REPORT 458-5

CHARACTERISTICS OF SYSTEMS IN LF, MF AND HF BROADCASTING

(Question 44/10)

(1970-1974-1978-1982-1986-1990)

1. Introduction

Question 44/10 relates to the possibility of standardizing one or more sound broadcasting systems on a world-wide basis. It is clear that the study of this complex question is not sufficiently advanced to achieve this aim. The present Report, therefore, is only a summary of the information available, intended to encourage administrations, broadcasting organizations and industry, to take an interest in these questions and to undertake the studies necessary to solve them.

At present, broadcasts in bands 5, 6 and 7, unlike sound and television broadcasting in band 8 (VHF), are operated throughout the world with an almost complete absence of internationally standardized transmission characteristics; with the exception of channel spacings and carrier frequencies for bands 5 and 6, but even these differ from Region to Region. The other transmission characteristics vary from country to country and in many cases even from transmitter to transmitter [EBU, 1971].

2. Systems available for standardization

The following list of possible systems cannot, at the present time, be considered complete. Several of these systems are compared in [Haviland, 1969]. This study also shows that interference between transmitters must be considered when defining a system and the importance of well defined channel spacings becomes apparent.

Each of the above systems may incorporate modulation processing devices (see Note 1). If such a device is necessary in the receiver to obtain full advantage of a similar device in the transmitter, the code should be completed by a suitable abbreviation. Thus, an amplitude-modulation double-sideband (DSB) system with envelope demodulation comprising a compressor in the transmitter and an expander in the receiver, is coded AM-DSB-ENV-COMPANDOR. An example is given in Annex I.

Descriptions of amplitude-modulation double-sideband systems with synchronous detection appear in [Netzband, 1969] and [CCIR, 1966-69] and it should be noted that only these systems can serve as transition systems from amplitude-modulation double-sideband systems with envelope detection to single-sideband (SSB) systems with synchronous detection. Receivers based on synchronous detection would produce undistorted audio-frequency signals for both the above systems ("receiver compatibility").

The transmission of compatible single-sideband signals (see Note 2) in sound broadcasting should be considered to be inappropriate for regular programme services, and consequently be restricted to exceptional cases for the following reasons:

- it is difficult to reduce out-of-band emissions in the suppressed sideband of a compatible SSB system;
- strong out-of-band emissions will result from maladjustments of compatible SSB transmitters;
- imperfections of conventional DSB receivers which have no bearing on the reproduction quality of DSB transmissions may give rise to dynamic distortion, when the automatic gain control follows a portion of the spectrum with high energy density rather than the carrier, or to non-linear distortion due to the amplitude and phase response within the radio-frequency and intermediate frequency stages;

– additional non-linear distortions, similar to those which affect double-sideband signals, which are to be expected when receiving sky-wave signals, cannot be reduced by using a different demodulation technique than envelope detection.

Note 1. – By "modulation processing" is understood any process consisting of altering certain characteristics of the modulation, such as the dynamic range, audio-frequency bandwidth, etc.

Note 2. – A transmission is said to be "compatible" if it can be received on an existing conventional receiver, without any modification whatsoever to the receiver, and then gives a quality of reception at least as satisfactory as that obtained at present in a double-sideband system.

TABLE I

Modulation	Detection	Code
Amplitude-modulation, double sideband	Envelope detection	AM-DSB-ENV
Amplitude-modulation, double sideband	Synchronous detection	AM-DSB-SYNC
Single-sideband, amplitude-modulation	Synchronous detection	SSB-SYNC
Frequency modulation (narrow band)		FM

3. Characteristics to be specified

For the systems mentioned in § 2, the characteristics whose standardization will be required are given below. This list is not necessarily complete.

3.1 *All systems*

- channel spacing,
- carrier frequencies,
- intermediate frequency or frequencies,
- receiver oscillator frequency stability.

Note 1. – The relationship between these characteristics is shown in Annex II.

- audio-frequency bandwidth of the programme,
- necessary bandwidth of emission,
- overall bandwidth of the receiver.

Note 2. – For the relationship between these characteristics and the channel spacing, see Annex I to Recommendation 639. Deviations from standardized values may be tolerated, if they do not result in unacceptable interference.

- characteristics of modulation processing devices.

In addition, the following characteristics should be standarized:

3.2 *Amplitude-modulation double-sideband systems*

- maximum depth of modulation.

When laying down the characteristics of the transmission system and the reference receiver, in order to obtain the best adaptation of a frequency-assignment plan to receivers of reasonable performance, the following rules should be taken into account:

3.2.1 The transmission system and receiver characteristic should be suitably related, particularly in the amplitude/frequency response. Harmonic distortion should be reduced to acceptable values. In this respect, criteria for the determination of tolerances for the overall amplitude and phase characteristics and ways for the assessment of practical values are set out and discussed in [Makiedonski, 1974].

3.2.2 The audio-frequency bandwidth transmitted should be related to the carrier spacing (see Annex I to Recommendation 639). (The precise bandwidth cannot be given; some administrations are of the opinion that it should be one half of the spacing between carriers, and some of them believe that this value may even achieve the value of unity; but this depends on the absolute channel spacing.)

3.2.3 Uniform carrier spacings, with nominal carrier frequencies being an integral multiple of the carrier spacing, should be adopted, at least within the broadcasting bands 5 and 6. (There are technical advantages also in the adoption of uniform carrier spacing on a world-wide basis, including band 7.)

3.2.4 The intermediate frequency (or frequencies) of the receiver should be chosen so that interference created internally within the receiver is at a minimum. With stable local oscillators (see Annex II) this condition is best met when the intermediate frequency (or frequencies) of the receiver is an integral multiple of the carrier spacing.

3.3 *Single-sideband systems with synchronous detection*

- degree of reduction of the carrier wave (see Recommendation 326, § 1.5),
- degree of suppression of the unwanted sideband,
- maximum permissible values of intermodulation products,
- auxiliary signals to obtain receiver synchronization.

The considerations under § 3.2 would apply, and where possible the same characteristics should be adopted.

Note 3. – More information about SSB systems is contained in Report 1059.

3.4 *Frequency modulation*

- modulation index,
- maximum modulation frequency.

4. Stereophonic systems in AM broadcasting

[CCIR, 1974-78] describes four methods of stereophonic broadcasting for use in bands 5 and 6. All systems described in this Document use conventional amplitude modulation for the stereophonic sum signal and angle modulation of the same carrier frequency for the stereophonic difference signal. They differ in the way the modulations are generated and in the resulting distribution of the audio information in the various radio frequency sidebands.

4.1 *Amplitude modulated stereo system used in Australia and Brazil*

The Australian Administration has, from February 1985, allowed its MF stations to broadcast in stereophony. Compatible quadrature amplitude modulation (CQUAM) has been adopted as the stereophonic modulation standard. Field tests [CCIR, 1982-86] confirmed that the occupied bandwidth of stereophonic transmission is not significantly greater than monophonic transmission. The Brazilian Administration adopted the same system on an exclusive basis since January 1986.

5. Preemphasis, demphasis and 10 kHz audio bandwidth limitation for administrations employing 10 kHz channel spacing in band 6 (MF)

With 10 kHz channel spacing broadcasting in band 6 (MF), a standard preemphasis, deemphasis and 10 kHz audio bandwidth limitation has been implemented [CCIR, 1986-90]. This produces a transmission/reception system with an overall audio frequency response that is essentially flat from 50 Hz to nearly 10 kHz (limited only by the receiver's choice of bandwidth), and reduces interference caused to stations operating +/- 20 kHz removed in frequency. Further, this entirely eliminates undesired high-order dynamic intermodulation products that contribute noise and interference in the MF band. This system is described in detail in Annex I.

REFERENCES

EBU [August, 1971] Modulation processing techniques in AM sound broadcasting. Doc. Tech. No. 3088.

HAVILAND, R. P. [June, 1969] A comparative study of communications systems using different modulation-demodulation techniques. *EBU Rev Tech.*, **115-A**, 102-108.

MAKIEDONSKI, A. [1974] Analiza znieksztalcen obwiedni sygnalu zmodulowanego przy jednoczesnym występowaniu znieksztalcen tlumieniowych i fazowych jego grupy falowej (Analysis of the envelope distortions of a signal modulated in the simultaneous presence of attenuation and phase distortion of its spectrum). *Prace Instytutu Łączności*, Vol. XXI, **375**.

NETZBAND, R. [April, 1969] A novel arrangement for the reception of single-sideband transmissions as a result of frequency error and harmonic distortion. *EBU Rev. Tech.*, **114-A**, 60-63.

CCIR Documents

[1966-69]: X/135 (USA).

[1974-78]: 10/279 (USA).

[1982-86]: 10/246 (Australia).

[1986-90]: 10/103 (United States of America).

BIBLIOGRAPHY

DOC [July, 1984] Report on AM stereo system tests. Lab. Rep. 110. Dept. of Communications (DOC), Belconnen ACT 2616, Australia.

LACHARNAY, S., FONTEYNE, J. and MASSUCCI, M. [1978] Radiodiffusion en ondes kilométriques et hectométriques (LF and MF broadcasting). *Radiodif.-Télév.*, **1**.

CCIR Documents

[1978-82]: 10/15 (EBU).

[1986-90]: 10/81 (Brazil).

ANNEX I

THE USE OF VARIOUS MODULATION PROCESSING DEVICES

In the OIRT, studies have been made of the possibilities of increasing the modulation factor, i.e., the sideband power, by either trapezoidal modulation or dynamic-compression operation, with transmitters operating in bands 6 (MF) and 7 (HF) [CCIR, 1970-74a]. The results were that dynamic compression without clipping should be preferred, if the objective is to increase the sideband power with a minimum loss of quality (e.g. for music programmes). On the other hand, if a loss of quality is deemed unimportant (e.g. for speech programmes), trapezoidal modulation leads to a higher degree of sideband power.

Experiments carried out in the RFZ, Berlin, have confirmed the assumed increase of the sideband power as follows:

– *Compression of the dynamic range by 12 dB:*

  average gain with a rise-time of 0.5 ms and a decay-time of 35 ms: $\approx$ 6 dB.

– *Compression of the dynamic range by 6 dB:*

  average gain with a long decline period (1.5 s) for programmes with a wide dynamic range: $\approx$ 3 dB.

– *Trapezoidal modulation*

  average gain with 5 dB increase in the level of the audio-frequency signal and clipping: $\approx$ 3 dB.

Studies have been carried out in Sweden concerning the improvement in radio-frequency wanted-to-interfering signal ratio obtained by using audio-frequency compression and expansion in connection with a double-sideband amplitude-modulation system and a frequency-modulation system with a maximum deviation of $\pm$ 5 kHz [CCIR, 1966-69].

The audio-frequency range was from 40 to 5000 Hz.

The compressor reduced the dynamic range of the audio-frequency signal, expressed in decibels, to half its value, the time-constants were 2 ms for the rise-time and 20 ms for the decay-time. The expander had characteristics reciprocal to those of the compressor.

The test results can be summarized as follows:

In the absence of interference, no change in quality was observed when using both compressor and expander in the system. The quality was also judged to be satisfactory by listening when only the compressor was used.

In the presence of co-channel interference, the radio-frequency protection ratios (dB) were found to be as follows:

	Type of modulation	
	Amplitude	Frequency
– without compressor and expander	40 to 50	40 to 45
– with compressor only	30 to 40	30 to 40
– with compressor and expander	20 to 25	25 to 30

It should be noted that these values were obtained when the unwanted transmitter was not equipped with a compressor.

In a more extensive study covering double-sideband amplitude-modulation only, the effect of compression and expansion on the radio-frequency protection ratio was investigated when the unwanted transmitter was also equipped with a compressor [CCIR, 1970-74b]. The study was made with different types of programmes transmitted by both the wanted and the unwanted transmitters.

When a speech programme was interfered with by another speech programme, the reduction of the radio-frequency protection ratio was about 15 dB when using compression and expansion. No deterioration in reproduction quality has been reported. When only compression was applied there was a smaller reduction in protection ratio of about 10 dB; in this case, the quality of the reproduced sound was considered lower than acceptable and significantly worse than when no compression was applied.

When a music programme was disturbed by either music or speech, the result depended to a great extent on the character of the wanted programme. The advantage obtained with compression plus expansion or with compression only, was always smaller than when the wanted programme was speech, and sometimes even negligible.

A new compression method has been developed in Japan based on digital techniques [CCIR, 1982-86a]. It is applicable to a wide variety of uses from high compression transmissions to high quality sound, with less degradation of signal quality.

The principle of operation is as follows:

- the signals are divided into segments of predetermined length that includes a certain number of zero crossing points;
- control is imposed individually on each segment;
- information for control of a segment is obtained from the segment itself;
- control is fixed within the same segment and changed only at the contours of segments.

For obtaining higher compression, this digital level compression method is able to solve most of the disadvantages inherent in conventional analogue methods. For various applications, the compression characteristics are easily selectable by reading the correct ROM addresses and the dynamic range of the original signal can be retrieved at the receiving end by the received control information.

With these features this compression method has a wide variety of applications.

Studies in the United States have resulted in the adoption of a standard preemphasis, deemphasis and 10 kHz bandwidth limitation that improves the overall audio frequency response and reduces interference [CCIR, 1986-90a]. Its characteristics are described below as they apply to U.S. broadcasting in the MF band.

MF Transmission Preemphasis

Preemphasis is the boosting of high audio frequencies prior to modulation and transmission. Today, most MF stations use preemphasis to varying extents. This preemphasis is employed in an attempt to compensate for the "narrow" response of most receivers. If preemphasis is not controlled, one station may interfere with MF receivers tuned to neighboring stations located on adjacent channels. Whether such interference is objectionable will depend on (1) the response characteristics of the receiver, (2) the amount and nature of transmission preemphasis, (3) the extent to which the station is employing compression/limiting techniques, and (4) whether the AM transmission system is bandlimited in the audio processor, transmitter or antenna.

Preemphasis is useful for improvement of the MF transmission-réception system audio response only to a limited extent for receivers using IF transformers. Many receivers using ceramic filters with narrow response characteristics cannot be improved by use of excessive preemphasis. These receivers do not recover the transmission of preemphasized high audio frequencies.

Each MF broadcast station should broadcast with audio preemphasis as close as possible (within the capabilities of the station's transmission system) to the recommended standard, without exceeding it. The curve applies for audio frequencies up to 10 kHz. See Figure 1. The curve describes the recommended transmission system static audio response of an AM station.

The recommended preemphasis curve is a single zero curve with a break frequency at 2122 Hz. It is similar to the 75 microsecond curve used for Region 2 FM broadcasting. To reduce the peak boost at high frequencies, a single pole with a break frequency of 8700 Hz is employed. Analysis has shown that the proposed curve is compatible with most existing AM receivers.

Measuring Performance.

The preemphasis curve for MF use is a static curve, and cannot be measured dynamically. Studies have shown that the dynamic and non-linear functions performed by most AM station audio processors will modify any given preemphasis curve. In addition, it is the audio response of the entire AM transmission system that indicates performance in accordance with the standard. For these reasons, measuring a station's preemphasis curve for the purpose of determining compliance with this standard should be performed in accord with the following specifications:

1. Compliance with the curve should be measured by sweeping the station's transmission system with audio tones. The dynamic functions of the MF station's processor, but not the frequency shaping circuits, must be disabled (i.e., in "proof" mode).

2. The net transmission system audio response is best measured by detecting the over-the-air signal. This will ensure that the AM transmitter and antenna combination are faithfully reproducing the preemphasized audio. Alternatively, if the transmitter and antenna combination is reasonably broadband, performance can be determined by static measurement of the audio signal prior to modulation.

MF Receiver Deemphasis

Receiver deemphasis results from the selectivity characteristics of a receiver's RF and IF stages and the response characteristics of the receiver AF section. A standard deemphasis curve permits MF broadcast stations to know, with certainty, the likely overall response characteristics of AM receivers.

AM receivers should complement the recommended transmission preemphasis characteristic by incorporating a net receiver system audio response described below. (The net system audio response of an AM receiver is the combined RF, IF, and AF audio response.) The deemphasis curve is characterized by a single pole at 2122 Hz and a single zero at 8700 Hz. It is the precise complement of the preemphasis standard described above. The preemphasis/deemphasis standards apply only for audio frequencies below 10 kHz; the implementation of preemphasis/deemphasis standards produces a transmission/reception system that is essentially flat to nearly 10 kHz and limited only by the AM receiver's choice of bandwidth.

Measuring Performance.

The deemphasis characteristic should be determined by measuring the overall frequency response in accordance with International Electrotechnical Commission ("IEC") Publication 315-3, Clause 11.2.

Although an optional enhancement to an AM receiver, using notch filters is recommended. If used, the notch filter should (1) have as high a "Q" as is practical, (2) adequately suppress the interfering carriers, and (3) not unduly degrade the desired bandwidth performance of the AM receiver.

10 kHz Bandwidth for MF Transmission.

Each MF broadcast station should modulate its transmitter with an audio bandwidth described by the specification in Figure 2. Appropriate and carefully designed audio low-pass filters as the final filtering prior to modulation can be used to implement this specification. The purpose of the bandwidth specification is to remove interference by controlling the occupied RF bandwidth of AM stations.

The audio envelope input spectrum to the MF transmitter should be -15 dB at 10 kHz, smoothly decreasing to -30 dB at 10.5 kHz, then remaining at -30 dB from 10.5 kHz until 11.0 kHz. At 11.0 kHz, the audio bandwidth should be -40 dB, smoothly decreasing to -50 dB at 15 kHz. Above 15 kHz, the audio bandwidth should remain at least -50 dB. The reference level is 1 dB above a 200 Hz sine wave at 90% negative modulation.

Measuring Performance.

An MF station is determined to be in compliance with this bandwidth characteristic by measurement of the station's audio bandwidth in accordance with the following parameters:

1. Audio bandwidth measurements should be obtained at the audio input terminals to the AM transmitter. For AM Stereo stations, audio bandwidth should be measured at the L+R audio input terminals to the RF modulator. Note that the bandwidth standard characterizes an audio bandwidth that represents station program material that has been modified by possibly non-linear circuits in the station's audio processor. For this reason, it is recommended that a test signal be used that adequately characterizes typical audio program material, rather than relying on static audio test tones. However, it may still be useful to measure bandwidth statically at the time that AM preemphasis is measured.

2. Audio bandwidth should be measured using a test signal consisting of USASI (United States of America Standards Institute) noise (see Figure 3) that is pulsed by a frequency of 2.5 Hz at a duty cycle of 12.5%. USASI noise is intended to simulate the long-term average spectra of typical audio program material. Pulsing of the noise is intended to simulate audio transients found in audio program material. USASI noise is a white noise source (i.e. noise with equal energy at all frequencies) that is filtered by (1) a 100 Hz, 6 dB per octave high-pass network and (2) a 320 Hz, 6 dB per octave low-pass network. See Figures 3 and 4. A pulsed USASI noise generator is shown in Figure 4. Using the attenuator pad, the ratio of peak-to-average amplitude should be 20 dB at the audio output of the pulser. The station's audio processor must be in normal operating mode.

3. A suitable swept-frequency or FFT (Fast Fourier Transform) spectrum analyzer should be used to measure compliance with the bandwidth specification. When a swept-frequency audio spectrum analyzer is used to measure compliance with the bandwidth specification, the analyzer's setup should consist of: (a) a 300 Hz resolution bandwidth, (b) 2 kHz/horizontal division, (c) 10 dB/vertical division, (d) a reference of 1 dB above 200 Hz (sine wave) 90% negative modulation, and (e) a display set to maximum peak hold (or equivalent function). The analyzer's operating span and sensitivity are adjusted as necessary to determine compliance. When a FFT analyzer is used to measure compliance with the bandwidth specification, the analyzer's setup should consist of (a) a reference set to 1 dB above 200 Hz (sine wave) 90% negative modulation, (b) Hanning window, (c) a horizontal span of 20 kHz, (d) a dynamic range of 80 dB (or available range), (e) a display set to maximum peak hold (or equivalent function).

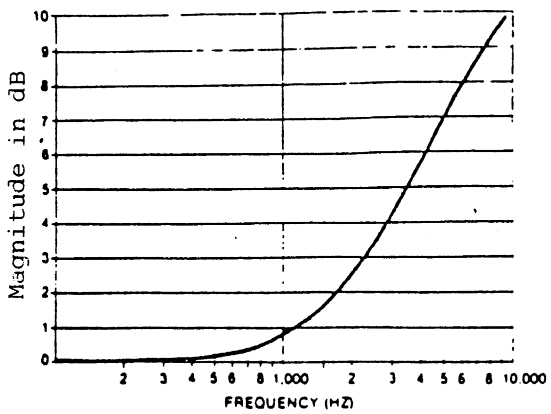

FIGURE 1 - Modified 75 µs AM Standard Preemphasis Curve

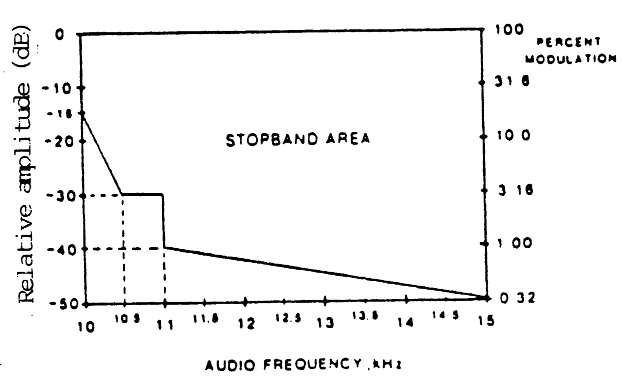

FIGURE 2 - NRSC Stopband Specification
(Audio Envelope Input Spectrum to AM Transmitter)

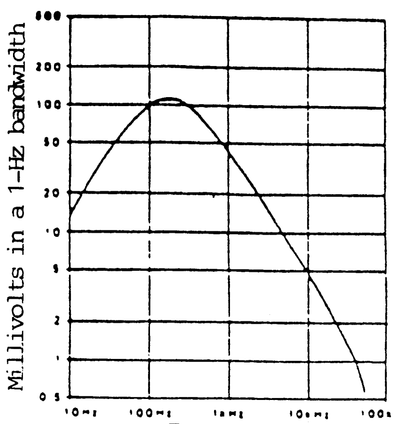

FIGURE 3 - Spectra of USASI Noise

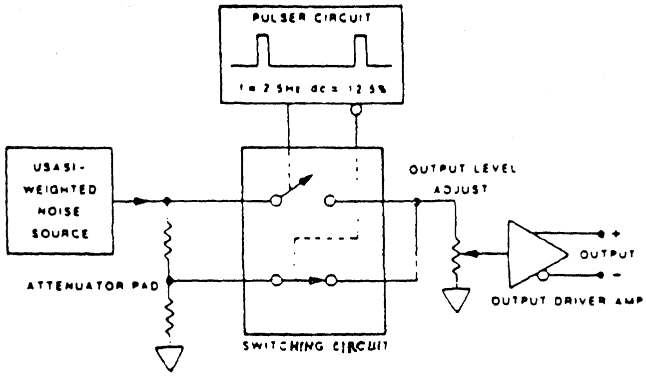

FIGURE 4 - Pulsed USASI Noise Generator

4. In the USSR [CCIR, 1986-90b], a method of measuring the mean modulation depth of double-sideband AM broadcasting signals has been developed.

The mean modulation depth of the AM signal is determined by the formula:

$$\bar{m} = \frac{\pi}{2} \cdot \frac{\int_0^T |U_A(t) - U_0| \, dt}{U_0 T}$$

where:

U_0 : signal carrier voltage amplitude,

$U_A(t)$: istantaneous value of AM signal amplitude,

T : averageing time.

The mean modulation depth is equal to the product of the maximum value of the modulation depth and the parameter of the AF modulating signal, designated the "relative mean voltage".

$$U_{rmv} = \frac{\pi}{2\sqrt{2}} \cdot \frac{\int_0^T | U(t) \, dt}{U_H T}$$

where:

U_{rmv} : relative mean voltage

U(t) - instantaneous value of the modulating signal,

U_H - voltage corresponding to the nominal level at the point of measurement.

The parameter U_{rmv} may be used for the numerical calculation of the variation in modulation depth obtained as a result of the dynamic compression of the modulation signal.

A digital broadcasting signal parameter meter has been developed which uses an algorithm to calculate the mean modulation depth and the RMV in accordance with the formulae given above. The instrument has two measuring channels, which enables simultaneous measurements to be made at the modulator input and the transmitter output, or at the input and output of the processor. It is used to monitor double-sideband AM transmitters and broadcasting signal preliminary processing devices.

REFERENCES

CCIR Documents

[1966-69]: X/153 (Sweden).

[1970-74]: **a.** 10/66 (OIRT); **b.** 10/43 (Sweden).

[1982-86]: **a.** 10/211 (Japan).

[1986-90]: a. 10/103 (United States of America); b. 10/256 (USSR).

BIBLIOGRAPHY

WAKURI, T. [November, 1984] An approach to digital level compression using the zero crossing points of programme signals. NHK Lab. Note No. 308.

ANNEX II

CHANNEL SPACING, PROTECTION RATIO AND INTERMEDIATE FREQUENCY

When choosing the carrier frequencies, channel spacing and also the intermediate frequencies to be used in receivers, it is important that they should be chosen so as to minimize interference from:

- the local oscillators of the receivers in use, or of nearby receivers, either by a fundamental or a harmonic frequency;
- harmonics of a transmitted frequency, or other possible intermodulation products [CCIR, 1963-66; SCART, 1966; CCIR, 1970-74].

If both the carrier frequencies and the intermediate frequency are an integral multiple of the carrier spacing, then all interfering products will also be integral multiples of the carrier spacing. Theoretically, therefore, maximum protection could then be obtained because the frequency difference between any interfering signal of this kind and the wanted carrier frequency, would be zero or a multiple of the channel spacing.

If these requirements are to be met with in a particular broadcasting band, it would be essential for the channel spacing to be uniform throughout the band. It would be more advantageous, moreover, if this condition could be met with in both bands 5 and 6, or better still, throughout bands 5, 6 and 7. On the other hand, this condition should be satisfied on a world-wide scale or at least in those areas, where a single frequency assignment plan exists or will be established [Eden, 1967].

However, it must be noted that the disturbance caused by an interfering signal increases rapidly as its frequency difference from the wanted signal increases from zero.

Under present-day conditions in band 7 the frequency differences might have any possible value and this may require an additional protection ratio of up to 17.5 dB. With the adoption of the proposed arrangement, the maximum frequency difference would depend on the accuracy with which the local oscillator frequency and the centre frequency of the intermediate-frequency passband can be controlled. To achieve an improvement close to the maximum possible it would be necessary to achieve stabilities of the order of 100 Hz. As far as the intermediate-frequency stability is concerned this could be achieved by using ceramic or mechanical filters rather than using conventional intermediate-frequency coils. The control of the initial tuning operation and the frequency drift of the local oscillator may require special techniques in which automatic frequency control may be required. The adoption of the proposal would, therefore, give little improvement in the short term, with existing receivers, but would offer the chance of substantial improvement in the future, without any disadvantages under present-day conditions.

REFERENCES

EDEN, H. [December, 1967] Perspectives of replanning LF/MF broadcasting, *EBU Rev.*, **106-A**, 242-251.

SCART [May, 1966] Contribution à l'étude de la fréquence intermédiare pour les récepteurs de radiodiffusion destiné à recevoir les émissions en ondes kilométriques et hectométriques. (Contribution to the study of the intermediate frequency for broadcasting receivers for LF and MF emissions). Fascicule de documentation SCART, VII, ed. SDSA, 16, rue de Presles, Paris, France.

CCIR Documents

[1963-66]: X/161 (France).

[1970-74]: 10/273 (France).

REPORT 1059-1

CHARACTERISTICS OF SINGLE-SIDEBAND SYSTEMS IN HF BROADCASTING

(Question 44/10)

(1986-1990)

1. Introduction

This Report has been developed in recognition that consideration is being given to the progressive introduction of single-sideband broadcasting systems into the bands now allocated to the Broadcasting Service at HF. Introduction of this form of broadcasting can be accomplished with greater technical harmony if there is sufficient guidance concerning the technical parameters involved. The following considerations will be concentrated on single-sideband amplitude modulation with synchronous demodulation. With respect to a necessary transition period from DSB to SSB, some consideration must also be given to the reception of SSB signals with reduced carrier by receivers with envelope detection. At the end of the transition, all of the advantages of SSB transmissions could then be realized, as follows:

- a more efficient utilization of the frequency spectrum and a reduction of interference;
- the capability of improving the required protection ratio between adjacent channels in the case of a sufficient carrier reduction;
- the capability of improving the quality of reception, in particular under poor propagation conditions (selective fading), with SSB receivers.

The modulation technique considered most suitable for the achievement of bandwidth saving is some form of SSB system. WARC-79 Recommendation 501 and No. 302 of the Radio Regulations suggest the use of SSB emissions to the maximum possible extent in AM systems. There are two types of such systems, namely single-sideband (SSB) and compatible single-sideband (CSSB).

The CSSB system is not suitable for use in amplitude-modulated sound broadcasting, principally because of its increased distortion; furthermore, a greater radiofrequency bandwidth is needed and adequate suppression of out-of-band emissions is likely to be difficult at HF.

In the event of the introduction of single-sideband amplitude modulation broadcasting, it would seem desirable to use the definitions existing in Recommendation 326.

According to this Recommendation, the carrier component is defined in relation to the peak envelope power P_p of a radio transmitter, by the acceptable intermodulation level D_n.

For single-sideband broadcasting transmitters, the acceptable intermodulation level D_n determines the non-linear distortion (quality), and the out-of-band radiation (adjacent channel interference).

In single-sideband reduced carrier systems, the precision of the locally re-inserted carrier is important for the reception quality.

The system parameters for a future SSB-system for sound broadcasting in band 7 (HF) must be chosen in such a way that the different requirements of the transition period (reception of SSB-signals with receivers using envelope detection) as well as those of the period thereafter, when only receivers with synchronous demodulation will be used, are taken into account [CCIR, 1978-82a].

The system specification of SSB is presented in Recommendation 640.