

RECOMMENDATION ITU-R BS.644-1<sup>\*,\*\*</sup>**Audio quality parameters for the performance of a high-quality sound-programme transmission chain**

(1986-1990)

The ITU Radiocommunication Assembly,

*considering*

- a) that the performance characteristics of international sound-programme circuits used for programme exchange are based on a hypothetical reference circuit (HRC) defined in ITU-T Recommendation J.11;
- b) that the performance of a realistic international sound-programme circuit shorter or longer than the HRC can be derived by calculation according to rules given in ITU-T Recommendation J.25;
- c) that the performance of the sound-programme transmission chain from the broadcasting house to the output of the receiver should therefore be based on a reference chain;
- d) that sound-programme signals intended for international exchange are conveyed from the broadcasting house to the international sound-programme centre (ISPC) on national circuits which are either the same as, or similar to, the national circuits which form part of the sound-programme transmission chain;
- e) that the performance of the sound-programme transmission chain should be described by a number of parameters;
- f) that for reasons of comparison, clearly defined parameters measured using test signals, should be used;
- g) that these parameters and their target values should be based on the perceptibility of the impairments by the human ear,

*recommends*

- 1** that the performance characteristics of a high-quality sound-programme transmission chain should be based on the reference chain described in Annex 1;
- 2** that the performance of the reference transmission chain from the output of the broadcasting house to the output of the receiver should be described by the parameters listed in Annex 2;

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\* This Recommendation should be brought to the attention of Telecommunication Standardization Study Group 9.

\*\* Radiocommunication Study Group 6 made editorial amendments to this Recommendation in 2002 in accordance with Resolution ITU-R 44.

- 3 that the limits given in Annex 3 be used as realistic target values for these parameters;
- 4 that for the purposes of planning the entire broadcasting chain from the microphone output to the receiver output, the subjective threshold values for quality parameters given in Annex 4 should be taken for guidance.

## ANNEX 1

### Reference sound-programme transmission chain

The reference sound-programme transmission chain is represented, for the purpose of this Recommendation, by Fig. 1 in which A, B, and C are 0 dBrs audio signal interface points, permitting performance comparison:

- A : interface between the programme source (broadcasting house) and the sound-programme circuit system;
- A-B : sound-programme circuit system comprising a single cable, radio, communication satellite or optical-fibre section;
- B : interface between the sound-programme circuit and emission-reception systems;
- B-C : emission-reception system, comprising a single broadcasting emitter or a single cable network and a high-quality monitoring receiver under optimum conditions;
- C : audio signal interface at the output of the receiver.

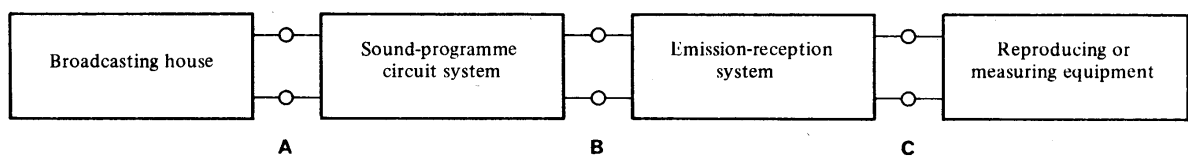


FIGURE 1 – Reference sound-programme transmission chain

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The EBU has published Recommendation R.50 to define the polarity of acoustic signals, electrical audio signals, audio signals at connectors, magnetic signals on audio and video tapes, digital audio signals and acoustic pressure signals for loudspeakers.

## ANNEX 2

**List of audio quality parameters**

The audio quality parameters listed below are those which are considered to be the most important in the analogue environment. Some parameters are also appropriate for connections incorporating digital systems, but some further parameters need to be specified for testing these digital systems.

All test signals should be at the measurement level (see Recommendation ITU-R BS.645) unless otherwise stated. For stereo, measurements should be made in both A and B channels.

**1 Nominal bandwidth**

The effective transmitted frequency band (for both A and B channels, in the case of stereo).

**2 Amplitude/frequency response**

The response should be expressed relative to the level at the reference frequency of 1 kHz.

**3 Group-delay variation**

The results of measurements throughout the nominal bandwidth should be expressed relative to the minimum group delay.

**4 Non-linear distortion**

Total harmonic distortion (THD) may be measured using a single-tone test signal at +9 dBu<sub>0s</sub>, at frequencies in the range from 40 Hz to 1 kHz, and the result expressed as a “separation” value (i.e. the difference in level between the test signal and the harmonics, expressed in dB).

Distortion may, alternatively, be measured using the total difference frequency (TDFD) method. This is an intermodulation test in which the test signal comprises two tones at frequencies chosen so that second-order and third-order intermodulation products (i.e. one of each) occur close in frequency, and may be measured together by a meter with a selective filter. Tests in the UK have shown that the TDFD method is less sensitive to noise than THD, and is not restricted to the use of test signals at frequencies below 5 kHz. This could be important when testing pre-emphasised connections, although care must be taken to avoid the possibility of overloading resulting from the pre-emphasis.

The TDFD method is described in Annex 5.

The duration for which the test signals are transmitted should be restricted in accordance with the appropriate ITU-T Recommendations of the N series.

## 5 Error in reconstituted frequency

The error in reconstituted frequency should not exceed the values in Annex 3, for individual channels.

For stereo, any differential frequency shift between A and B channels is unacceptable.

## 6 Error in amplitude/amplitude response

A 1 kHz tone is applied to the input of the transmission chain at +9 dBu0s and at –31 dBu0s alternatively. The output levels should be measured selectively, and the difference should be within the limits specified in Annex 3.

The duration for which the test signal is transmitted should be restricted in accordance with the appropriate ITU-T Recommendations of the N series.

## 7 Level stability

The difference in the level of a constant reference signal, at the measurement level, applied to the input of the transmission chain, from its nominal level should be determined for a 24 h period. This difference may vary over the 24 h period, and include a fixed adjustment error.

For stereo, the level stability should be determined concurrently for both A and B channels.

## 8 Noise (and single-tone interference)

Weighted noise voltage level (and, where appropriate, single-tone interference voltage level) is measured in accordance with Recommendation ITU-R BS.468. The permitted maximum signal-to-(weighted) noise ratio (PMS/N, expressed in dB) is determined as the difference between the permitted maximum signal (PMS) and the weighted noise levels.

- Idle-channel noise.
- Programme-modulated noise, measured with a 60 Hz sinusoidal test signal applied to the input of the transmission chain at +9 dBu0s and –31 dBu0s. The fundamental and low-order harmonics are suppressed at the input to the measuring instrument by a high-pass filter with a cut-off frequency between 200 Hz and 400 Hz. The insertion loss of the filter should be at least 56 dB at 60 Hz, taking into account the attenuation of the weighting filter (24 dB at this frequency).

The duration for which the test signal is transmitted should be restricted in accordance with the ITU-T Recommendations of the N series.

- If necessary, steady single-tone interference should be measured selectively at the frequencies at which tones are detected or may be expected. The measurement should be made using a bandpass filter (having effectively 0 dB insertion loss in the passband) in conjunction with the noise measuring apparatus described in Recommendation ITU-R BS.468, or using a spectrum analyser and correcting the measured level by the corresponding weighting factor.

## 9 Disturbing modulation by power supply

This is the ratio of a 1 kHz sinusoidal test signal to the highest level unwanted side-component resulting from modulation of that test signal caused by interference from conventional a.c. (50/60 Hz) line power supply sources.

## 10 Stereo: level difference between A and B channels

The difference in level between the A and B outputs of the transmission chain should be measured with the same sinusoidal test signal applied to both channel inputs simultaneously.

## 11 Stereo: phase difference between A and B channels

The difference in phase between the A and B outputs of the transmission chain should be measured with the same sinusoidal test signal applied to both channel inputs simultaneously.

Conservation of the polarity of audio signals is recommended.

## 12 Stereo: crosstalk between A and B channels

### – *Linear cross-talk*

A sinusoidal test signal is applied to each channel in turn, and the level of the signal in the other channel is measured selectively. The cross-talk attenuation (expressed in dB) is the difference between the levels of the signal in the two channels.

### – *Non-linear cross-talk*

The conventional test signal for simulating sound-programme signals specified in ITU-T Recommendation J.19 is applied to each channel in turn. If the other channel is influenced by non-linear cross-talk, an increase in the weighted noise level may be observed.

The increased noise voltage  $N_s$  formed by the sum of two noise contributions: the idle-channel noise  $N_0$  and the non-linear cross-talk noise  $N_{cT}$ . The latter voltage can be calculated using the following formula:

$$N_{cT} = \sqrt{(N_s)^2 - (N_0)^2}$$

The non-linear cross-talk is expressed as the ratio of the permitted maximum signal to this level of non-linear cross-talk noise (i.e.  $PMS/N_{cT}$ ).

## ANNEX 3

TABLE 1

Realistic target values for the reference  
sound-programme transmission chain

Parameter (for details see Annex 2)	Test signal		Sound-programme circuit system (A-B) (1)	Emission- reception system (B-C) (1)	Whole reference chain (A-C) (1)
	Frequency	Level			
Nominal bandwidth (see § 1)	–	–	40 Hz-15 kHz	40 Hz-15 kHz	40 Hz-15 kHz
Amplitude/frequency response (see § 2)	40 Hz -125 Hz 125 Hz - 10 kHz 10 kHz- 14 kHz 14 kHz- 15 kHz	–12 dBu0s	+0.2/ –1.0 dB +0.2/ –0.2 dB +0.2/ –1.0 dB +0.2/ –1.4 dB	+0.4/ –1.5 dB +0.4/ –0.4 dB +0.4/ –1.5 dB +0.4/ –2.3 dB	+0.5/ –2.0 dB +0.5/ –0.5 dB +0.5/ –2.0 dB +0.5/ –3.0 dB
Group-delay/variation (see § 3)	40 Hz 15 kHz	–12 dBu0s	18 ms 4 ms	37 ms 8 ms	55 ms 12 ms
Total harmonic distortion (see § 4)	40 Hz -125 Hz 125 Hz - 1 kHz	+9 dBu0s	–46 dB –52 dB	–42 dB –48 dB	–40 dB –46 dB
Error in reconstituted frequency (see § 5)	Any	–12 dBu0s	0.2 Hz	0	0.2 Hz
Error in amplitude/amplitude response (see § 6)	1 kHz	+9 dBu0s –31 dBu0s	0.4 dB	0.8 dB	1.0 dB
Level stability (over 24-h period) (see § 7)	1 kHz	–12 dBu0s	0.6 dB	0.8 dB	1.0 dB
Noise and single-tone interference: – idle-channel noise – programme-modulated noise – programme-modulated noise – single-tone interference (see § 8)	– 60 Hz 60 Hz –	– –31 dBu0s +9 dBu0s –	53 dB 53 dB 43 dB 73 dB	56 dB 56 dB 46 dB 76 dB	51 dB 51 dB 41 dB 71 dB
Disturbing modulation by power supply (see § 9)	1 kHz	0 dBu0s	51 dB	47 dB	45 dB
Level difference between A and B channels (see § 10)	40 Hz -125 Hz 125 Hz - 10 kHz 10 kHz- 14 kHz 14 kHz- 15 kHz	–12 dBu0s	1.0 dB 0.5 dB 1.0 dB 2.0 dB	1.0 dB 0.5 dB 0.5 dB 1.0 dB	1.5 dB 1.0 dB 1.5 dB 2.5 dB
Phase difference between A and B channels (see § 11)	40 Hz 40 Hz -125 Hz 125 Hz - 10 kHz 10 kHz- 15 kHz 15 kHz	–12 dBu0s	15° (2) 10° (2) 15°	10° (2) 8° (2) 10°	20° (2) 15° (2) 25°
Cross-talk between A and B channels: – linear cross-talk  – non-linear cross-talk (see § 12)	40 Hz 40 Hz -300 Hz 300 Hz - 4 kHz 4 kHz- 15 kHz 15 kHz  Sound programme simulating signal (see ITU-T Recommendation J.19)	0 dBu0s	46 dB (2) 56 dB (2) 46 dB 60 dB	22 dB (2) (2) 26 dB (3)	21 dB (2) 35 dB (2) 25 dB (3)

(1) The values given for sections A-B and B-C, and for the whole transmission chain (A-C) are desirable, and are to be taken into account when designing new national broadcasting networks.

(2) Values within this range are obtained by linear interpolation between the values for adjacent ranges on a graph with a logarithmic frequency scale.

(3) Administrations are requested to suggest target values.

## ANNEX 4

Subjective threshold values are thresholds of perception, found by subjective statistical research under ideal listening conditions. They have been ascertained from test results obtained by ITU-R members and from international literature.

TABLE 2  
Subjective threshold values

Parameter	Frequency of test signal	Subjective threshold value
Amplitude/frequency response (see § 2 of Annex 2)	40 Hz -125 Hz 125 Hz - 10 kHz 10 kHz- 14 kHz 14 kHz- 15 kHz	$\pm 1.0$ dB $\pm 0.5$ dB $\pm 1.0$ dB $\pm 2.0$ dB
Group delay variation (see § 3 of Annex 2)	40 Hz-15 kHz	(see Note 1)
Non-linear distortion THD (see § 4 of Annex 2)	40 Hz-1 kHz	-52 dB (see Note 2)
Error in reconstituted frequency (see § 5 of Annex 2)	Any	0.25 Hz
Error in amplitude/amplitude response (see § 6 of Annex 2)	1 kHz	(see Note 1)
Level stability (over a 24-h period) (see § 7 of Annex 2)		1 dB (see Note 3)
Noise and single-tone interference: – idle-channel conditions – test signal level: +9 dBu0s – test signal level: -31 dBu0s – single-tone interference (see § 8 of Annex 2)	– 60 Hz 60 Hz –	70 dB (see Note 1) (see Note 1) 80 dB
Disturbing modulation by power supply (see § 9 of Annex 2)		(see Note 1)
Level difference between A and B channels (see § 10 of Annex 2)	40 Hz -125 Hz 125 Hz - 10 kHz 10 kHz- 14 kHz 14 kHz- 15 kHz	2.0 dB 0.5 dB 1.5 dB 2.0 dB
Phase difference between A and B channels (see Note 4) (see also § 11 of Annex 2)	40 Hz 40 Hz -125 Hz 125 Hz - 10 kHz 10 kHz- 15 kHz 15 kHz	45° (see Note 5) 30° (see Note 5) 90°
Cross-talk between A and B channels: – linear cross-talk  – non-linear cross-talk (see § 12 of Annex 2)	40 Hz 40 Hz -300 Hz 300 Hz - 4 kHz 4 kHz- 15 kHz 15 kHz  Sound-programme simulating signal (ITU-T Recommendation J.19)	15 dB (see Note 5) 20 dB (see Note 5) 15 dB  (see Note 1)

NOTE 1 – Administrations are invited to contribute figures for this value.

NOTE 2 – This figure assumes that the distortion is predominantly second and third harmonic.

NOTE 3 – A change of 1 dB is perceptible only if it is a sudden change.

NOTE 4 – The tolerances given for phase differences between A and B channels are for stereo listening, and they provoke unacceptable variations in the amplitude/frequency response of the mono (A + B) signal. The requirements for the mono signal therefore place a more stringent specification on the phase difference between A and B channels, and this is taken into account in Table 1 of Annex 3.

NOTE 5 – Values within this range are obtained by linear interpolation between the values for the adjacent ranges on a graph with a logarithmic frequency scale.

## ANNEX 5

**Special methods for measuring  
audio quality parameters****1 Measurement of non-linear distortion at high frequencies in pre-emphasized circuits****1.1 Introduction**

Non-linear distortion in sound-programme circuits is most often measured by the total harmonic distortion (THD) method, in which a pure tone is input to the circuit as a test signal. At the output, the pure tone is removed from the signal by a filter, either high-pass or band-stop, and the signals remaining are measured as THD, with their magnitude expressed as a ratio with respect to the magnitude of the whole output signal.

Because non-linear distortion is commonly greatest at the highest levels in the circuit, it is usually tested at or very near to the maximum level of which the circuit is capable, in broadcasting practice, usually at the permitted maximum level (PML), i.e. +9 dBu0s.

If the frequency of the test signal is more than one-third the band limiting frequency, e.g. if in a 15 kHz circuit the testing signal is above 5 kHz, then any third harmonic components produced in the circuit will not reach the output, and the THD test is valueless.

Thus non-linearity in the upper half-decade of a band limited circuit, e.g. between 5 kHz and 15 kHz in a 15 kHz circuit, can be tested only by an intermodulation method, in which two tones at frequencies  $f_1$  and  $f_2$  are mixed together to produce the test signal, and the major in-band intermodulation products between them, at frequencies  $f_2 - f_1$  for even-order distortion and  $2f_1 - f_2$  for odd-order distortion, are measured and summed r.m.s. as a measure of the non-linearity of the circuit.

This test in fact measures the intermodulation, or “beat”, products. It should be noted that it is the spurious inharmonic products that cause most, if not all, of the subjective degradation of sound quality due to non-linearity, rather than the harmonics which tend to simply add “brightness” to the sound.

The use of high-frequency preemphasis, for the purpose of reducing noise, increases the incidence and severity of such distortion and a method of measuring it is clearly essential.

**1.2 Test method**

Of the different methods that have been proposed for intermodulation testing, the preferred method is that in which the two test frequencies  $f_1$  and  $f_2$  are related as  $2f_0$  and  $3f_0 + \Delta$ , where  $\Delta$  is small. The major in-band intermodulation products are consequently at  $f_0 + \Delta$  and  $f_0 - \Delta$  for even-order and odd-order non-linearities respectively. These products are measured together through a band-pass filter of centre frequency  $f_0$  and with a narrow pass-band of  $\pm\Delta$ , i.e.  $2\Delta$  wide. The filter removes the initial test signals and at the same time much of the in-band noise, making measurement possible to levels that are more than 10 dB below the broadband noise in the circuit.



The test frequencies need to be as high in the band as possible, but not so high as to risk attenuation of the higher frequency components due to the roll-off of response near the band edge, which could affect the accuracy of the non-linearity measurement. In the test adopted in the draft revision of IEC Publication 268 – Sound system equipment: Part 3, Amplifiers – the two test frequencies are 8000 Hz and 11950 Hz. In this configuration,  $f_0$  is 4000 Hz and  $\Delta$  is  $-50$  Hz. This is shown schematically in Fig. 2.

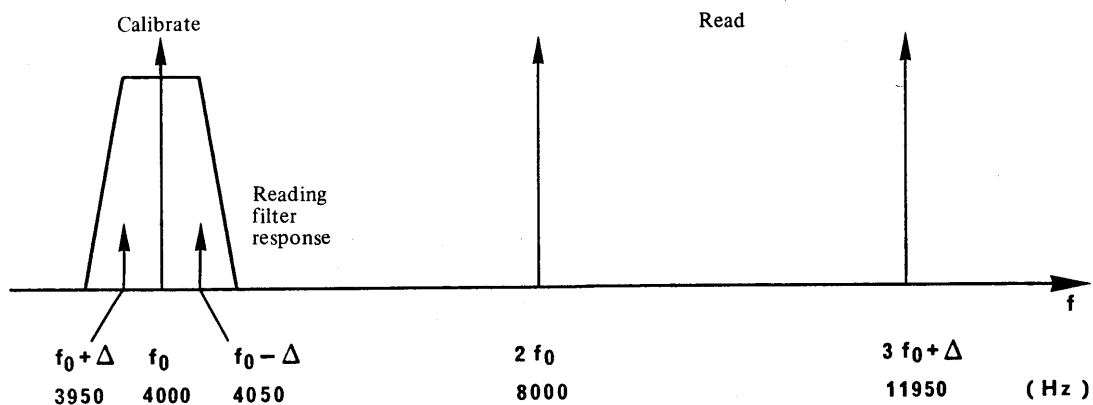


FIGURE 2 – Frequencies of testing tones and in-band distortion products

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In the above method, the amplitudes of the two test tones at  $2f_0$  and  $3f_0 + \Delta$  are equal. The peak amplitude of a mixture of two equal amplitude tones, whose frequencies are not related by small whole numbers, is twice the peak amplitude of each individual tone. Thus, to test the circuit to a peak amplitude equal to that of a single sine wave at +9 dBu0s (PML), the amplitude of each tone must be +3 dBu0s. The power of this sum signal is twice that of each tone, i.e. equal to a single sine wave of +6 dBu0s in the above case. However it must be emphasized that the limitation on the level that can be carried by a programme circuit is always the peak amplitude, i.e. peak voltage, peak current, peak deviation, etc. which is twice the amplitude of each tone, i.e. 6 dB greater.

The r.m.s. magnitude of the sum of the in-band intermodulation products, measured through the band-pass filter, is expressed as a ratio with respect to the r.m.s. value of one of the component tones (sine waves) in the test signal, i.e. to a single sine wave at +3 dBu0s in the above case. This ratio is close numerically to that obtained by a THD measurement of the same non-linear mechanism, so that its significance is readily appreciated by staff more familiar with the older method. The magnitude of the test signal at the output may be measured directly by a true r.m.s. meter and using a level 3 dB lower for the reference level. This method is suitable when the magnitude of the level needs to be established only in the case of a measuring instrument, whose indicating device reads r.m.s.

### 1.3 Need for calibration signal

However, it is important that signals passing along sound-programme circuits can be monitored by the types of meters commonly used for level measurements in broadcasting studios. These do not read pure r.m.s. levels. Both the VU meter, which reads average, and the “peak programme meter” (PPM), which reads quasi-peaks, are calibrated using a sine wave signal and should therefore be sufficiently accurate for reading levels of single sine waves. They do not read true r.m.s., nor true peaks, for any other kind of signal, and in particular for a two-tone test signal. However this latter level can be inferred precisely from the readings of such meters by first sending, prior to the main test signal, a single sine wave calibrating signal whose level is standardized with respect to that of the two-tone test signal.

The calibrating sine wave is at frequency  $f_0$ , e.g. 4 kHz when  $2f_0$  is 8 kHz. If its level was +3 dBu0s, it could be used directly to calibrate the reference level for the distortion measurements. However a signal at Alignment Level, 0 dBu0s, is more suitable, first for reading on VU meters, and secondly for testing circuits with high-frequency preemphasis, as is shown in Table 3. As the signal is a sine wave, it can be read accurately by standard level meters, and with the frequency  $f_0$ , it passes directly through the band-pass filter used in measuring distortion, and thus provides a check on the calibration of the distortion reading instrument.

### 1.4 Measurement in preemphasized circuits

The two-tone method, measuring total difference-frequency distortion (TDFD), is superior to the THD method for measurements at all frequencies, but at high frequencies it is the only one that is effective. So far, it has been assumed that the transmission channel has a flat amplitude response, and that its peak amplitude capability is the same at all frequencies.

However, in broadcasting, such a capability can be expected only from amplifiers and studio quality digital equipment. In almost all other signal-handling processes, in recording, transmission and FM emission, the high frequencies are preemphasized to a standardized characteristic before passing through the main signal-handling process. At the output, the signal is deemphasized in a complementary manner. The purpose is to reduce the effect in the output of noise at the higher frequencies, on the implicit assumption that programme material rarely contains high-frequency components near the peak capability of the circuit. While this assumption may not be true for some kinds of programme material which is already preemphasized, especially some present-day programme material, the use of preemphasis is a convention which broadcasters have for a long time agreed to use.

A transmission circuit is tested conventionally by feeding a test signal into it at a point where the frequency response is flat, i.e. before any preemphasis. Likewise the output is read conventionally at a point where the frequency response is again flat, i.e. after deemphasis. Thus any test signal that is fed into the input of the circuit will reach the main transmission path in a preemphasized form. A high-frequency signal that is already optimized to fully load a flat circuit would therefore drastically overload the preemphasized path.

For example, if the IEC test signal comprised of a mixture of two tones at 8 000 Hz and 11 950 Hz, each at +3 dBu0s, is passed through a circuit preemphasized to ITU-T Recommendation J.17 with 6.50 dB loss at 800 Hz, its peak level, initially identical with that of a single sine wave at +9 dBu0s,

will rise after preemphasis to the equivalent of +14.89 dBu0s. This is 2.89 dB above the overload point of +12 dBu0s specified for ITU-R digital coding methods A1, A2, A3 and A4, and only 0.11 dB below the overload point of +15 dBu0s for Method B5.

If however such a signal, optimized for testing a flat path, is first deemphasized to the same characteristic before feeding the input, it will reach the preemphasized path in exactly the same form, and test it to exactly the same extent, as it would have done unemphasized in a flat path. Such deemphasis does not require a frequency-sensitive network as the signal is comprised of only the two component tones at  $2f_0$  and  $3f_0 + \Delta$ , the same effect is produced equally well by attenuating each by the appropriate deemphasis within the generator before they are mixed and fed to the input. At the output, the magnitude of the intermodulation products is compared with the reference level, which is taken as 3 dB below the r.m.s. sum of the two testing tones. Note that no complementary preemphasis is needed before reading the output signal. The ratio of distortion to total signal is measured as it appears in the output.

For the method in which the signal level is monitored by broadcasting level meters using a calibration tone at  $f_0$ , near 4 kHz, the level of the sum of the test tones, after deemphasis at the output, is inferred from the magnitude of the calibration tone. The difference is then estimated between the calibration level and the mean between the powers of the two testing tones, tabulated as “r.m.s. mean 8 kHz + 12 kHz” in Table 3. The magnitude of the intermodulation products is then adjusted by this difference and read as a ratio to the calibration level. This involves attenuation of the distortion products by 3 dB in the flat condition and gains of various amounts for the different preemphasis standards. A schematic layout of the equipment is shown in Fig. 3.

Table 3 illustrates the levels in various parts of the circuit. It assumes that while no intermodulation products ( $-\infty$  dBu0s) are present in the input test signal, intermodulation products at the same level, taken arbitrarily for the purpose of illustration as  $-37$  dBu0s, are produced in each type of preemphasized path, when the test signal levels are identical. It thus demonstrates how, with increasing preemphasis, the same ratio of high-frequency intermodulation in the preemphasized path produces a greater ratio of in-band distortion products in the deemphasized output.

### 1.5 Use with different preemphasis characteristics

Although the frequencies of the test signals in the figures are those recommended in IEC Publication 268, Part 3, i.e.  $2f_0$  at 8000 Hz and  $3f_0 + \Delta$  at 11950 Hz, slightly different sets of frequencies are used for the different preemphasis standards used in broadcasting. Figure 4 shows schematically how each set of frequencies is offset from its neighbour. An offset ratio of approximately 1.03 is sufficient to ensure that the signals for one preemphasis standard, with its special adjustments of level and gain, cannot be received by mistake in equipment set to receive another standard requiring different adjustments. With the pass-bands of the measurement filters each approximately  $\pm 1\%$  about  $f_0$ , any calibration signal received from a neighbouring channel will be attenuated by at least 30 dB. This procedure, if the control settings are inappropriate for the incoming signal, alerts the receiving operator, and makes measurement with the wrong preemphasis characteristic virtually impossible.

TABLE 3

Signal levels, gain adjustments and relative values for intermodulation measurements

Pre-emphasis type		Flat	ITU-T Recommen- dation J.17	50 $\mu$ s/15 $\mu$ s	50 $\mu$ s
<i>Input level (dBu0s) before pre-emphasis</i>					
calibration	4 kHz	0.0	0.0	0.0	0.0
component $f_1$	8 kHz	+3.0	-2.6	-3.7	-5.6
component $f_2$	$\approx$ 12 kHz	+3.0	-3.1	-5.2	-8.8
distortion	$\approx$ 4 kHz	$-\infty$	$-\infty$	$-\infty$	$-\infty$
<i>Level (dBu0s) in pre-emphasized path</i>					
calibration	4 kHz	0.0	+3.5	+3.5	+4.1
component $f_1$	8 kHz	+3.0	+3.0	+3.0	+3.0
component $f_2$	$\approx$ 12 kHz	+3.0	+3.0	+3.0	+3.0
distortion	$\approx$ 4 kHz	-37.0	-37.0	-37.0	-37.0
<i>Output level (dBu0s) after de-emphasis</i>					
calibration	4 kHz	0.0	0.0	0.0	0.0
component $f_1$	8 kHz	+3.0	-2.6	-3.7	-5.6
component $f_2$	$\approx$ 12 kHz	+3.0	-3.1	-5.2	-8.8
r.m.s. mean	8 kHz + 12 kHz	+3.0	-2.8	-4.4	-6.9
distortion	$\approx$ 4 kHz	-37.0	-40.5	-40.5	-41.1
<i>Distortion ratio (dB)</i>		-40.0	-37.7	-36.1	-34.2

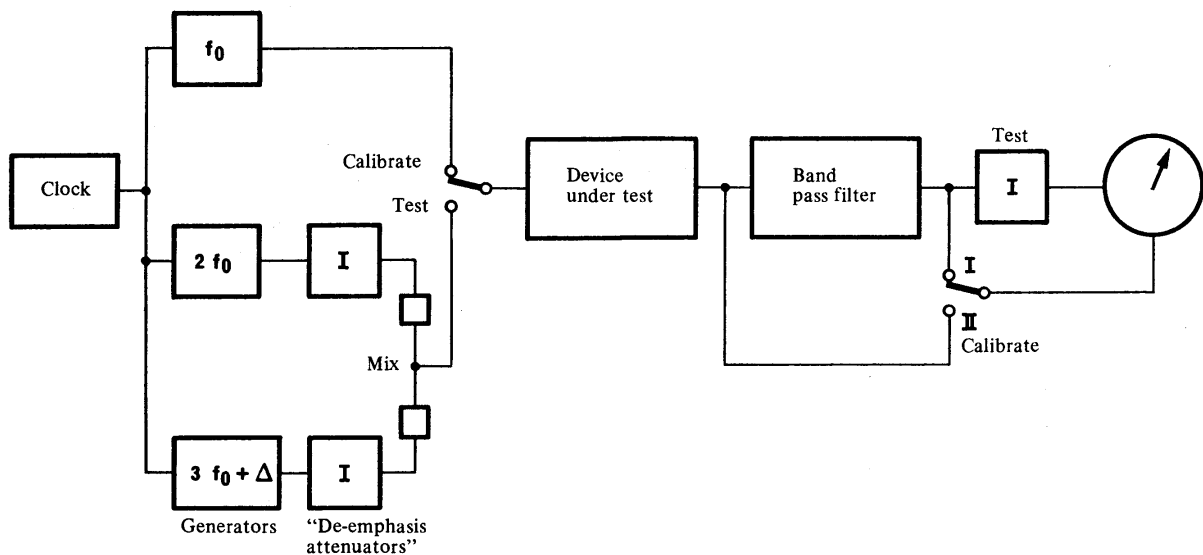


FIGURE 3 – Block schematic of measuring equipment

In mode I, calibrate and test signals are sent sequentially.

In mode II, test (two-tone) only signal is sent. Generator frequencies, attenuations and band-pass filters change with pre-emphasis.

It is anticipated that the flat or ITU-T Recommendation J.17 characteristics will be used in testing transmission chains, and the 50  $\mu$ s/15  $\mu$ s and 50  $\mu$ s characteristics in testing emission chains. A chain that is made up of links with different characteristics is tested for the one with the greatest high-frequency preemphasis.

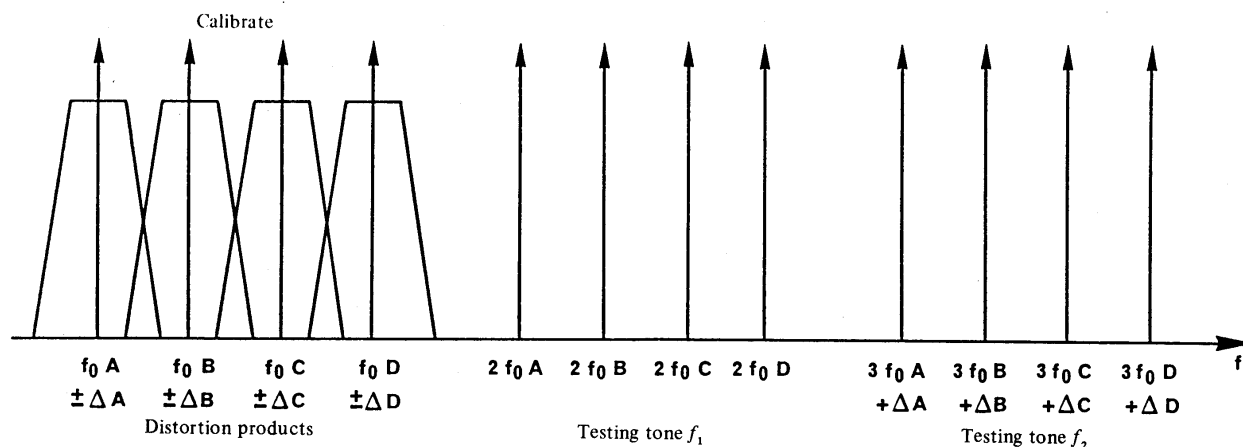


FIGURE 4 – Testing tones and distortion products, with different pre-emphasis characteristics A, B, C and D

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## 1.6 Conclusion

The method described herein provides a straightforward method of measuring non-linear distortion at the upper end of the audio frequency band. It is easily used and allows measurements to be made in a variety of preemphasized circuits with a minimum risk of error in the alignment of levels or in the choice of preemphasis characteristics.

## 2 Method of measurement of impulsive noise

During the transmission of sound signals, short duration noise frequently appears, which when acoustically reproduced is perceptible in the form of clicks and is disturbing to the listener. For the quality parameter “impulsive noise” there is at present no appropriate measuring procedure available which takes into account the subjective disturbing effect and the so-called annoyance factor.

Comprehensive subjective-statistical studies have been made in the German Democratic Republic on all the relevant factors that influence the subjective annoyance of pulses. As a result of these investigations the following method of measurement of impulsive noise has been suggested:

For the tone-compensated measurement of impulsive noise it is suggested that a “pulse weighted peak value  $\hat{U}_{IW}$ ” which is defined as follows, should be determined:

$$\hat{U}_{IW}(\text{dB}) = 20 \cdot \log \left( \frac{\hat{U}}{U_0} \right) + 20 \cdot \log (k_1 \times k_2 \times k_3)$$

where  $k_1$ ,  $k_2$  and  $k_3$  are objective rating factors for the pulse duration  $t_i$ , pulse shape  $u(t)$  and pulse repetition frequency  $f_p$  which result from predetermined subjective evaluation tables. The determination of the value  $\hat{U}_{IW}$  can only be made with a computer-assisted method of measurement. To do this, the arriving pulse is first stored and analysed. The pulse amplitude  $U$  and the pulse

duration  $t_i$  can directly be measured, likewise the period until the arrival of the next pulse. The pulse shape can be determined by spectral analysis by means of FFT analysers or (simpler) by comparison with previously determined standard pulse shapes. From the measured values for  $t_i$ ,  $u(t)$  and  $f_p$  the factors  $k_1$ ,  $k_2$  and  $k_3$  can be determined using the subjective evaluation tables, and finally the value  $\hat{U}_{IW}$  can be calculated.

When evaluating the quality of studio equipments and transmission links, the value  $\hat{U}_{IW}$  may serve as a measure of the important parameter “impulsive noise”. However, further studies are required to determine an admissible value.

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