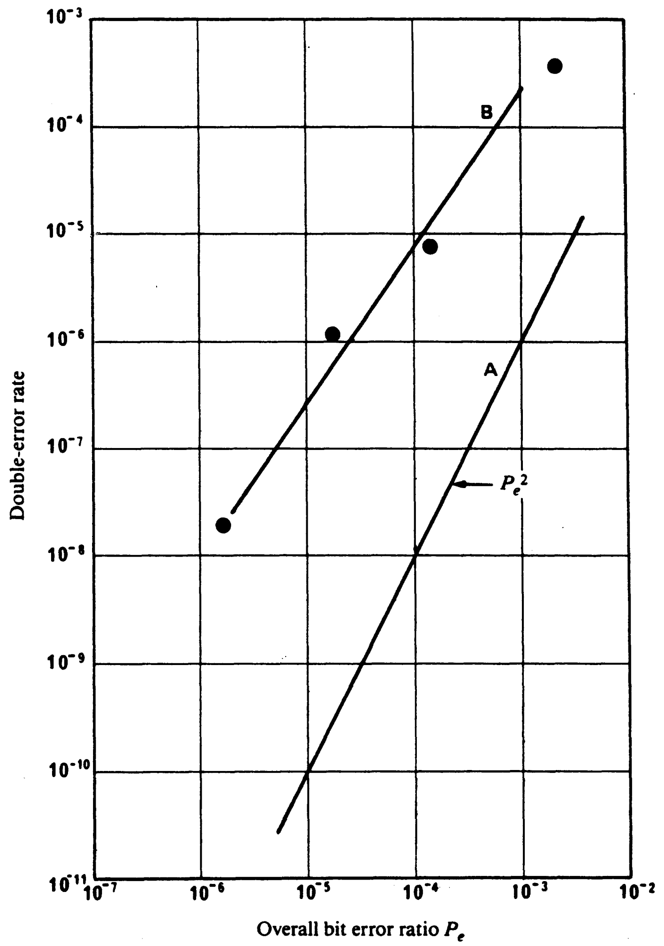

FIGURE 11 – *Double-error rate versus overall bit error ratio*

A: independent errors (theoretical: P_e^2)
B: measured for 2-4-PSK with differential demodulation using analogue delay line and multiplier

2. Vision and sound quality for the C-MAC system

Subjective tests have been conducted with a group of observers (see § 2.3.2 of Annex I).

The sound tests have been carried out with conventional differential demodulation of the 2-4-PSK signal, but with different coding laws, multiplexing systems and protection levels. Two combinations of these different aspects have been the subject of subjective assessments in accordance with Recommendation 562. These results were obtained on different occasions, 37 observers being present for combination 1 and 15 observers for combination 2. To aid comparison between Fig. 12 and Figs. 13a and 13b below, the same bit error ratio was used.

Combination 1: Structure map multiplex used in conjunction with a near-instantaneous companding law, Hamming code protection and transmission of the scale factor within the parity bits. Because of the error protection technique used, the capacity of this combination was reduced from eight to six 15 kHz sound channels.

Combination 2: Packet multiplex as defined in Report 1073 using near-instantaneous companding law with parity protection (first level) and with Hamming code protection (second level). Linear coding using parity protection (first level) and with Hamming code protection (second level). The capacity for 15 kHz sound channels ranges from eight channels in the case of near-instantaneous companding law first level protection to four channels in the case of linear coding with second level protection. In the case of near-instantaneous companded second level protection and in the case of linear coding first level protection, the capacity is six channels.

The corresponding results expressed using the 5 grade quality scale are shown in Fig. 12 for combination 1 and in Figs. 13a and 13b for combination 2.

It can be seen from Fig. 12 that the best vision quality is such that the C-MAC system may be considered as "transparent" to the 4:2:2 standard of Recommendation 601. From Figs. 13a and 13b it can be seen that all four high quality sound coding options of the C-MAC/packet system approach the reference grade (>4.8) above 9.5 dB *C/N*.

3. Vision and sound failure points

For combination 1 and for the near-instantaneous companding law of combination 2 described in § 2, subjective tests in which the vision and the sound elements of the programme were presented simultaneously have been conducted in order to define the relative failure points of sound and vision.

The programme consisted of a transmission of a musical composition played by a guitar.

The following values for the *C/N* ratio at the failure points (quality grade 1.5) were obtained:

Combination 1: *Vision*

- Normal demodulator: 4 dB
- Threshold-extension demodulator: 4 dB

Sound: 4 dB

Combination 2: *Vision*

- Normal demodulator: 3 dB
- Threshold-extension demodulator: 2 dB

Sound: 4.5 dB

The differences from the results presented in § 2 (Figs. 12 and 13) are due to the differences in the assessment methods and evaluation criteria used.

4. Limit of speech intelligibility

For combination 2 the limit of intelligibility of speech was assessed with simultaneous presentation of vision and sound. A figure of 3 dB was obtained for that limit.

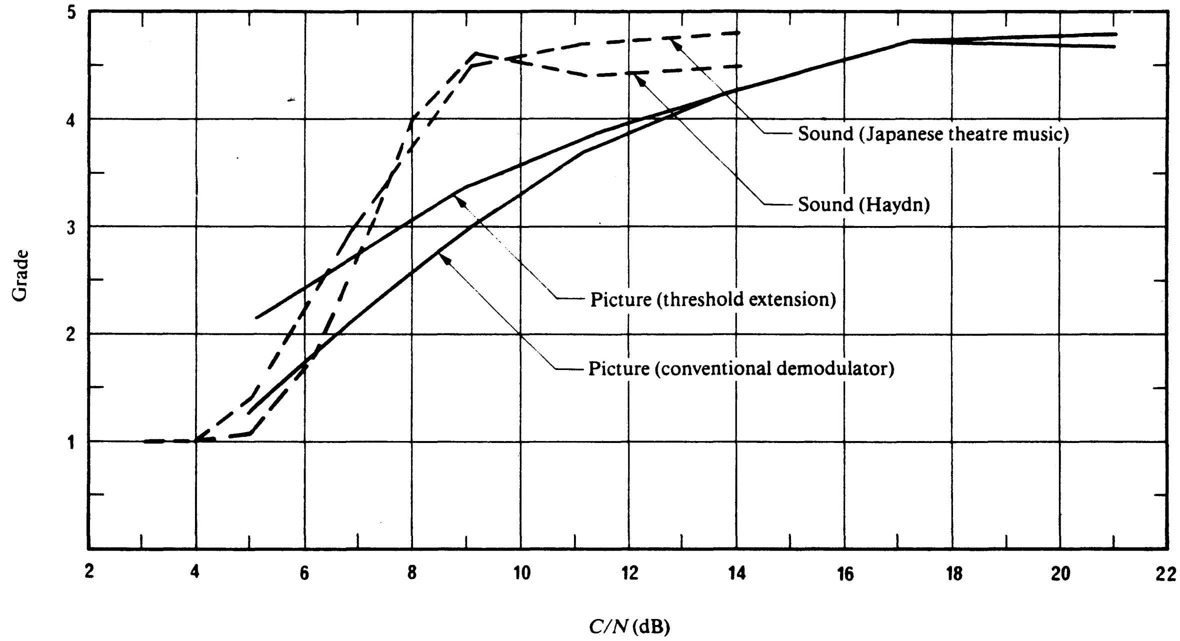

FIGURE 12 – *Quality as a function of the C/N ratio for combination 1 (37 observers)*

REFERENCE

MASAMURA, T., SAMEJIMA, S., MORIHIRO, Y. and FUKETA, H. [June, 1979] Differential detection of MSK with nonredundant error correction. *IEEE Trans. Comm.*, Vol. COM-27, **6**, 912-918.

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CLARK, D. S. [1985] Demodulation techniques for 20 Mbit/s 2-4 PSK signals. IBA E & D Report No. 133/85.

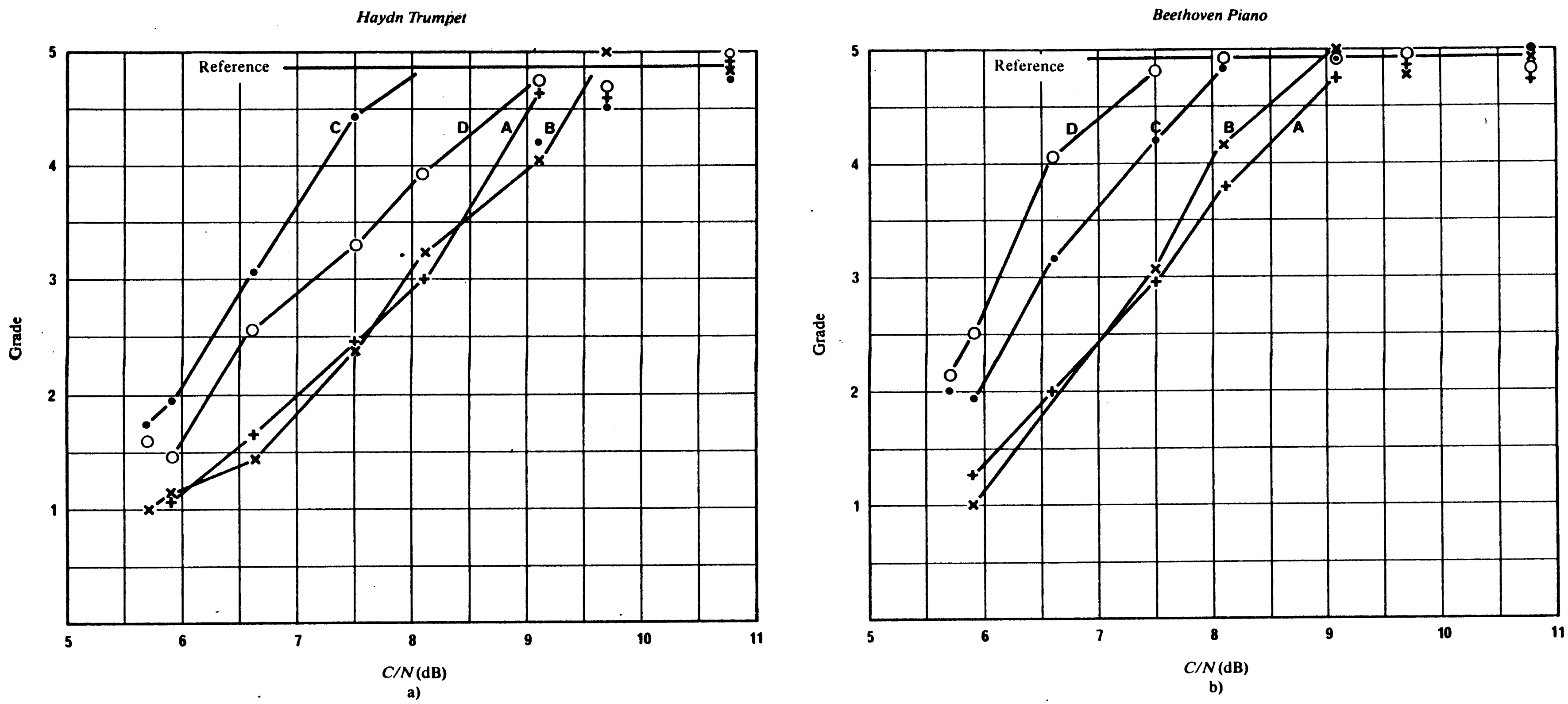

FIGURE 13 – *Subjective quality of C-MAC/packet sound coding options**

Key – A companded first level – protected
B linear first level – protected
C companded second level – protected
D linear second level – protected

* For the same bit error ratio, these results could also be valid for the D2-MAC/packet system which uses the same coding law and the same error protection as the C-MAC/packet system. The relation between bit error ratio and *C/N* is shown in Fig. 8 of Annex IV for the C-MAC/packet system and in Fig. 4 of Annex III for the D2-MAC/packet system.