CHAPTER 14

FIDELITY AND DISTORTION

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SECTION 1 : INTRODUCTION

(i) Fidelity (ii) Types of distortion (iii) Imagery for describing reproduced sound.

(i) Fidelity

True fidelity is perfect reproduction of the original. In the case of an amplifier, true fidelity means that the output waveform is in all respects an amplified replica of the input waveform.

In acoustical reproduction, true fidelity is achieved if the listener has the same aural sensations that he would have if present among the audience in the studio or concert hall.

In practice, true fidelity can only be regarded as an ideal to be aimed at.

The concept of " hole in the wall " listening put forward by Voigt (e.g. Ref. A49) is a useful device. The listener, surrounded by his listening room, is imagined as being in the concert hall and able to hear directly, through his open window, the sound coming from the stage, together with echoes. Echoes from his own room will come from all directions, but echoes from the concert hall only come through the open window. This is probably the nearest approach to true fidelity which there is any hope of approaching with single channel transmission. See also Ref. A50.

The purpose of high fidelity reproduction of music is to satisfy a particular listener, who is primarily interested in the emotions arising from what he hears. The complete process involves sensations and emotions which cannot be treated objectively and must bring in personal preferences and differences of opinion.

It is manifestly impossible to reproduce at the two ears of the listener an exact equivalent of the sounds which he would hear in the concert hall. The greatest deficiencies are probably the single point sound reproducer (the loudspeaker) and the volume level. The use of stereophonic reproduction is briefly mentioned in Chapter 20 Sect. 6(v)—it is impracticable with the existing radio and disc recording techniques. In the home it is rarely that orchestral reproduction is heard at the same maximum volume level as in the concert hall. This is the real dilemma from which there is no satisfactory escape. If we listen at a lower volume level we suffer attenuation of the low and high frequencies. If we apply tone correction in the form of bass and treble boost, it is difficult to gauge the correct amount, and most correcting circuits only give a rough approximation to the ideal. If we leave the adjustment of tone controls to the listener, he will adjust them to please himself and the result is usually far from a true reproduction of the original.

Refs. A10, A20.

(ii) Types of distortion

Distortion is lack of fidelity. In the case of an amplifier, distortion occurs when the output waveform differs in some respect from the input waveform. The purpose of the amplifier, and also of the whole equipment, is to reproduce the input waveform, not a band of frequencies. The latter may be necessary in order to achieve the former, but is only a means to an end.

Distortion may be grouped into six main classes.

1. Non-linear distortion (also known as amplitude distortion), and resulting in
1a. Harmonic distortion and
1b. Intermodulation distortion.
2. Frequency distortion (unequal amplification of all frequencies).
3. Phase distortion.
4. Transient distortion.
5. Scale distortion (or volume distortion).
6. Frequency modulation distortion.

There are other features in reproduction which are not normally classed as distortion, although they affect the listener as being untrue to the original. These include background noise and the use of a single point source of sound. Background noise includes needle scratch, hum and adjacent channel whistles. There are also some types of distortion peculiar to loudspeakers—see Chapter 20 Sect. 7.

(iii) Imagery for describing reproduced sound

The following is based largely on Ref. A48, and is merely a proposal which has not been generally accepted. It is included for general interest.

(a) Frequency Range Notation

Extreme Lows	Below 100 c/s
Lows	100— 300 c/s
Lower Middles	300— 800 c/s
Upper Middles	800—1500 c/s
Lower Highs	1500—4000 c/s
Highs	4000—8000 c/s
Extreme Highs	Above 8000 c/s

(b) Distortion

General : Dirty, non-linear distortion.
Overload : Hash-up, mush-up.
Thump (sudden rectification when signal hits bottom).
Sound often becomes strident if harmonic energy peaks in the lower highs.
Records : Fuzz or lace. Inability of stylus to track at high groove curvatures.
Crackle : Same as fuzz, but occurring principally on high-amplitude peaks.
Rattles or buzz, rub or wheeze.
Swish—scratch periodic with rotation of record.
Carbon Microphones : Frying, popping, sizzle.
Sub-harmonics : Breakup, birdies, tweets.
Intermodulation : Harsh, rough.

Cross-over distortion : Marbles, garble.
Transient distortion : Hang-over.
Attack—good or slurred.
Intermodulation with peak in the high-frequency region :
Violins sound wiry.
Male voices have kazoo.
Brass instruments show jamming in upper octaves.

(c) General Terms

Position presence : Localization, mass of sound ; advance, come forward, stand out ; distant, dead, recede, lost.
Intimacy presence : Intimate, rapport.
Detail presence : Transparent, translucent, clear, opaque, acoustic fog, veiled, muddy.
Source size : Live ; broad, volume, floods out, big tones, well-focused tones ; dead and flat, compressed, from a hole in the wall, out of a barrel.
Realism : Presence, natural, life-like, pleasing ; canned music.
Reproduction : Realistic, perfect, adequate.

	Lows	Lower Middles	Upper Midd les	Lower Highs	Highs	Extreme Highs
Excess	Grunt	Sock		Tinkly*	Harsh	
		Muddy		Shrill		
		Solid		Brassy	Hard	
		Dead, dull or thick		Metallic		
	Boom	Flat-sounding	Masculine	Bright		
		Body	Baritone	Brilliant		
		Mellow		Crisp	Brittle	Brittle
Deficiency		Lean	Warm		Soft	Soft
			Soprano			
		Thin				
		Tinny				

*If confined to upper part of region.

SECTION 2 : NON-LINEAR DISTORTION AND HARMONICS

(i) Non-linearity (ii) Harmonics (iii) Permissible harmonic distortion (iv) Total harmonic distortion (v) Weighted distortion factor (vi) The search for a true criterion of non-linearity.

(i) Non-linearity

A distortionless amplifier has an input voltage versus output voltage characteristic that is a straight line passing through the origin (*OA* in Fig. 14.1), the slope of the line indicating the constant voltage gain. When non-linearity occurs, as may happen with curvature of the valve characteristics, the input-output characteristic becomes curved as in *OB*, thereby indicating that the gain of the amplifier is a variable quantity.

(ii) Harmonics

One effect of this non-linearity is the production of harmonic frequencies in the output when a pure sine-wave input voltage is applied, hence the name harmonic distortion. For example, if the input voltage is a pure sine wave of frequency 100 c/s, the output may consist of a fundamental frequency 100 c/s, a second harmonic of 200 c/s, a third harmonic of 300 c/s, and so on. Only the fundamental frequency is present in the input, and the harmonics are products of the non-linearity (for mathematical treatment see Chapter 6 Sect. 8).

FIG. 14.1

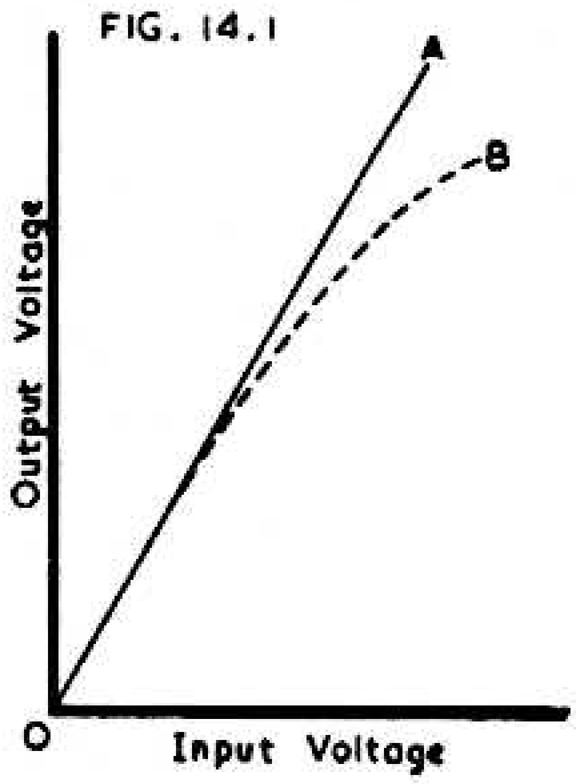

Fig. 14.1. Linearity characteristic (or transfer characteristic) — OA for distortionless amplifier ; OB for non-linear amplifier.

Harmonics themselves are not necessarily displeasing, since all musical instruments and voices produce complex sounds having many harmonics (or overtones). But all sounds have certain relationships between the fundamental and harmonic frequencies, and it is such relationships that give the sound its particular quality*. If certain harmonics are unduly stressed or suppressed in the reproduction, the character of the sound (i.e. the tone) will be changed. For example, it is possible for a displeasing voice to be reproduced, after passing through a suitable filter, so as to be more pleasing.

The critical ability of the human ear to distinguish harmonic distortion depends upon the frequency range being reproduced and also upon the volume. Thus, with wide frequency range, the limit of harmonic distortion which can be tolerated is noticeably lower than in the case of limited frequency range, for the same volume level.

If the amplifier does not amplify the harmonics to the same extent as the fundamental, the effective harmonic distortion will be changed. Bass boosting reduces the harmonic distortion of bass frequencies, while treble boosting increases the harmonic distortion of fundamental frequencies whose harmonics are in the frequency range affected by the boosting.

The shape of the audibility curves for the ear, which has maximum sensitivity at about 3000 c/s (Fig. 19.7), indicates that there is some effective boosting of harmonics compared with the fundamental for harmonics up to about 3000 c/s, and an opposite effect for fundamental frequencies above 3000 c/s. This effect, however, is quite small at frequencies up to 1500 c/s, at the usual maximum listening levels (say 65 to 80 db on a sound level meter). At lower listening levels the effect will be appreciable, but the distortion is normally fairly low at these levels in any case. With fundamental frequencies above 3000 c/s there is an appreciable reduction of harmonic distortion at all levels (this effect has been referred to by Ladner, Ref. E14). For example, with a fundamental frequency of 3000 c/s, at a loudness level of 80 phons, the second harmonic will be attenuated by 9 db and the third harmonic by 15 db.

Harmonic voltages or currents may be expressed in the form of a percentage of the fundamental voltage or current. For example, if the fundamental voltage is 100 volts, and there is a second harmonic voltage of 5 volts, the second harmonic percentage is 5%.

Some harmonics are dissonant with the fundamental and (unless at a very low level) are distinctly unpleasing to the listener as regards their direct effect, quite apart from

*These remarks apply to sounds which have slow attack and recovery times. In the case of most orchestral instruments this effect is largely overshadowed by the more prominent effects of transients. Many sounds (e.g. vowels) have an inharmonic content in addition to the harmonic content (Ref. E14).

the secondary effect on intermodulation. The following table gives the relevant details when the fundamental frequency (C) is taken for convenience as 250 c/s.

Harmonic	Harmonic frequency	Musical scale*	Effect
2nd	500	C^{i}	
3rd	750	G	
4th	1000	C^{ii}	
5th	1250	E	
6th	1500	G	
7th	1750	—	dissonant
8th	2000	C^{iii}	
9th	2250	D	dissonant
10th	2500	E	
11th	2750	—	dissonant
12th	3000	G	
13th	3250	—	dissonant
14th	3500	—	dissonant
15th	3750	B	dissonant
16th	4000	C^{iv}	
17th	4250	—	dissonant
18th	4500	D	dissonant
19th	4750	—	dissonant
20th	5000	E	
21st	5250	—	dissonant
22nd	5500	—	dissonant
23rd	5750	—	dissonant
24th	6000	G	
25th	6250	G #	dissonant

*Natural (just) scale. See Chapter 20 Sect. 8(iv).

(iii) Permissible harmonic distortion

The effect of non-linear distortion usually first becomes apparent to the listener through the production of inharmonic frequencies by intermodulation when two or more frequencies are present in the input (see Sect. 3). In practice, therefore, the percentages of the various harmonics which can be tolerated are fixed by their indirect effects rather than by their direct effect.

Under ideal theoretical conditions†, the magnitudes of the individual harmonics to produce equal intermodulation distortion are inversely proportional to the order of the harmonic. For example, 5% second harmonic produces the same intermodulation distortion as 3.3% third harmonic or 0.77% thirteenth harmonic—see below (v) Weighted distortion factor.

In order to make true comparisons between different sound systems, it is desirable to specify the frequency ranges, and the amplitudes of all the harmonics up to, say, the thirteenth. In the special case of single Class A triodes (below the overload point) the effect of the harmonics higher than the third may be neglected.

For a typical **single Class A triode** (type 2A3, Ref. E3) the harmonics, as derived from Olson, are :

Harmonic	2nd	3rd	4th	5th	
level	− 30	− 50	− 70	—	db
percentage	3.16	0.3	0.03	‡	%

With push-pull operation, the third harmonic becomes the dominant harmonic and, if the grid excitation is increased to give the same value of total harmonic distortion as with the single valve, the higher harmonics become more significant. With **Class AB_1 operation,** the higher harmonics are still further increased and, when weighted in proportion to the order of the harmonics, odd harmonics up to the thirteenth are prominent, and harmonics up to the twenty fifth may be appreciable.

†These conditions require the distortion to be similar to that produced by a Class A triode with not more than about 5% second harmonic distortion and without running into positive grid current.
‡Less than 0.01%.

With a **single pentode** the weighted values of the higher order harmonics are quite appreciable. The values of distortion—columns 1 and 2—are derived from Olson† for type 6F6 (Ref. E3).

Harmonic	level (db)	percentage	weighting factor‡	weighted distortion‡
2nd	−20	10	1	10%
3rd	−25	5.6	1.5	8.4%
4th	−35	1.8	2	3.6%
5th	−45	0.56	2.5	1.4%
6th	−62	0.08	3	0.24%
7th	−62	0.08	3.5	0.28%
8th	−63	0.07	4	0.28%
9th	−65	0.06	4.5	0.27%
10th	−73	0.022	5	0.11%
11th	−74	0.02	5.5	0.11%
12th	−73	0.022	6	0.13%
13th	−67.5	0.04	6.5	0.26%
14th	−74	0.02	7	0.14%

‡These are based on the ideal theoretical condition for equal intermodulation distortion ; they may differ considerably from the true effect on the listener.

With any system of amplification other than Class B, the percentage of total harmonic distortion, as defined in (iv) below, decreases as the power output level is reduced. Moreover, the percentages of the highest order harmonics decrease more rapidly than those of the lower order harmonics, as the power level is reduced.

The use of **negative feedback*** merely reduces all harmonics in the same proportion and does not affect their relative importance, except when the overload point is approached.

Comparative tests have been carried out by Olson (Ref. E3) which indicate

1. That slightly greater distortion is permissible with speech than with music.

2. That a higher value of total harmonic distortion is permissible with a single triode than with a single pentode, for the same effect on the listener.

3. That the permissible distortion decreases as the cut-off frequency is increased. However the rate of change varies considerably with the three categories of distortion, as given below.

4. **Perceptible distortion** (for definition see below).

Distortion becomes perceptible when the measured total harmonic distortion reaches the level of 0.7% for music, and 0.9% for speech, with a frequency range of 15 000 c/s and a pentode valve. Even with the very limited frequency range of 3750 c/s a total harmonic distortion of 1.1% on music, and 1.5% on speech, is perceptible with a pentode valve. The difference between pentode and triode is negligibly small.

5. **Tolerable distortion** (for definition see below).

The measured total harmonic distortion to give tolerable distortion on music, with either triode or pentode, increases about four times when the frequency range is reduced from 15 000 to 3750 c/s. The value of tolerable distortion with a pentode is 1.35% total harmonic distortion with a frequency range of 15 000 c/s, and 5.6% with a frequency range of 3750 c/s on music, and 1.9% and 8.8% respectively on speech. The amount of permissible total harmonic distortion with a triode is greater than that with a pentode for the same level of tolerable distortion, in the approximate ratio of 4 to 3.

The effect of frequency range on tolerable distortion is much greater than on perceptible distortion.

6. **Objectionable distortion** (for definition see below).

Objectionable distortion, on music, with either triode or pentode, increases about 5.5 times when the frequency range is reduced from 15 000 c/s to 3750 c/s ; the

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*It is assumed that the feedback network is not frequency-selective.

effect of frequency range on objectionable distortion with music is thus greater than its effect on tolerable distortion and much greater than on perceptible distortion. The values of total harmonic distortion to provide objectionable distortion are 2% with a frequency range of 15 000 c/s and 10.8% with a frequency range of 3750 c/s for music, and 3% and 12.8% respectively for speech, with a pentode. The value of total harmonic distortion with a triode to give the same level of objectionable distortion as a pentode is approximately in the ratio 5/4.

Definitions—Perceptible distortion is defined as the amount of distortion in the distorting amplifier which is just discernible when compared with a reference system of very low distortion. Tolerable distortion is the amount of distortion which could be allowed in low-grade commercial sound reproduction. Objectionable distortion is the amount of distortion which would be definitely unsatisfactory for the reproduction of sound in phonograph and radio reproduction. The two latter are dependent upon personal opinion.

Test conditions—Information derived from these tests has been tabulated below, the conditions being

Triode : single 2A3, power output 3 watts.
Pentode : single 6F6, power output 3 watts.

The sound was reproduced in a room with acoustics similar to a typical living room with a noise level of about 25 db. The tests were performed with a limited number of critical observers.

Test Results (Ref. E3)

Cut-off frequency		3750	5000	7500	10 000	15 000 c/s
		Objectionable distortion				
Music	Triode	14.0	8.8	4.8	3.4	2.5%
	Pentode	10.8	6.0	4.0	2.8	2.0%
Speech	Triode	14.4	10.8	6.8	5.6	4.4%
	Pentode	12.8	8.8	6.4	4.4	3.0%
		Tolerable distortion				
Music	Triode	6.8	5.6	4.4	3.4	1.8%
	Pentode	5.6	4.0	3.2	2.3	1.35%
Speech	Triode	8.8	7.2	4.8	3.6	2.8%
	Pentode	8.8	5.2	4.0	3.0	1.9%
		Perceptible distortion				
Music	Triode	1.2	—	0.95	—	0.75%
	Pentode	1.1	—	0.95	—	0.7%
Speech	Triode	1.4	—	1.15	—	0.9%
	Pentode	1.5	—	1.2	—	0.9%

(iv) Total harmonic distortion

The **distortion factor** of a voltage wave is the ratio of the total r.m.s. voltage of all harmonics to the total r.m.s. voltage.

The **percentage of total harmonic distortion** is the distortion factor multiplied by 100 :

$$D = \frac{\sqrt{E_2^2 + E_3^2 + E_4^2 + \dots}}{\sqrt{E_1^2 + E_2^2 + E_3^2 + E_4^2 + \dots}} \times 100 \quad (1)$$

where D = percentage of total harmonic distortion
E_1 = amplitude of fundamental voltage
and E_2 = amplitude of second harmonic voltage, etc.

If the distortion is small, the percentage of total harmonic distortion is given approximately by

$$D \approx \sqrt{(H_2\%)^2 + (H_3\%)^2 + \dots} \quad (2)$$

where $(H_2\%)$ = second harmonic percentage, etc.
with an error not exceeding 1% if D does not exceed 10%.

The measurement of total harmonic distortion is covered in Chapter 37 Sect. 3(ii).

(v) Weighted distortion factor

Some engineers use a weighted distortion factor in which the harmonics are weighted in proportion to their harmonic relationship :

$$W.D.F. = \tfrac{1}{2}\sqrt{\frac{(2E_2)^2 + (3E_3)^2 + \ldots}{E_1^2 + E_2^2 + E_3^2 + \ldots}} \qquad (3)$$

For example, if $E_1 = 100$, $E_2 = 5$, $E_3 = 8$ volts,
i.e. second harmonic $= 5\%$
and third harmonic $= 8\%$
then $W.D.F. = 0.129$ (or 12.9%)
and Total distortion factor $= 0.094$ (or 9.4%).

When all the distortion is second harmonic, this weighted distortion factor is the same as the total distortion factor.

With this weighted distortion factor it is desirable to measure harmonics up to a high order (say the thirteenth) at frequencies where the high order harmonics come within the frequency band of the amplifier.

(vi) The search for a true criterion of non-linearity

Tests carried out by Olson—see (iii) above—indicate that, under the test conditions, the value of total harmonic distortion is a fairly accurate indication of " perceptible distortion ". Total harmonic distortion may thus be regarded as a true criterion of the subjective effects of distortion in an amplifier, provided that the value is of the order of 1% or less, and provided that overloading or " kinks " in the linearity characteristic do not occur. It has yet to be proved whether or not total harmonic distortion of the order of 1% is a true measure of the subjective effects of distortion under all conditions.

It is generally admitted that the value of total harmonic distortion does not provide a true criterion of non-linearity between different types of amplifiers under all possible conditions, even though the measurements may be made over the whole audible frequency range, and even though the frequency ranges may be identical.

It is the writer's opinion that the conventional weighted distortion factor—see (v) above—also fails to provide a true criterion of non-linearity.

It has been found that when measuring intermodulation distortion, the r.m.s. sum method is less effective than the arithmetical peak sum method in comparing different types of amplifiers—see Sect. 3. Since the principle of intermodulation testing inherently gives harmonic weighting, an obvious step would be to adopt the arithmetical sum of the weighted harmonics, using the conventional weighting factor.

One interesting investigation by Shorter (Ref. A45) has indicated that a very much more drastic weighting, using the square of the conventional weighting factor and including all harmonics with at least 0.03% amplitude, showed distortion values in the correct subjective sequence. The results obtained by Shorter are summarized in the table on page 611.

One inference which may reasonably be drawn is that any sharp kinks in the linearity curve, as usually occur in any Class AB_1 or AB_2 amplifier, have a far more serious subjective effect than is indicated by any of the standard methods of measuring distortion—whether total harmonic distortion, conventional weighted distortion factor or the standard form of intermodulation testing.

The excellent work by Olson—as described in (iii) above—has provided the correlation between frequency range and harmonic distortion for either music or speech, triode or pentode, for three levels of subjective distortion. What still remains to be done is for a comprehensive investigation to be made into the equivalent subjective effects (preferably following the definitions laid down by Olson) of all types of amplifiers—both good and bad design—measuring the individual harmonics up to, say, the twenty-fifth. The tests need only be taken with a single frequency range, since the effect of frequency has been adequately covered by Olson.

The test on each amplifier should be made at maximum rated power output into a resistive load, and also, if desired, at a specified level below maximum rated power

Results obtained by Shorter

System	Subjective Classification of Distortion	R.M.S. Sum of Harmonics	Weighted* Distortion	More drastic† Weighted Distortion	Linearity Curve‡
A	Bad	2.3%	5.1%	19.4%	Pronounced kink
D	Bad	3.7%	6.7%	16.5%	Pronounced kink
B	Perceptible	3.3%	5.1%	8.6%	
E	Just perceptible	0.62%	1.3%	4.5%	Slight kink
C	Just perceptible	2.6%	2.8%	3.3%	
F	Not perceptible	0.41%	0.8%	2.2%	

*As (v) above.
†With square of weighting factor of (v) above.
‡Indicated on a C.R.O. (input versus output voltage).

output. Tests should also be made on the same amplifiers for intermodulation distortion, using both r.m.s. and peak sum methods, with suitable test frequencies.

In all cases, any distortion measurements should be supplemented by

(1) An oscillographic inspection of the waveform, and

(2) An oscillographic inspection of the linearity (input versus output) characteristics for sharp " kinks."

From these results it should be possible to derive a system of harmonic weighting such that the weighted harmonic distortion is a true criterion of the subjective effects of the distortion.

SECTION 3 : INTERMODULATION DISTORTION

(i) Introduction (ii) Modulation method of measurement—r.m.s. sum (iii) Difference frequency intermodulation method (iv) Individual side-band method (v) Modulation method of measurement—peak sum (vi) Le Bel's oscillographic method (vii) Comparison between different methods (viii) Synthetic bass.

(i) Introduction

Intermodulation distortion is one of the effects of non-linearity when more than one input frequency is applied. It is evident to the listener in two forms, amplitude-modulation of one frequency by another, and the production of sum and difference frequencies. See also Chapter 2 Sect. 9(iv).

An example of amplitude-modulation is the effect of a non-linear amplifier on the reproduction of a choir with a heavy organ bass accompaniment. The choir is amplitude-modulated by the organ—a displeasing effect to the listener. This effect is negligibly small in amplifiers if the total harmonic distortion is less than say 2%, but it is apparent with some loudspeakers (see Chapter 20 Sect. 7).

The formation of sum and difference frequencies is the second and more serious form of intermodulation distortion. These frequencies are normally in-harmonic and are the principal cause of the distorted reproduction noticed by any listener—sometimes described as harsh, buzz, rough or unpleasant.

In actual operation there are very many input frequencies applied simultaneously, but it is possible to make comparative tests using only two frequencies. Tests for intermodulation distortion may be made in accordance with any of the methods described below. For measurement see also Chapter 37 Sect. 3(ii)A.

The indicated value of intermodulation differs significantly with the method used, and the conditions of testing. It is important, in all cases, to specify both the method and the conditions of testing.

In all cases the two input voltages must be mixed, without intermodulation, before reaching the input terminals of the amplifier— bridge networks or hybrid coils are commonly used.

(ii) Modulation method of measurement—r.m.s. sum

In accordance with this method, one of the two input frequencies is preferably weaker than the other (a ratio of 4 : 1 in voltage or 12 db is used generally throughout this handbook unless otherwise indicated), and the stronger is usually lower in frequency than the other*. The sum and difference frequencies are usually expressed in the form of the percentage modulation of the weaker (high frequency) fundamental voltage. For one pair of sidebands the modulation percentage is given by

$2 \times$ sideband amplitude $\times$ 100/fundamental amplitude.

When testing, in order to make a fair comparison with single-frequency conditions, the input should be adjusted so as to give the same peak output voltage as under single frequency conditions. **With the 4 : 1 ratio of the two input voltages, the equivalent single-frequency power output is 25/17 or 1.47 times the indicated power output under I.M. conditions** (Ref. B14). The indicated power output is that calculated from the r.m.s. voltage across the load resistance.

Since there are many of these sum and difference frequencies, plus harmonics of both applied frequencies, it is necessary to take account of their magnitudes. This may be accomplished, in accordance with this method, in special equipment for the measurement of total intermodulation distortion by reading the r.m.s. sum of all extraneous frequencies (Refs. B5, B10). For measurement see Chapter 37 Sect. 3(ii).

There is no simple **relationship between harmonic and intermodulation distortion**. For example, it has been shown that in record manufacture the excessive polishing of masters greatly increases intermodulation distortion but does not much affect harmonic distortion (Ref. B6). See also Refs. A45 (correspondence), B7, B11, B19, B20.

If all harmonics are within the frequency range of the amplifier it may be shown that—

1. If only second harmonic is present (a condition which never occurs in practice), the ratio of total intermodulation distortion to second harmonic distortion is 3.2 (Refs. B7, B11).
2. If only third harmonic is present (another condition which never occurs in practice), the ratio of total intermodulation distortion to third harmonic distortion is approximately 3.84 at low values of distortion (Refs. B7, B11).
3. If the distortion is small, the intermodulation sidebands are approximately given by (Ref. B8) :

$$\left.\begin{array}{l}\text{Modulation percentage of first}\\ \quad\text{intermodulation sideband}\end{array}\right\} \approx 2 \times \text{ second harmonic distortion percentage.}$$

$$\left.\begin{array}{l}\text{Modulation percentage of second}\\ \quad\text{intermodulation sideband}\end{array}\right\} \approx 3 \times \text{ third harmonic distortion percentage.}$$

Thus intermodulation distortion is automatically weighted by the order of the distortion.

As a very rough approximation, the ratio I.M./H.D. may be taken as :

3.2 for single-ended triodes

and 3.8 for push-pull (triodes or pentodes)

where I.M. = total intermodulation distortion (r.m.s. sum)

and H.D. = total harmonic distortion,

*With a-f systems with peaked response in the middle-to-high frequency range, such as high efficiency speech systems and hearing aids, better results are obtained by having the higher input frequency stronger than the lower frequency (Ref. B13). The two methods of measurement will, of course, give different results.

provided that the operation is restricted to the normal low-distortion region (Refs B7, B11). The ratio tends to increase as the distortion increases. These ratios may increase suddenly as the amplifier reaches the overload point. They are also affected if a lower I.M. frequency is selected which gives appreciable attenuation. The ratio is increased if there is any second harmonic cancellation in successive stages in the amplifier or any attenuation of harmonics. In practice therefore, with many disturbing factors, the ratio may vary from less than 1 to more than 6—see summary Ref. B19.

The usual test frequencies are 40, 60, 100, 150 or 400 c/s and 1000, 2000, 4000, 7000 or 12 000 c/s. It is helpful to make tests at two low frequencies—one of these (40 or 60 c/s) should approximate to the low frequency limit of the amplifier ; the other may be 100 or 150 c/s. The distortion at the lower of these two frequencies is largely influenced by any iron-cored transformers, while that at the higher of these two frequencies gives a more normal overall value. The upper frequency may approximate to half the upper frequency limit of the amplifier—this does not, however, give a stringent test of the distortion at high frequencies.

Intermodulation distortion may be visually observed with a C.R.O. (Refs. B9, B19) —see also Sect. 3(vi) below.

Reliable intermodulation measurements may be made in the presence of considerable noise since the latter is excluded by filters to a greater degree than with harmonic distortion. It may also be successfully applied in the case of restricted frequency range where the harmonics would be outside the range of the amplifier. In such a case it is important to remember that the distortion is not zero merely because the harmonics are not reproduced.

Permissible intermodulation distortion

The following is an arbitrary grouping which may be useful (Modulation method—r.m.s. sum) :

Extremely high fidelity a-f amplifier (40 c/s)	I.M. less than 2%
High fidelity a-f amplifier (40 c/s)	I.M. less than 4%
Good fidelity a-f amplifier (60 c/s)	I.M. less than 8%
Fairly good fidelity a-f amplifier (60 c/s)	I.M. less than 20%
Typical radio receiver—a-f amplifier only (150 c/s)	I.M. less than 40%

The values quoted are only a rough guide, since so many factors are involved. They apply to the equivalent single frequency maximum power output, and the lower test frequency is given in brackets.

See also comments in (vii) below.

Tests with 1 : 1 voltage ratio

More stringent tests at the higher frequencies may be made by the use of a 1 : 1 voltage ratio.

(iii) Difference-frequency intermodulation method

In accordance with this method, two frequencies f_1 and f_2, of equal amplitude, are applied to the input and the relative amplitude of the intermodulation component at the difference frequency ($f_2 - f_1$) is then considered a measure of the inter-modulation distortion, which may be expressed as a percentage of either output voltage. A wave analyser may be used to measure the difference frequency. Typical lower frequencies are 1000, 5000 and 9000 c/s, while difference frequencies range from 50 to 500 c/s (Ref. A51) or even higher (Ref. B22). For the precise interpretation of results, a fairly low difference frequency is desirable.

Equipment has been designed so that the frequencies of the two input voltages can be varied over the a-f range with a constant difference frequency maintained between them (Refs. B3, B22).

See also comments in (vii) below.

References A51, B3, B12, B13, B22.

(iv) Individual sideband method

In accordance with this method the individual sidebands are measured separately, and some attempt made to select the most important ones (Ref. B1).

If all significant sidebands are measured, the distortion may be recorded in accordance with the peak sum modulation method—see (v) below.

(v) Modulation method of measurement—peak sum

This newer method of defining and measuring intermodulation distortion (Ref. B16) has outstanding advantages over the older method based on the r.m.s. sum. This peak sum method will measure the arithmetical sum of the amplitudes of the modulation products involved, with no discrimination against the weaker modulation products.

In accordance with this method, the percentage intermodulation is defined as the arithmetical sum of the amplitudes of the " in phase " modulation products divided by the amplitude of the high frequency carrier,

$$\text{i.e. percentage intermodulation} = \frac{A_1 + A_{-1} + A_2 + A_{-2} + \dots}{A} \times 100$$

Where A_1 and A_{-1} are the peak amplitudes of the modulation products of frequencies $\omega_2 \pm \omega_1$,

A_2 and A_{-2} are the peak amplitudes of the modulation products of frequencies $\omega_2 \pm 2\omega_1$, etc.,

A is the peak amplitude of the high frequency carrier,

ω_1 is the angular velocity of the low frequency input,

ω_2 is the angular velocity of the high frequency input.

The input amplitude of ω_2 is 12 db below that of ω_1.

In order to measure the sum of these voltages accurately, it is necessary to use a peak-reading voltmeter. Ref. B16 describes such an analyser primarily for use with frequencies of 400 and 4000 c/s for testing pickups, but also capable of use with input frequencies less than 400 or greater than 4000 c/s.

See also comments in (vii) below.

(vi) Le Bel's oscillographic method

The use of an oscilloscope to give a qualitative indication of intermodulation distortion is well known—a good description is given in Ref. B9. A quantitative method has subsequently been developed by Le Bel (Ref. B19). The two voltages of different frequencies are applied to the input of the amplifier in the usual way, the output is passed through a high-pass filter to an oscilloscope with the sweep adjusted to cover one cycle of the low frequency. If there is no intermodulation, the high frequency wave has constant peak amplitude and the envelope is rectangular as in Fig. 14.1A.

FIG. 14.1A FIG. 14.1B

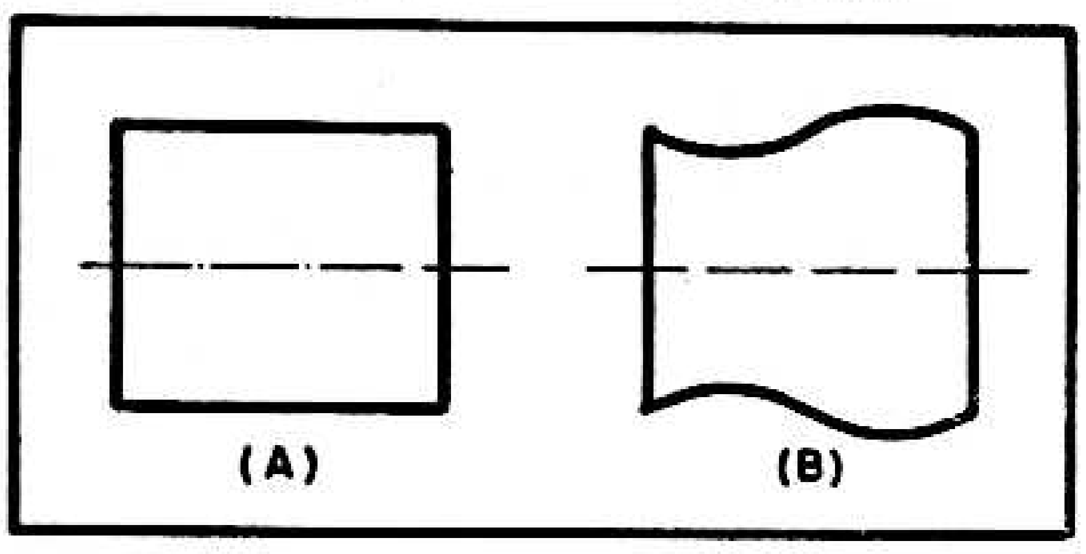


FIG. 14.1C

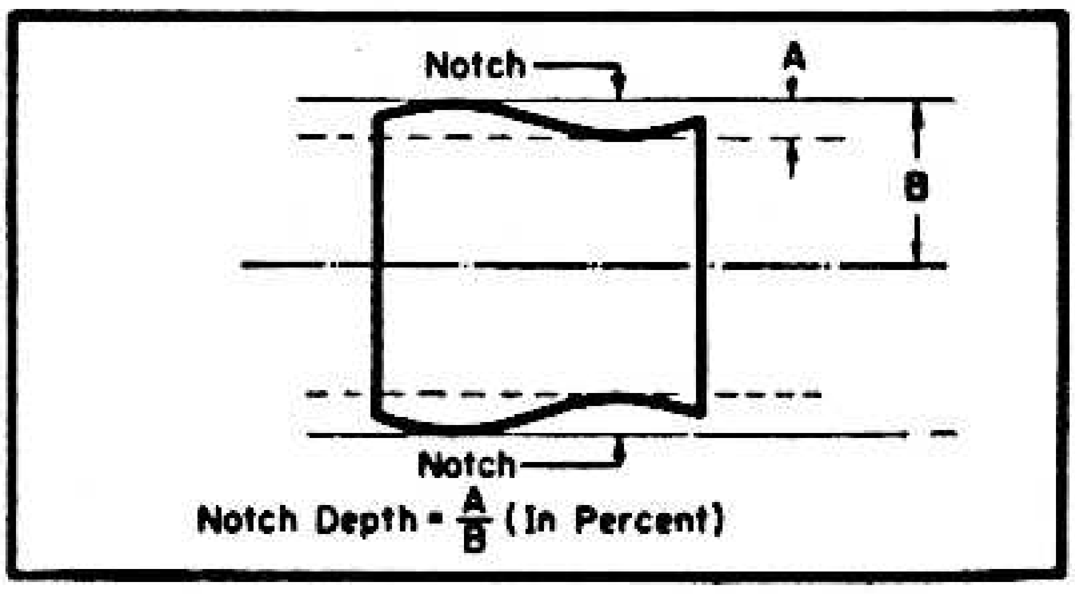


Fig. 14.1A & B. *Envelope of oscilloscope images without intermodulation (A), and with intermodulation (B). Ref. B*19.
Fig. 14.1C. *Definition of notch depth (Ref. B*19).

Intermodulation results in an envelope such as Fig. 14.1B. Normally, the intermodulation causes a " notch " or notches in the envelope and the notch depth is defined by Fig. 14.1C. The experimental relationship which has been determined between notch depth and intermodulation (using the normal modulation method of measurement—r.m.s. sum—as in Ref. B2) is shown in Fig. 14.1D. This curve is practically linear below 50% notch depth (i.e. 10% intermodulation), so that a scale may be used directly on the screen for measuring the value corresponding to each notch. If there is more than one notch, Le Bel defines total notch depth as the arithmetical

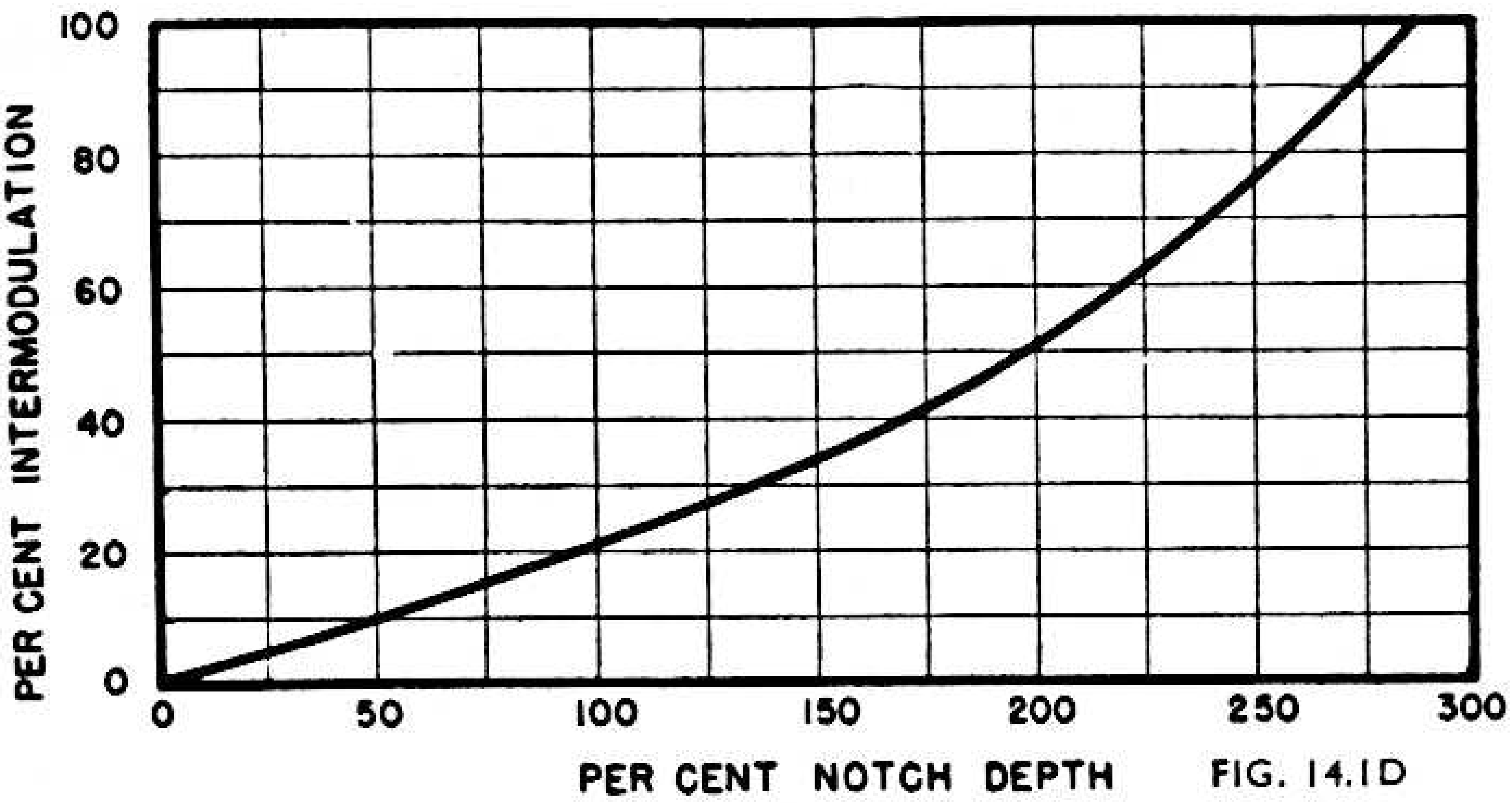


Fig. 14.1D. *Relation between notch depth and per cent. intermodulation (Ref. B19).*

FIG. 14.1E

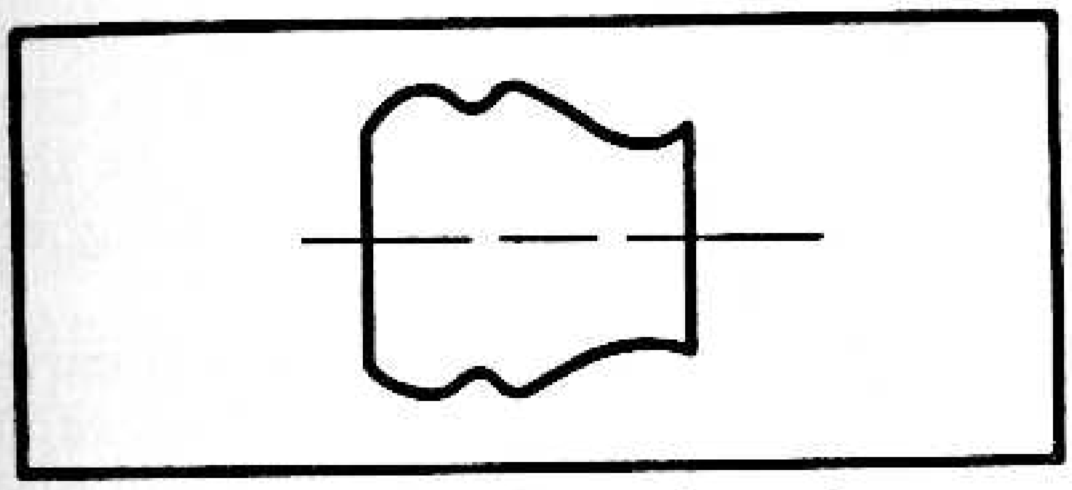


Fig. 14.1E. *Envelope of notch pattern with insufficient bias, for single-ended stage (Ref. B19).*

FIG. 14.1F

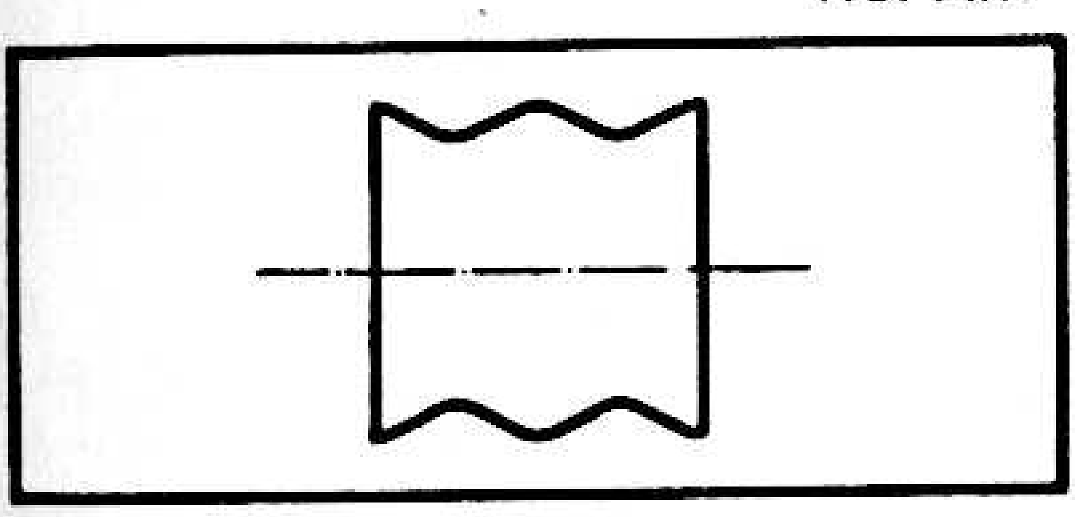


Fig. 14.1F. *Normal notch pattern for push-pull stage (Ref. B19).*

FIG. 14.1G

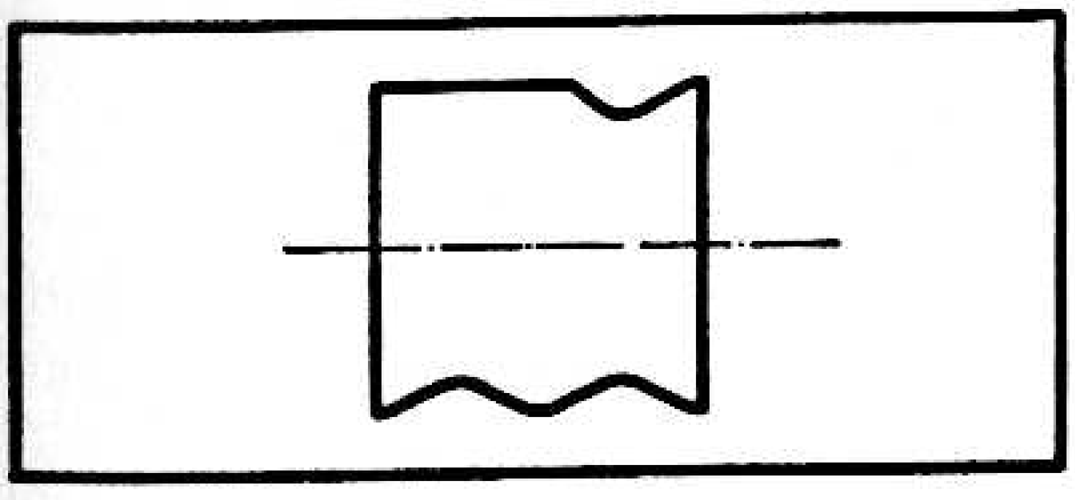


Fig. 14.1G. *Push-pull output stage with single-ended driver stage showing effect of driver overload (Ref. B19).*

FIG. 14.1H

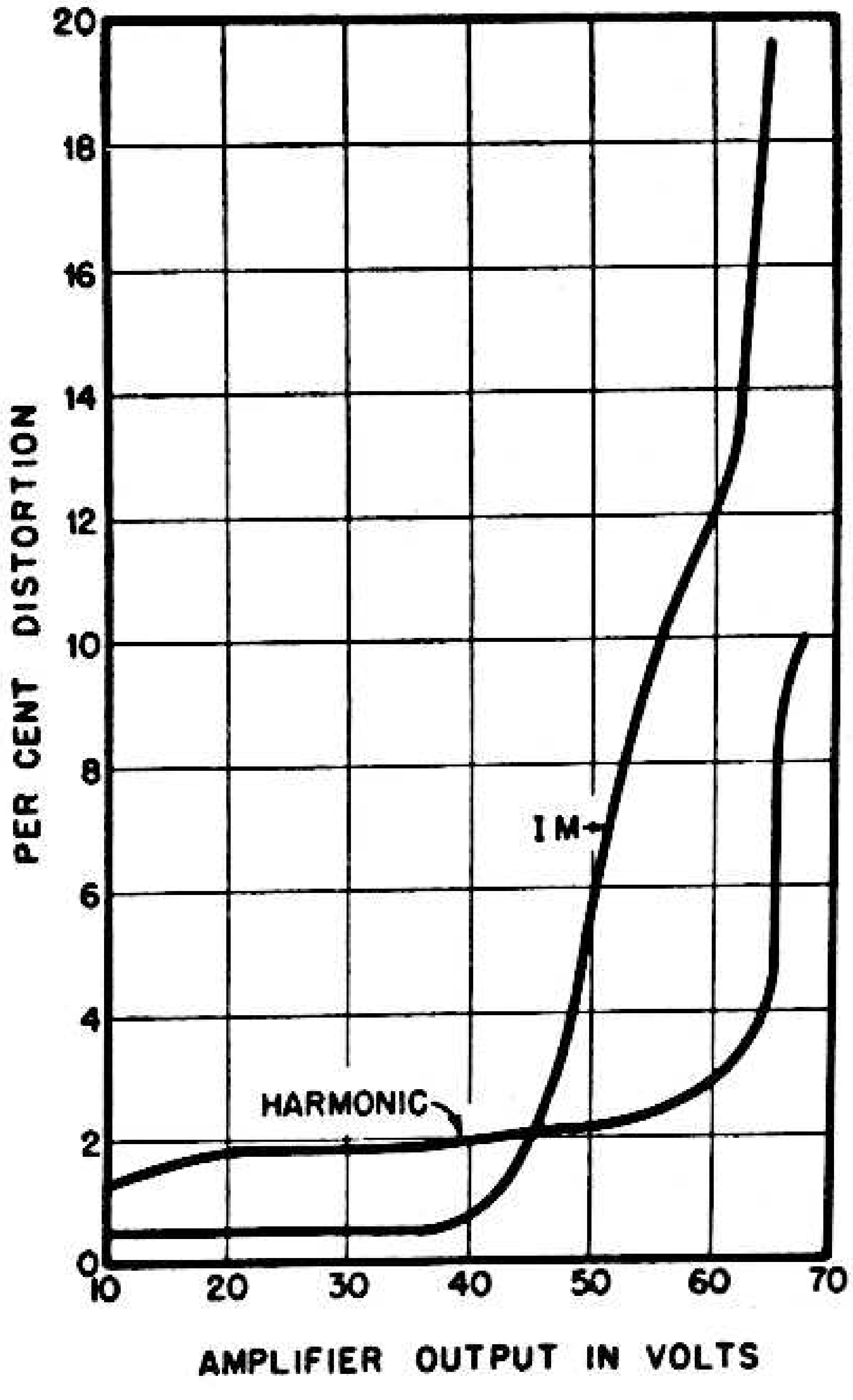


Fig. 14.1H. *Intermodulation (IM) and harmonic distortion characteristics of a push-pull amplifier showing that the ratio of the two parameters changes (Ref. B19).*

sum of the individual notch depths. Each notch in the low frequency cycle on top and/or bottom of the envelope is counted. Some typical envelope patterns are shown in Figs. 14.1E, F and G. If a peak instead of a notch occurs at any point, this indicates regeneration.

The harmonic and intermodulation* distortion characteristics of a push-pull amplifier are shown in Fig. 14.1H. At low output levels the intermodulation is less than the harmonic distortion, while at high output levels the reverse is true. This

*In accordance with Le Bel's oscillographic method.

indicates that the method is a much more sensitive indication of the approach towards the overload point than is total harmonic distortion.

See also comments in (vii) below.

(vii) Comparison between different methods

In recent years the modulation method of measurement based on the r.m.s. sum has been widely used. However this method has a serious shortcoming in that the lower amplitude modulation products have a negligibly small effect on the reading, which is almost entirely controlled by a few modulation products of high amplitude. It has been clearly demonstrated that this method does not give an indication proportional to the subjective effect on the listener, particularly when discontinuities (" kinks ") occur in the valve characteristics as with Class AB or Class B operation, or when running into grid current with a high impedance source.

This defect is at least partially overcome by the use of the peak reading analyser described in (v) above, which is distinctly preferable as giving an indication more in line with the results from listening tests.

It seems that Le Bel's method described in (vi) above is an oscillographic equivalent of the peak sum method, and close correlation would be expected between them.

It remains to be proved whether or not the peak sum method gives readings proportional to the subjective effect on the listener under all conditions. It may be that some " weighting " method is required to give even greater prominence to the higher-order products that arise from sharp kinks in the linearity characteristic.

All modulation methods with the lower frequency at the higher level are primarily tests on low frequency distortion and, as such, fulfil a useful purpose. It is essential, for design purposes, to have separate measurements of distortion both at low and high frequencies. Insufficient investigation has been made into the general use of modulation methods with the higher frequency at the higher level, for testing high frequency distortion (however see Refs. B13, B19).

The difference-frequency method is very suitable for testing distortion at middle frequencies, and also at high frequencies where conventional* modulation methods are deficient. Unfortunately there are no published data to enable a comparison to be made with the peak sum modulation method.

The distortion indicated by the several methods may differ appreciably. For example, consider high-frequency pre-emphasis as used in recording and F-M. If distortion takes place after pre-emphasis and before de-emphasis, and if the distortion is measured after de-emphasis, the percentage harmonic distortion is lowered by the de-emphasis ; the modulation method of measuring intermodulation is not significantly altered by de-emphasis ; while the difference-frequency method gives a higher value as the result of de-emphasis (Ref. B13).

See also Ref. A43 and Chapter 37 Sect. 3(ii)A.

(viii) Synthetic bass

(A) When two or more input frequencies are applied to a non-linear amplifier the output will include sum and difference frequencies located about each of the higher input frequencies. For example with input frequencies of 50 and 150 c/s, the output will include frequencies of 50, 100, 150 and 200 c/s. Even if the lowest frequency is very much attenuated by the amplifier, the sum and difference frequencies tend to create the acoustical impression of bass. With more than two input frequencies the effect is even greater, so that fairly high distortion has the effect of apparently accentuating the bass.

(B) Owing to the peculiar properties of the ear, a single tone with harmonics may be amplified, the fundamental frequency may be completely suppressed, and yet the listener hears the missing fundamental (Ref. A18).

These two effects assist in producing " synthetic bass "—Chapter 15 Sect. 12(ii)—when the natural bass is weak or entirely lacking. It should be emphasized that this is not the same as true bass, and does not constitute fidelity.

*With the lower frequency having the greater amplitude.

SECTION 4 : FREQUENCY DISTORTION

(i) Frequency range (ii) Tonal balance (iii) Minimum audible change in frequency range (iv) Sharp peaks.

(i) Frequency range

Frequency distortion in an amplifier is the variation of amplification with the frequency of the input signal. A high fidelity a-f amplifier should have nearly constant amplification over the whole range audible to the most critical listener—from say 30 to 20 000 c/s (see Sect. 7). There is no serious difficulty in designing such an amplifier, but it can only be usefully applied when it is fed from a wide-range source and excites a wide-range loudspeaker system.

In practice there are limitations imposed by both the source and the loudspeaker, but it is good practice to design the amplifier for full-power-handling capacity with negligible non-linear distortion over a wider frequency range than it will normally be required to handle. The limitation in frequency range (if not inherent in the source) should preferably be applied either by a filter between the source and the input terminals, or in an early stage in the amplifier—see Chapter 15 Sect. 1(iii).

The full frequency range is only audible at very high levels as is demonstrated in Sect. 7. If a particular equipment is to be operated always at comparatively low levels (e.g. dinner music) or in noisy locations, reduction of the frequency range is quite correct and may even be beneficial.

Amplifiers for other than wide-range high-fidelity are usually designed to meet the requirements of the average listener, with a frequency range (at maximum orchestral level) of about 45 to 15 000 c/s, which may be still further reduced to about 70-13 000 c/s at typical levels for home listening—see Sect. 7(ii).

In commercial quantity-produced equipment the frequency range is often restricted to the extreme. By this means non-linear distortion and hum are reduced to a bearable level without expense. Unfortunately, frequency range can only be extended, while remaining free from obvious distortion, at a cost which rises at a rapidly increasing rate—largely due to the loudspeaker. It is important to reduce the distortion before widening the frequency range—see Sect. 12(ii).

(ii) Tonal balance

It has been found that good **tonal balance** between high and low frequencies is obtained when the product of the limiting frequencies is about 500 000* (Refs. A8, A17, A38, C1, C10).

For example :

1. 25-20 000 c/s. Very wide frequency range,
2. 33-15 000 c/s. Wide frequency range,
3. 50-10 000 c/s. Fairly wide frequency range,
4. 60- 8 000 c/s. Medium frequency range,
5. 100- 5 000 c/s. Restricted frequency range,
6. 150- 3 300 c/s. Very restricted frequency range,

but these should not be taken as more than a general guide. (5) is typical of a medium quality console, and (6) a mantel model receiver with " mellow " tone. These frequency ranges should include both amplifier and loudspeaker. Wide frequency range is only comfortable to the listener so long as other forms of distortion are imperceptible.

(iii) Minimum audible change in frequency range

Just as with sound level there is a minimum audible change in level, so there is also a minimum audible change in frequency range. The limen has been proposed (Ref. A25) as being the minimum difference in band width that is detectable by half the observers. Tests were made on a limited number of observers resulting in the following list of frequencies in steps of 1 limen : 15 000, 11 060, 8000, 6400, 5300,

*Various authorities place this figure from 400 000 to 640 000.

4400 c/s. It would therefore be possible to reduce the frequency range of an amplifier from 15 000 c/s (say the limit of hearing in a particular case) to 11 000 c/s without a non-critical listener noticing the restriction. The same principle also holds in reverse, when widening the frequency range. (See also Ref. A9).

(iv) Sharp peaks

One particularly objectionable form of frequency distortion is that due to sharp peaks in the output, as may be caused by loudspeaker cone resonances, especially in the 2000-3500 c/s range. Sharp troughs are relatively unimportant through their direct effects, although they may be accompanied by poor transient response.

The effect of restricted frequency range on articulation is covered in Sect. 11.

SECTION 5 : PHASE DISTORTION

Phase distortion is the alteration by the amplifier of the phase angle between the fundamental and any one of its harmonics or between any two component frequencies of a complex wave. Phase distortion causes the output waveform to differ from the waveform of the input voltage.

A fixed phase shift of 180° (or any multiple of 180°) at all frequencies does not constitute phase distortion. A phase shift which is proportional to the frequency also does not cause phase distortion.

It has been demonstrated in monaural listening (Ref. A23) that the tonal quality of a complex steady-state waveform depends not only upon the component frequencies and their relative amplitudes, but also—over certain ranges of frequency and level—upon the envelope shape, which is controlled by the phase angles. Large changes in aural perception can occur, through changes in phase only, among frequencies which are related harmonically or otherwise. These effects vary from a raucous to a smooth-sounding quality, depending on the relative phases. The frequency range, over which these effects were observed, increases as the acoustical level increases, at least up to 60 phons.

The authors state that " in the case of very complex, but continually varying sounds, such as voice or a musical instrument with many harmonics, phase effects of this type are probably not noticeable under usual listening conditions because the patterns are coming in and out non-simultaneously in various parts of the frequency range."

Phase shift has a serious effect on transients (see Sect. 6).

In the light of the known effects, and with insufficient data to determine what degree of phase shift is permissible for high fidelity reproduction, it is wise to reduce the phase shift in amplifiers to the lowest practicable value.

With normal circuits, the phase shift is determined by the overall attenuation at the extreme low and high frequencies of the sound system. If the frequency characteristic is practically flat over the whole useful frequency range, the phase shift may normally be neglected. However, when a high-pass or low-pass filter is used to limit the frequency range, or when some form of tone control is used, the phase angle may be quite large over a limited frequency range.

Refs. A15, A23, A28, A40, A42, A57, E11.

SECTION 6 : TRANSIENT DISTORTION

(*i*) *General survey* (*ii*) *Testing for transient response.*

(i) General survey

It has been demonstrated by Prof. Richardson that it is the attack and decay times of sounds that largely determine their tonal quality, rather than their harmonic content (Ref. A41).

An amplifier which gives fairly good reproduction of steady tones may give serious distortion with transients. The distortionless reproduction of a short pulse requires

1. Very wide frequency response—possibly higher than the limit of audibility.
2. No phase distortion.
3. No " hang-over." The duration should not be greater than the original pulse.

" Hang-over " effects are caused by insufficiently damped *LC* circuits such as may occur with a-f transformers, tone correction circuits and filters—see Chapter 15 Sect. 1(ix). Loudspeakers are particularly prone to this effect, and it is desirable for them to have heavy acoustical and electro-magnetic damping, with reduction of mass of moving parts.

Some measurements of the phase shift introduced by microphones and loudspeakers are given in Ref. A40.

References : A40, A41, E1.

(ii) Testing for transient response

Amplifiers and loudspeakers may be tested by applying to the input terminals a waveform with a sharp discontinuity—e.g. square wave or saw-tooth.

One of the most serious problems with certain kinds of music is the occurrence of extremely high transient peaks which may possibly reach a level 20 db above the reading of the volume indicator. This effect has been observed with choral music, orchestral string passages and similar highly complex sounds (Ref. E15—see also A3). These peaks cause overloading and distortion in the amplifier and loudspeaker, but particularly in recording. Some form of peak limiting is usually adopted in recording and broadcasting in order to avoid blasting or " buzz." However, for true fidelity, these transient peaks should be reproduced in their original form.

One interesting approach is the use of **white-noise** as a test signal. White-noise is random fluctuation noise, but the construction of a white-noise signal generator whose output power is uniformly distributed over the a-f region of the spectrum is difficult. One white-noise generator, which heterodynes a selected portion of the r-f noise spectrum to produce nearly uniform voltage over the a-f range, is described in Ref. D18. A simpler method is described in Ref. D17 which employs a disc recording of white-noise (40 to 20 000 c/s) together with bands having high frequency cut-off at 7000, 9000 and 12 000 c/s and low frequency cut-off at 80 and 150 c/s. This is recorded on a constant velocity basis. A flat noise voltage characteristic may be obtained by playing back with a velocity pickup of the dynamic or magnetic type, without any equalizing network. The recording is corrected for translation loss—4 db at 20 000 c/s. The following comments are based on Ref. D17.

Loudspeakers may be tested for undamped resonances by applying white-noise signal, picking up the sound by a suitable microphone and applying the output to the vertical plates of an oscilloscope. The sweep frequency is then varied to pick up the resonances. Cross modulation may be recognised by a periodic thinning out of the fine-grain high frequency noise. It is stated that a miniature condenser microphone is particularly suitable for this test.

When an amplifier is overloaded, the peaks on the oscilloscope appear beaded.

The elliptical pattern, obtained by applying the white-noise to both pairs of plates with approximately 90° phase difference, may be used as a sensitive indication of overloading. The circuit is arranged to give larger ellipses with higher frequencies.

SECTION 7: DYNAMIC RANGE AND ITS LIMITATIONS

(i) Volume range and hearing (ii) Effect of volume level on frequency range (iii) Acoustical power and preferred listening levels (iv) Volume range in musical reproduction (v) The effect of noise.

(i) Volume range and hearing

Every source of sound, for example speech or music, has a variation in sound level from its minimum to its maximum. The loudness level is measured in phons (see Chapter 19 Sect. 5) while the maximum variation in level may be expressed in decibels —this is known as the **volume range,** and is measured by a standard sound-level meter or volume indicator. As this instrument does not indicate short sharp peaks, the **peak dynamic range** is usually 10 or occasionally up to 20 db greater than that indicated by the instrument.*

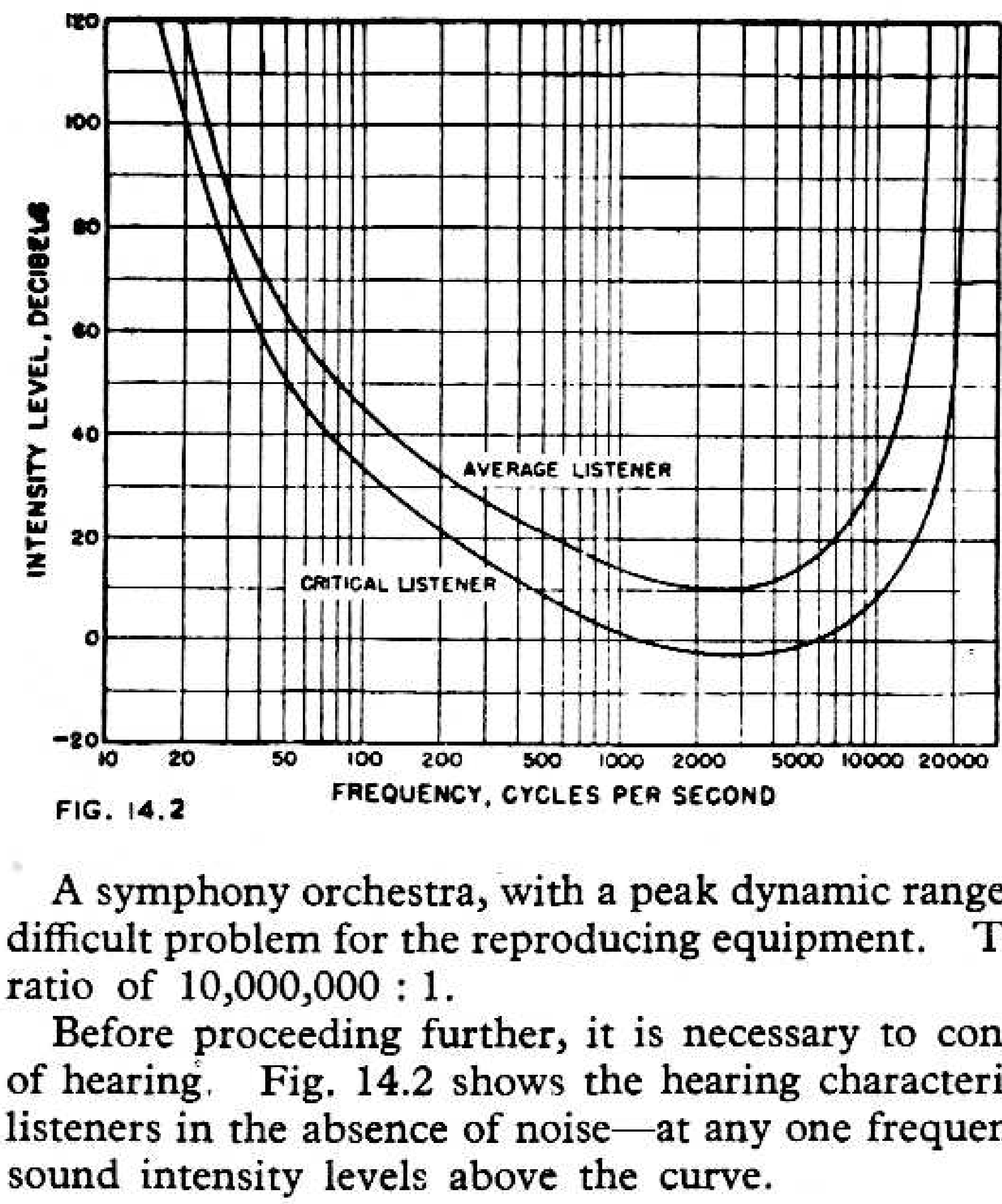


FIG. 14.2

Fig. 14.2. Threshold of audibility curves for average and very critical (5% most acute) listeners in the absence of noise. Reference level 0 db = 0.0002 dyne per sq. cm. Curves by courtesy of Jensen Radio Manufacturing Company, based on Fletcher (Ref. A3).

A symphony orchestra, with a peak dynamic range of up to 70 db provides the most difficult problem for the reproducing equipment. This range is equivalent to a power ratio of 10,000,000 : 1.

Before proceeding further, it is necessary to consider some of the characteristics of hearing. Fig. 14.2 shows the hearing characteristics of average and very critical listeners in the absence of noise—at any one frequency, a listener is only able to hear sound intensity levels above the curve.

Fig. 14.3. Masking levels for noise in average and very quiet residences. Curves by courtesy of Jensen Radio Manufacturing Company, based on Refs. A3, D6, D7, E9.

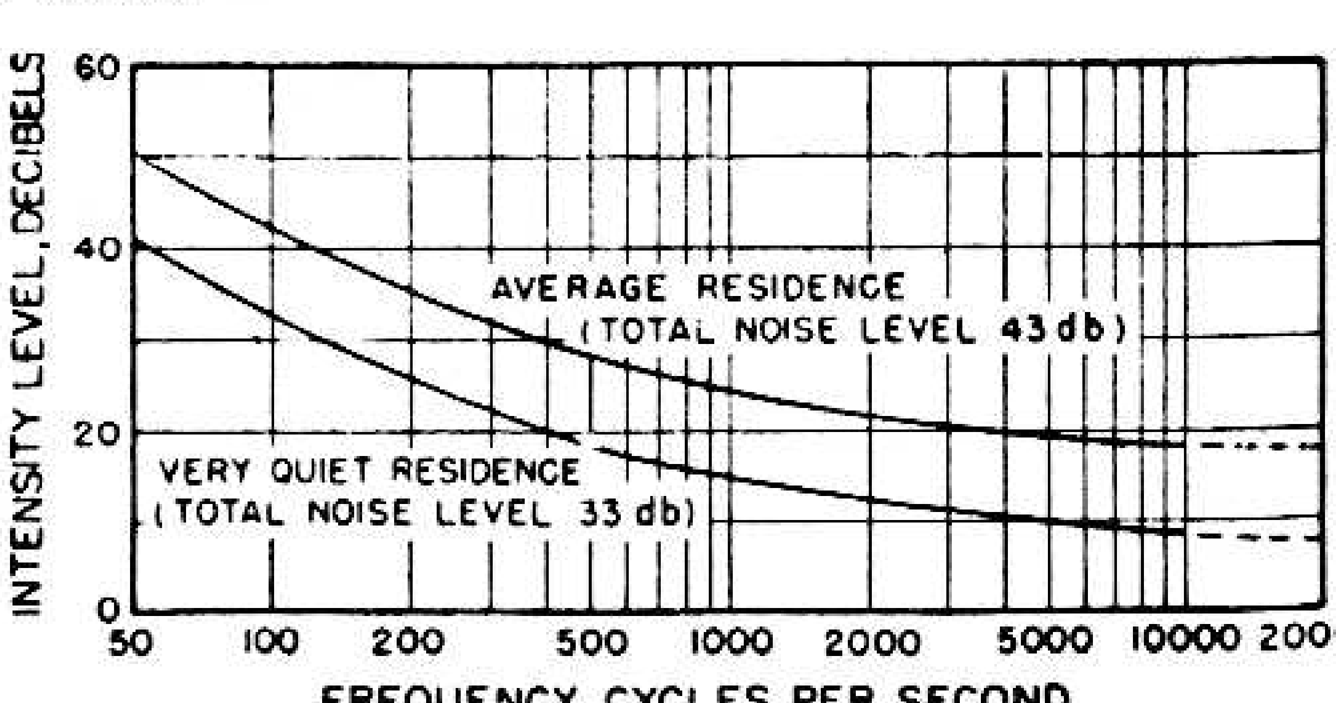


FIG. 14.3

The masking† effect of room noise is shown by Fig. 14.3 for average and for very quiet residences. These curves may be applied directly to the hearing characteristics of Fig. 14.2. The result is shown in Fig. 14.4 for average conditions and in Fig. 14.5

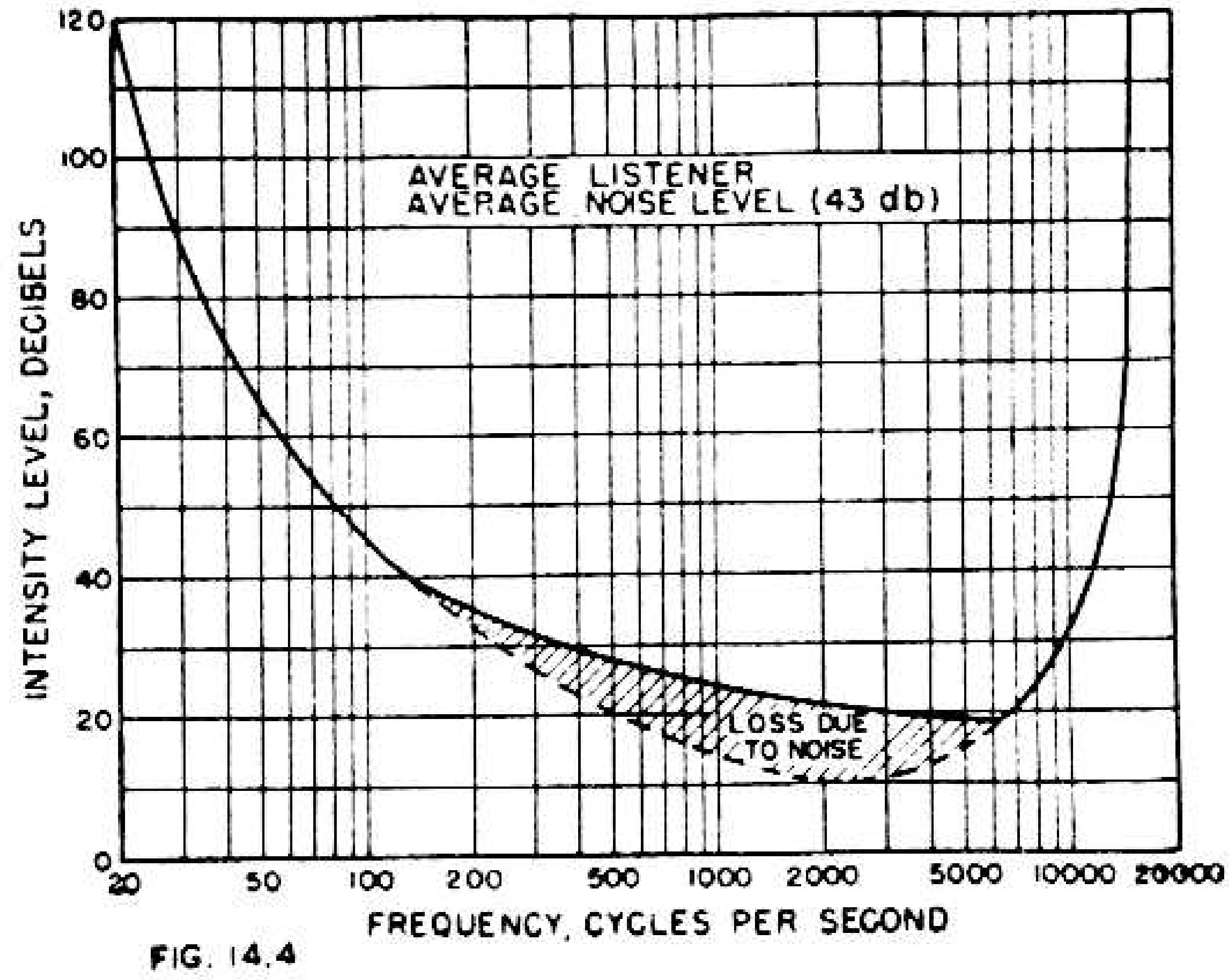
*The 10 db ratio is given in Ref. A51, quoting Refs. E7, E16. The 20 db ratio is mentioned in Sect. 6(ii), based on Refs. E15 and A3.

†The masking level is the level of pure tones which can just be perceived in the presence of noise. The masking level is higher than the noise (spectrum) level at the same frequency by a margin of 15db up to 1000 c/s, increasing to 28 db at 10 000 c/s (Ref. A53).

for a very critical listener and low room noise level (these latter are the extreme conditions for high fidelity). In each case the room noise reduces the effective hearing over a frequency range from about 150 to 6000 or 9000 c/s.

The effect of noises other than room noise is covered in (v) below.

Fig. 14.4. *Effective hearing characteristic for average listener with average noise level. Curves by courtesy of Jensen Manufacturing Company.*



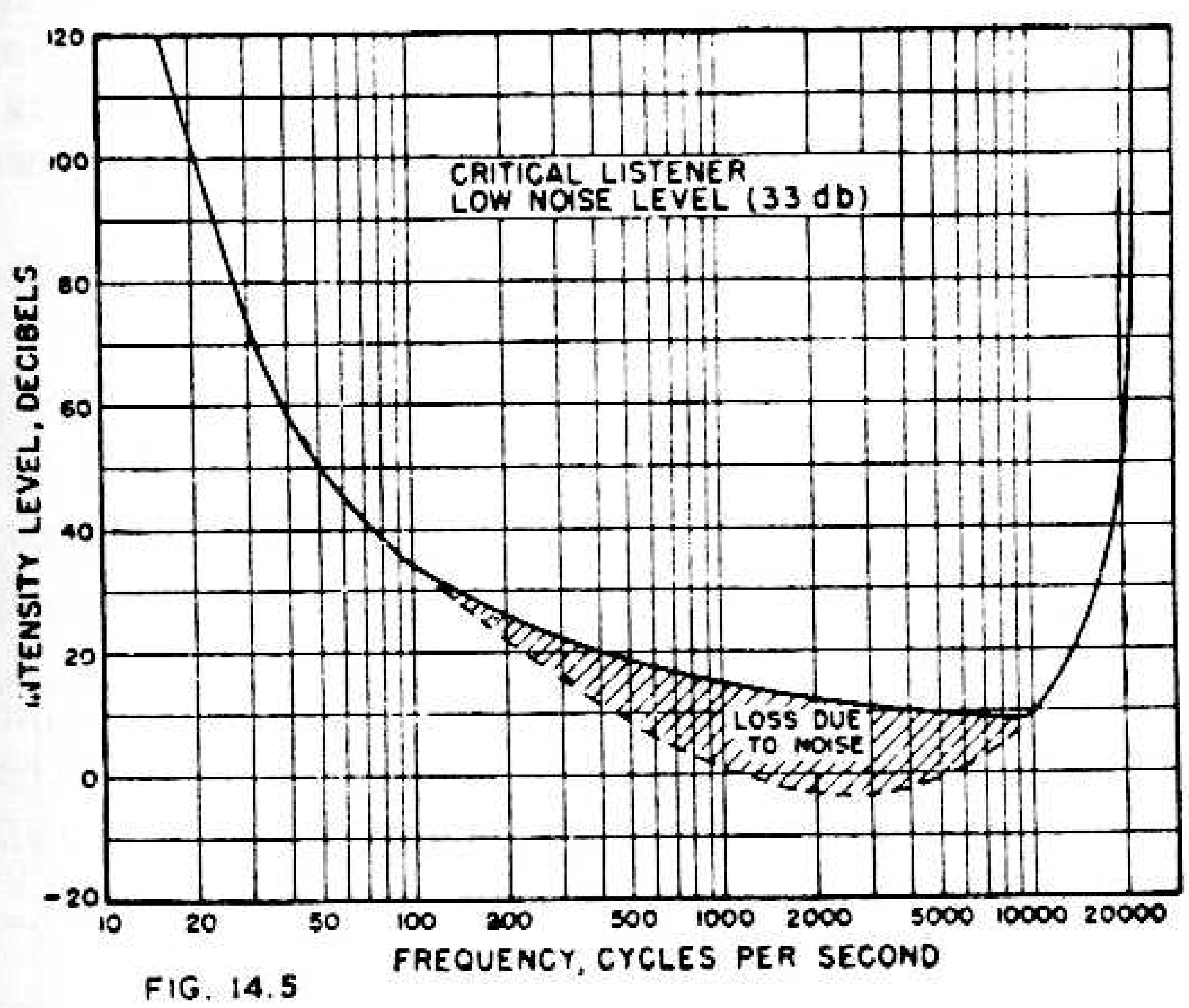


Fig. 14.5. *Effective hearing characteristic for very critical listener with low room noise level. Curves by courtesy of Jensen Manufacturing Company.*

(ii) Frequency range

When listening to the rapidly varying intensity levels which occur with speech or music, the ear appears to integrate these varying sounds over about 1/4 second intervals. When considering frequency range, the measurements of the sound levels created by the various types of musical instruments should therefore be reduced to their **equivalent sound levels in 1/4 second intervals.**

If we compare the loudness of a narrow band of thermal noise with that of a pure tone having the same intensity, the two will be judged to have equal loudness if the width of the transmitted frequency band of noise is limited to a critical value called the **critical bandwidth** (Ref. A3). For this reason, measurements on the sound levels of musical instruments should be reduced to intensity levels which would have been obtained if the frequency bandwidths used in the filters had been equal to the critical bandwidths (Ref. A3).

The integrated sound energy in a critical band over a 1/4 second interval will sound as loud as a pure tone in the same frequency band which produces the same sound energy in each 1/4 second interval (Ref. A3).

Fig. 14.6 gives the maximum r.m.s. intensity levels in 1/4 second intervals in critical frequency bands for certain musical instruments and an orchestra at a distance of 20 feet from the sound source, together with the threshold of audibility curve for average listeners, indicating that a frequency range from 40 to 15 000 c/s meets all normal requirements for an average listener to an orchestra.*

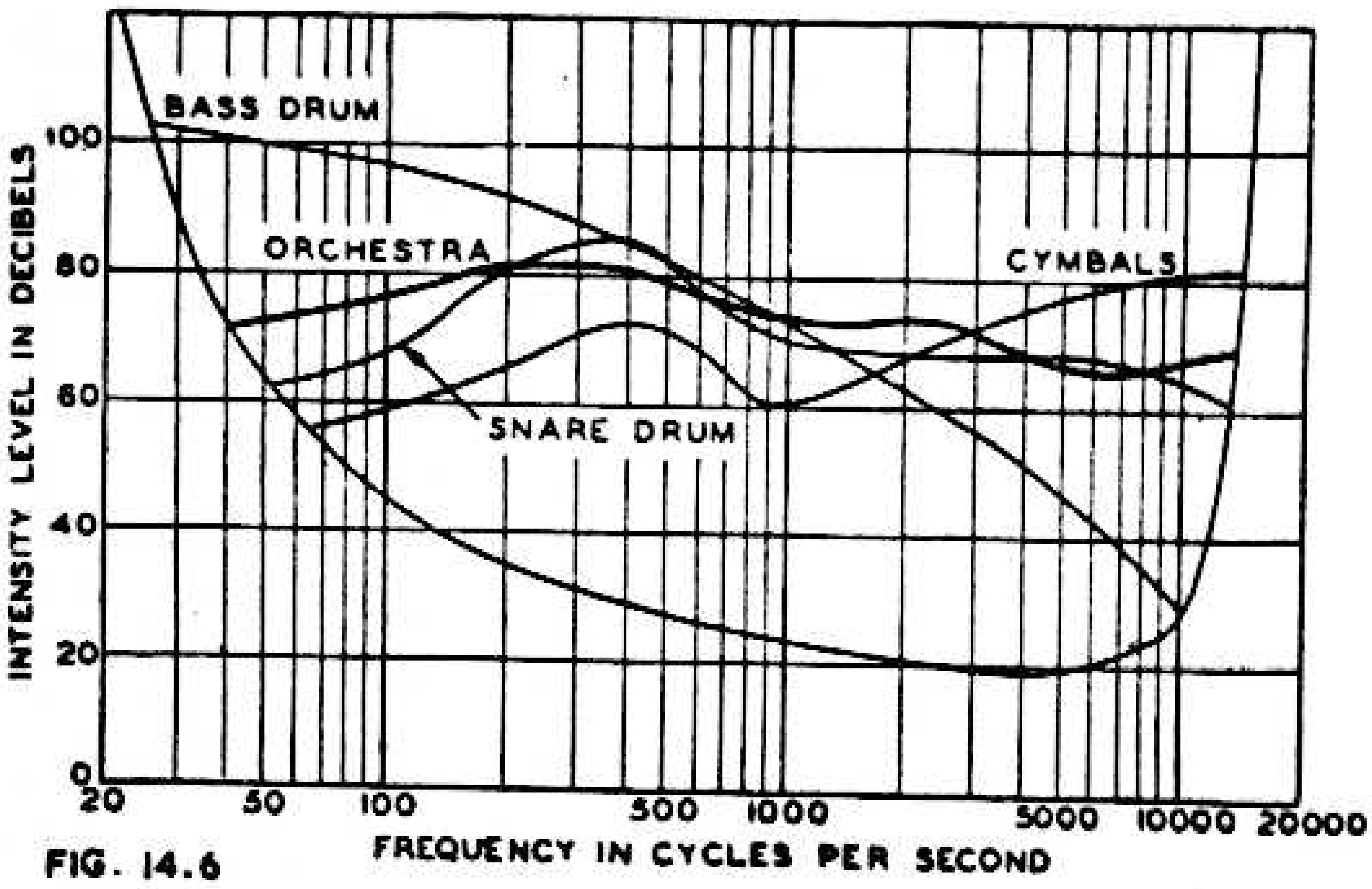


FIG. 14.6

Fig. 14.6. *Maximum r.m.s. intensity levels in* 1/4 *second intervals in critical frequency bands for whole orchestra and certain instruments. After Fletcher* (*Ref. A*3).

Fig. 14.7 (highest curve) shows the same " orchestra " curve as Fig. 14.6, but combined with the threshold of audibility curve for a very critical listener, indicating that the maximum possible frequency range is from 32 to 21 000 c/s. These are the most extreme conditions to be taken into account for high fidelity reproduction. If this orchestra is reproduced at lower levels as indicated by the lower " orchestra " curves in Fig. 14.7, the frequency range is more restricted, particularly at the lower end. The values for both average and critical listeners are tabulated below :

Peak value	100	90	80	70 db
Average listener				
Lower frequency	45	50	70	100 c/s
Upper frequency	15 000	14 000	13 000	12 000 c/s
Critical listener				
Lower frequency	32	40	50	70 c/s
Upper frequency	21 000	20 000	19 000	18 000 c/s

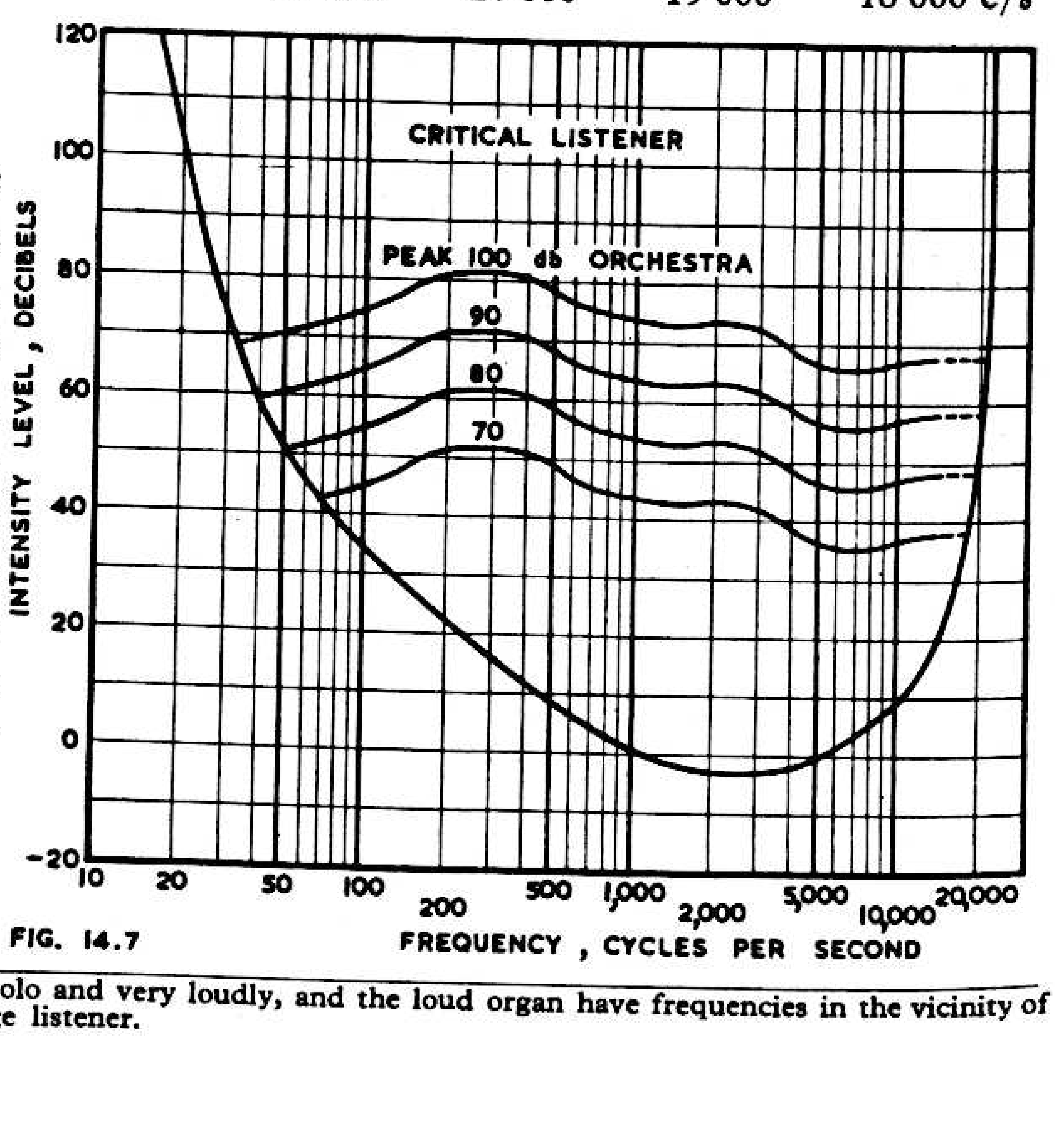


FIG. 14.7

Fig. 14.7. *Maximum r.m.s. intensity levels in 1/4 second intervals in critical frequency bands for whole orchestra, with threshold of audibility curve for very critical listener. Also illustrating effect on frequency range of reproduction at lower levels.*

*The bass drum, played solo and very loudly, and the loud organ have frequencies in the vicinity of 30 c/s audible to an average listener.

The curves of Fig. 14.6 and the orchestra curves used in Fig. 14.7 are only suitable for determining the audible frequency ranges of the various sound sources—they cannot be used for calculating the peak power.

(iii) Acoustical power and preferred listening levels

The r.m.s. peak sound level at a desirable seat in a concert hall for " full orchestra " is not likely to exceed a value of 100 db (Refs. A51, E13).

If we accept the value of 100 db as being " maximum orchestral level," then a maximum r.m.s. peak acoustical power of about 0.4 watt will be required for reproduction at the same level in a fairly large living room—see Chapter 20 Sect. 6(ii). If the loudspeaker is 3% efficient over the frequency band, then the maximum power from the amplifier will need to be 13 watts—with a more efficient loudspeaker, the power from the amplifier would be less.

Peaks above 100 db may occur occasionally with a large orchestra or choir, but they are usually " peak limited " before being broadcast or recorded. However in the design of the amplifier it is desirable to allow a margin to provide for possible peaks above this level.

Since sound levels are normally measured with a sound-level meter, it is necessary—when considering acoustical reproduction—to make allowance for the margin of 10 db or more between this level and the r.m.s. peak level.

Preferred listening levels

Tests conducted by the B.B.C. (Ref. C12) on listeners give the following preferred maximum sound level, as measured with a sound level meter :

	Public		Musicians	Programme Engineers		Engineers
	Men	Women		Men	Women	
Symphonic music	78	78	88	90	87	88 db
Light music	75	74	79	89	84	84 db
Dance music	75	73	79	89	83	84 db
Speech	71	71	74	84	77	80 db

Individual variations varied from 60 to 97 db for symphonic music, but in all cases 50% of the subjects were within $\pm$ 4 db of the mean. Increasing age showed a preference for lower listening levels.

The following table gives an approximate guide for home listening :

Loudness	Sound-level meter reading	Normal r.m.s. peak
Very loud	80 db	90 db
Loud (serious listening)	70 db	80 db
Medium (as a background)	55-65 db	65-75 db

(iv) Volume range in musical reproduction

(A) Orchestral reproduction

If the full peak orchestral dynamic range of 70 db is to be reproduced, the receiver (if any) and amplifier must have a peak dynamic range of the same amount. This may be accomplished with the peak sound intensity in the room either equal to that in the concert hall, or at a lower level. However, if the reproduced dynamic range is to be effective it must all be above the noise masking level.

Fig. 14.8 shows the extreme conditions for high fidelity reproduction, with the threshold of audibility curve for a very critical listener, combined with the masking

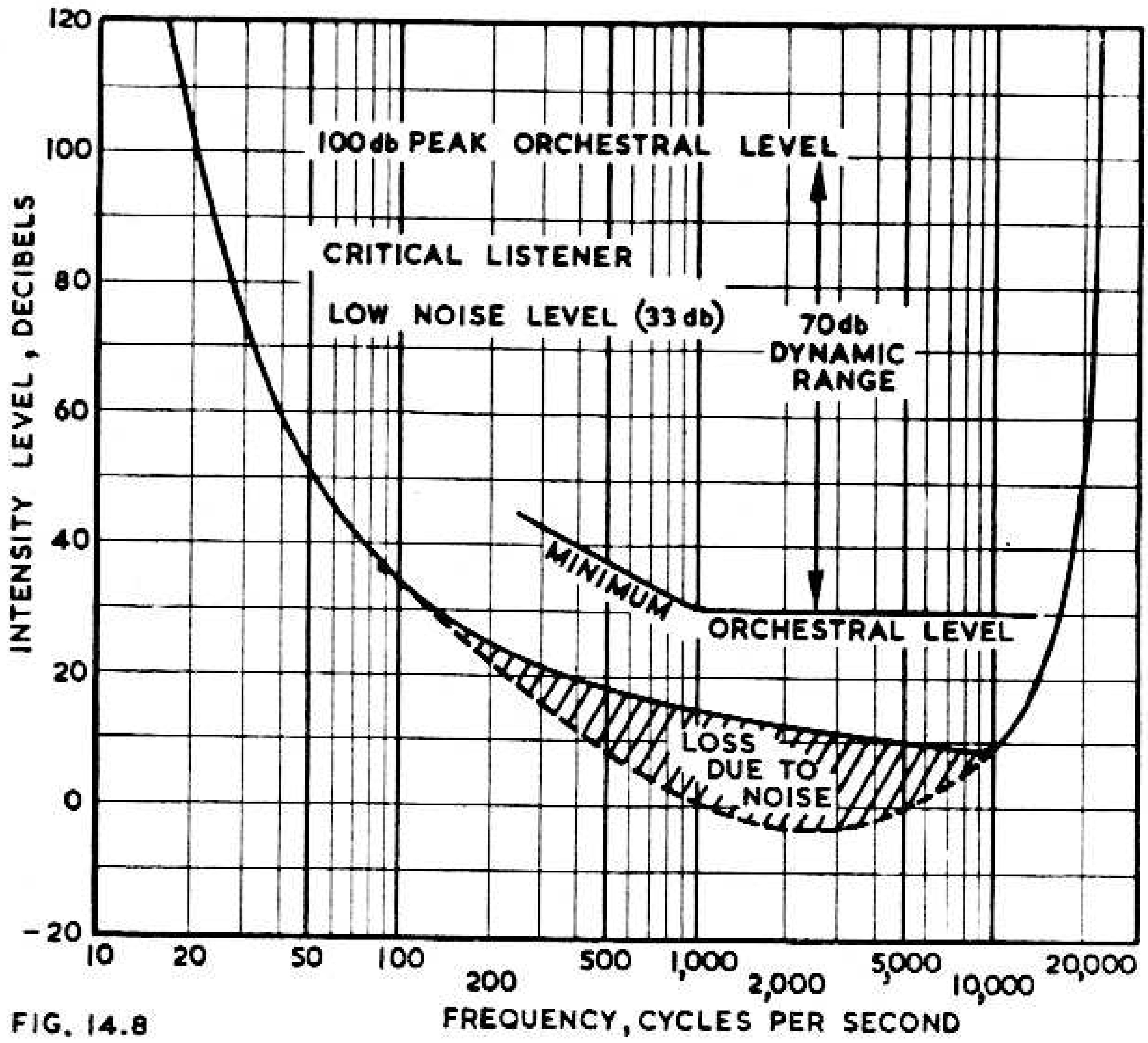


FIG. 14.8

Fig. 14.8. *Effective hearing characteristic for very critical listener with low room noise level, showing* 70 *db dynamic range for orchestra in a concert hall. Minimum orchestral level based on Ref. E*13.

effect of low room noise. The 70 db dynamic range of an orchestra in a concert hall is shown, with the minimum orchestral level of 30 db well above the masking level of the noise. In this case it would be possible to reproduce the full dynamic range at a level 15 db lower than the level in the concert hall, without loss due to noise. With an average listener and average noise level, however, the reproduction could not be more than 5 db below the level in the concert hall without loss due to noise. Any further reduction in level will result in the masking of the softest passages by the noise.

(B) Volume range broadcast or recorded

The volume range actually broadcast or recorded is frequently less than that of the original.

Source	Volume range
F-M broadcast (direct)	at least 60 db (F.C.C.)
A-M broadcast (direct)	at least 50 db (F.C.C.)
Transmission over telephone lines	40-50 db*
ditto (special lines)	50-75 db*
Lateral-cut disc records :	
shellac	30-45 db*
microgroove, say	40-56 db*
transcription (orthacoustic)	45-58 db*
Vertical-cut (hill and dale)	45-50 db*

To reduce the volume range, it is usual to employ automatic peak limiters together with volume compressors, although some manual adjustment is also made on occasions. To restore the volume range, it is sometimes possible to use volume expansion (see Chapter 16).

(v) The effect of noise

Noise as heard by the listener is contributed by many sources—for example by background noise in the studio, by the microphone and pre-amplifier, the transmitter, the receiver, the a-f amplifier and room noise. If the sound source is a record, there will be additional noise due to the recording. Under conditions for good fidelity, noise due to the transmitter and receiver (if any) can be made negligibly small. The remaining noise sources may be grouped as

*Maximum signal to noise ratio. The maximum dynamic range may be somewhat higher—see Sect. 7(v).

1. Noise from the sound source.
2. Noise from the amplifier.
3. Room noise.

It is assumed here that hum from the amplifier has been made inaudible, and then the total* noise arising from amplifier and room noise may be determined by the nomogram Fig. 19.6A when both component values are expressed in decibels. By following a similar procedure, this value may then be combined with noise from the sound source to give the total noise from all sources.

The value of room noise is 43 db for the average level (Fig. 14.4) and 33 db for the low noise level (Fig. 14.5).

It is usually simpler to combine these noises electrically than acoustically, a convenient point being the primary of the output transformer. The equivalent electrical value of the room noise for a typical living room of 3000 cubic feet is given approximately by†

Room noise	Electrical noise power (milliwatts)
33 db	$0.009/\eta$
43 db	$0.09/\eta$

Where η = loudspeaker efficiency percentage.

If the loudspeaker efficiency is 3%, the equivalent electrical value of the room noise is approximately − 25 dbm for low room noise, or − 15 dbm for average room noise.

The masking value of the total noise can only be determined accurately by calculating or measuring the total noise spectrum and applying the masking curves of Fletcher and Munson (Ref. E9 Fig. 15 ; also reproduced in Olson Ref. E3 Fig. 12.30). However for the simple case where room noise predominates, it may be taken as a reasonable approximation that the curves of Fig. 14.3 may be moved vertically as required and that the masking ordinate at 1000 c/s is 18 db below the total noise level.

Thus the masking effect at 1000 c/s of room noise alone is approximately − 43 dbm for low room noise, or − 33 dbm for average room noise, for 3% loudspeaker efficiency and a room volume of 3000 cubic feet. If the total noise from other sources is equal to the room noise, these values would be increased by 3 db.

For high fidelity reproduction, the amplifier and pre-amplifier noise should be kept below the lowest room noise level. Under the preceding conditions, with room volume 3000 cubic feet and loudspeaker efficiency 3%, a reasonable value for total amplifier noise‡ appears to be about − 30 dbm, giving a masking level 5 db below that of the lower value of room noise. For values of loudspeaker efficiency other than 3%, the value of total amplifier noise is given by

$$\text{amplifier noise power} = 0.003/\eta \text{ mW} \qquad (1)$$

$$\text{or amplifier noise in dbm} = 10 \log (0.003/\eta) \qquad (2)$$

where η = loudspeaker efficiency percentage.

See also Chapter 18 Sect. 2(ii) pages 782-783 for noise in pre-amplifiers ; Chapter 19 Sect. 6(iii) page 829 for the measurement of noise in amplifiers.

SECTION 8 : SCALE DISTORTION

When music or speech is reproduced in a room with flat frequency response and at a level such that the listener experiences the same sensation of loudness as he would in the concert hall or studio, the tonal balance will be the same as in the original (assuming that his seat occupies the same position as the microphone). If the level of reproduction is less than the original, the bass will be weak in comparison with the middle frequency range from 600 to 1000 c/s, while the frequencies above 1000 c/s will also be attenuated, although only slightly. This is illustrated in Fig. 14.9 for one selected condition with 90 phons average original level, reproduced at an average level of 70 phons ; the loss at 50 c/s is over 10 db.

*It is assumed that the room noise can be treated as random noise. Random noises are additive.
†Based on Chapter 20 Sect. 6(ii)B.
‡For measurement of amplifier noise see Chapter 19 Sect. 6.

FIG. 14.9

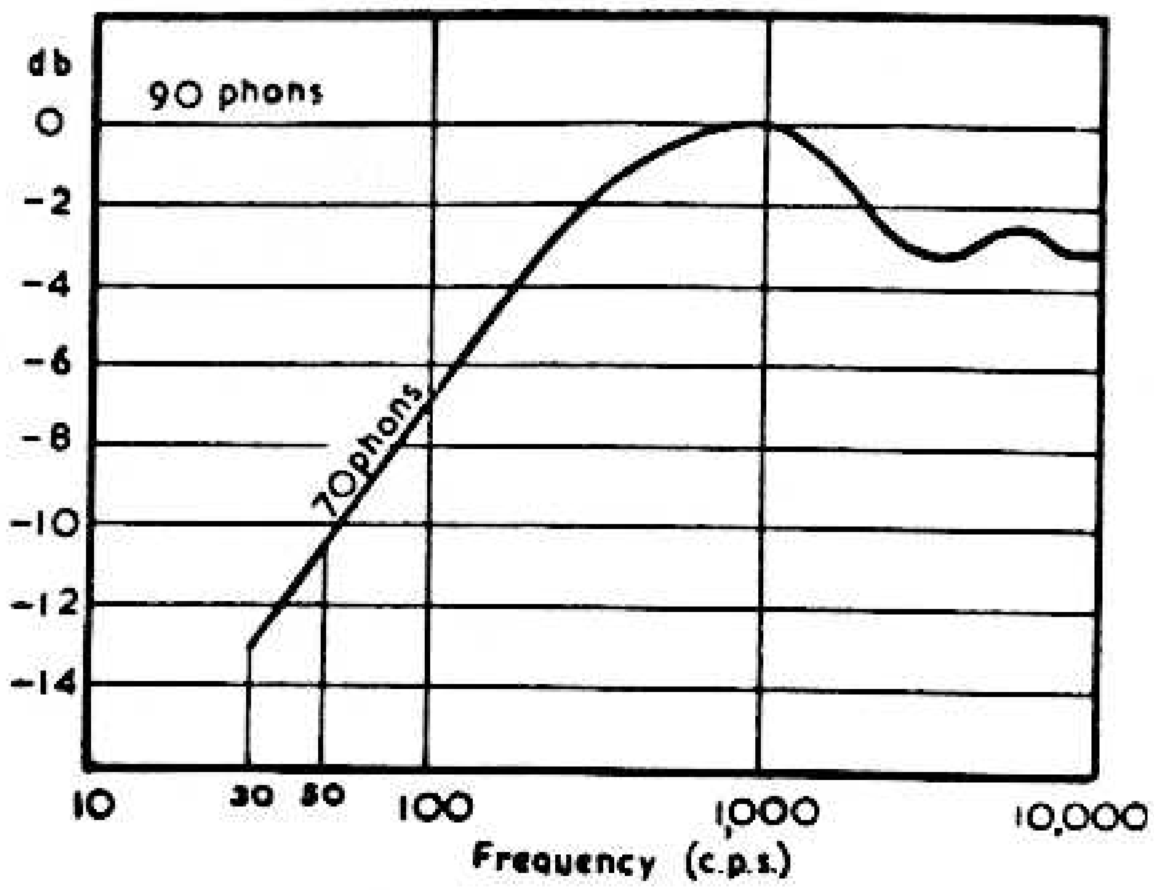


Fig. 14.9. *Loss of bass and high audio frequencies caused by reproduction at a loudness of* 70 *phons as compared with the original at* 90 *phons* (*based on Fig.* 19.7).

On the other hand, if the reproduction is louder than the original, the bass (and to some extent the treble) will be accentuated. This commonly occurs with the human voice when reproduced at a high level, unless some correction is applied in the studio amplifier.

It is for this reason that orchestral or organ music, when reproduced at normal room volume, sounds weak and uninteresting. The application of the correct amount of bass boosting assists considerably in maintaining realism and tonal balance. This application of bass boosting may be made automatically with the adjustment of the volume control, provided that maximum setting of the control corresponds to maximum volume from the loudspeaker on musical peaks (Chapter 15 Fig. 15.59). This provision makes a second (semi-fixed) volume control desirable in reproduction from records—it calls for an exceptionally flat a.v.c. characteristic in a receiver, in addition to the double volume control. If the automatic arrangement is adopted without these precautions being taken, it is only partially effective, and many users prefer a manually operated bass-boost. The automatic arrangement is only satisfactory when the tonal balance is correct at the maximum setting of the volume control ; it merley boosts the bass when the setting of the control is lowered.

Refs. A7, A29, A30, A33, A34, A56.

SECTION 9 : OTHER FORMS OF DISTORTION

(*i*) *Frequency-modulation distortion* (*ii*) *Variation of frequency response with output level* (*iii*) *Listener fatigue.*

(i) Frequency-modulation distortion

Frequency-modulation distortion (or " wow ") tends to occur in recording, and is a cyclic variation of a higher frequency by a lower frequency. Frequency-modulation distortion may also occur in loudspeakers—see Chapter 20 Sect. 7(ii).

(ii) Variation of frequency response with output level

The frequency response of an amplifier should be substantially the same at all working levels. Amplifiers should be tested for frequency response at maximum output and at 30 and 60 db below maximum output. Output transformers are a frequent cause of variations in frequency response at different levels.

(iii) Listener fatigue

There is no doubt that listening to reproduced music causes listener fatigue much more rapidly than listening to the original. This listener fatigue is caused by the necessity for mental processes arising from the unnatural effects in the hearing system.

Probably the creation of synthetic bass, intermodulation distortion and transient distortion play important parts in producing this fatigue.

An excessively high background noise level, whether caused by hum, record scratch or any form of extraneous noise, is also an important factor contributing to listener fatigue.

Ref. C7.

SECTION 10 : FREQUENCY RANGE PREFERENCES

(i) Tests by Chinn and Eisenberg (ii) Tests by Olson (iii) Single-channel versus dual-channel tests (iv) Summing up.

(i) Tests by Chinn and Eisenberg (Ref. C1)

This was the first large scale series of tests on a number of listeners in order to determine their preferences in the frequency range of reproduced music. The ranges were (3 db attenuation) :

Wide	35	to	10 000	c/s.
Medium	70	to	6 500	c/s.
Narrow	150	to	4 000	c/s.

The equipment included a single channel amplifier and loudspeaker system.

Only 12% of the listeners preferred the wide range on classical music, and 21% on male speech ; the remainder had no preference or were divided between narrow and medium. Tests were made at indicated sound intensities of 50, 60 and 70 db, the preference being between 60 and 70 db, varying with the type of music ; a somewhat higher sound intensity was preferred for speech than for music.

These tests gave rise to considerable discussion—see also (iv) below.

Refs. C1, C2, C3, C4, C5, C6, C7, C8, C9, C14, A12, A34.

(ii) Tests by Olson (Ref. C10)

In this case preferences were made on the frequency range for orchestral music with the listeners in the same room as the orchestra. No amplifying equipment was used ; the restriction of frequency range was accomplished by an acoustical filter. Listeners were given a choice between a high frequency limit of 5 000 c/s and full frequency range.

From 66% to 69% of the listeners preferred the full frequency range on music, and there was a majority preference for the full range on speech in " all acoustical " tests.

Refs. C10, C11, C7.

(iii) Single-channel versus dual-channel tests

Webster and McPeak (A19) carried out tests with single channel and dual channel (audio perspective) transmission of a live programme, compared with transcribed music. The majority preferred the transmission of a live programme to transcribed music. The majority also preferred the dual-channel to the single-channel transmission. The results, however, have been seriously challenged regarding their interpretation (C7, A12). The tests gave no indications of the preferred frequency range.

Hilliard (A27) expressed the opinion that the entertainment value of a two channel system reproducing frequencies up to 3500 c/s is better than a single channel system with 15 000 c/s response.

It seems to be generally agreed that any satisfactory form of stereophonic reproduction is very much preferable to single channel—see Chapter 20 Sect. 6(v).

(iv) Summing up

The tests by Chinn and Eisenberg indicated that, under their conditions of test, the majority preferred a restricted frequency range. The tests by Olson indicated that, under his "all acoustical" listening conditions the majority preferred an unrestricted frequency range. Obviously the conditions of the tests are of vital importance—see discussion (i) above, also summing-up by Olson, C10.

Subsequent amplifier tests by Olson have indicated that an audience was able to perceive a value of 0·7% total harmonic distortion (see Sect. 2 and Ref. E3). A reasonable inference is that the reduction in frequency range preferred by listeners in the Chinn and Eisenberg tests was caused by distortion present in the sound system.

It has been pointed out (Ref. C14) in connection with the Chinn and Eisenberg tests, that the choice between "wide" and "medium" (at 60 and 70 db) and between "medium" and "narrow" (at 50 and 60 db) was complicated by inadequate discriminability.

SECTION 11 : SPEECH REPRODUCTION

(i) The characteristics of speech (ii) Articulation (iii) Masking of speech by noise (iv) Distortion in speech reproduction (v) Frequency ranges for speech.

(i) The characteristics of speech

It has been shown (Ref. A3) that conversational speech audible at 2½ feet occupies a frequency band from 62 to 8000 c/s. The average conversational speech power is about 10† microwatts, loud shouting power is about 5 milliwatts, while the softest conversational level is about 0.01 microwatt (Ref. E5). The peak dynamic range is about 40 db for any one speaker in ordinary conversation, but may reach 56 db as an extreme limit in the general case (Ref. A3).

The peak power in conversational speech may reach 100 or even 200 times the average speech power. The octave with maximum energy in men's voices is from 300 to 600 c/s, and for women's voices from 555 to 1110 c/s, the energy in the octave being 31% of the total for both cases (Ref. E5).

The characteristics of a man's conversational speech at a distance of 2½ feet have been given as (Ref. A3) :

Level of speech above threshold (± 5 db)	63 db
Long root-mean-square intensity level	68 db
Peak level in 1/8 second intervals exceeded 5% of the time	85 db
Root-mean-square level in 1/8 second intervals exceeded only 1% of the time	78 db

Refs. A3, D5, D9, D20, E4, E5, E6.

(ii) Articulation

The articulation of speech is defined as the percentage of syllables interpreted correctly (Ref. D8). The results for an average listener receiving syllables at optimum intensity are given below (Ref. A3) :

Articulation	Minimum frequency	or	Maximum frequency
98%	40 c/s		15 000 c/s
98	100		12 000
98	250		7000
96	570		5000
94	720		3900
90	960		3100
80	1500		2300
70	1920		1970
60	2300		1700
50	2600		1500

†A more recent test gave 34 microwatts for men and 18 microwatts for women (Ref. D5).

Columns 2 and 3 are the results of separate tests with a high-pass filter and a low-pass filter respectively. They cannot be combined to give a pass-band with the limiting frequencies as tabulated, except as a rough approximation for high percentages of articulation (> 90%) ; in this case the two articulation values should be multiplied together (e.g. 96% × 96% = 92% articulation approximately for a pass band from 570 to 5000 c/s).

There is no loss in articulation with a frequency range (pass band) from 250 to 7000 c/s, while the loss is quite small with a range from 570 to 5000 c/s. With continuous speech the loss in articulation is noticeably less than with syllables.

Owing to the unnaturally high-pitched tone and obvious lack of balance, these frequency ranges are extended towards the bass for all communication purposes.

An articulation index has been proposed (Ref. D3) which may be calculated for any known conditions, including noise. A value of 1.0 is the maximum (for perfect articulation) while a value of 0.5 or over is entirely satisfactory.

Those who are interested in the intelligibility of speech either with or without noise interference should read Refs. D10, D11, and D12.

Experiments have shown that intelligibility is impaired surprisingly little by the type of amplitude distortion known as **peak clipping**. Conversation is possible even over a system that introduces " infinite " peak clipping, i.e., that reduces speech to a succession of rectangular waves in which the discontinuities correspond to the crossings of the time axis in the original speech signal. The intelligibility of the rectangular speech waves depends critically upon the frequency-response characteristics of the speech transmission circuits used in conjunction with the " infinite clipper " A word articulation value of 97 was obtained using firstly a filter providing a frequency-response characteristic rising 6 db/octave, followed by the infinite peak clipper, followed in turn by a filter with a characteristic falling at the rate of 6 db/octave (Ref. D13).

References to articulation : A3, D3, D8, D10, D11, D12, D13, D19.

(iii) Masking of speech by noise

Noise has a similar masking effect with both speech and music. If the noise level is reasonably low there is very little effect on the reproduction of speech, since the dynamic range is not excessive. It is sometimes required to operate loudspeakers in very noisy locations, and in such cases it is helpful to provide the amplifier with a bass cut-off at 500 c/s, since the maximum intensity levels of speech (particularly declamatory* speech) occur below this frequency, and the effect of the lower frequencies on the articulation is negligible. This permits the middle frequencies from say 1000 to 3500 c/s to be boosted to the threshold of pain at 120 db, thus bringing them above the noise. An alternative arrangement is to give a treble boost of 6 db per octave over the whole useful frequency band, up to at least 3000 c/s—this will automatically reduce the bass frequencies as desired (D2, D4). This latter method requires much less acoustical power—19% of that of a flat system with 500 c/s cut-off —but the effect sounds unnatural.

Refs. A3, D2, D3, D4, D14, D15, D16, E4.

(iv) Distortion in speech reproduction

A considerable degree of distortion is possible before speech becomes objectionably distorted—about 15% second harmonic with a high frequency cut-off at 3750 c/s. This may be put in the form of about 48% intermodulation distortion (r.m.s. sum) with a similar cut-off frequency.

If the frequencies below 500 c/s are attenuated, the distortion for a given loudness will be much less than for a wide frequency range.

*Very loud public speaking.

(v) Frequency ranges for speech

Application	Articulation	Frequency range
High fidelity reproduction	98%	62-8000 c/s
Good fidelity	98%	150-7000 c/s
Fair fidelity (Public address)	96%	200-5000 c/s
Restricted bass*. Unbalanced	96%	500-5000** c/s
Restricted bass and treble*	95%	500-4000** c/s
Very restricted*	90%	500-3000** c/s
Telephone		300-3400** c/s

*For noisy locations. **Response may be peaked.

SECTION 12 : HIGH FIDELITY REPRODUCTION

(*i*) *The target of high fidelity* (*ii*) *Practicable high fidelity* (*iii*) *The ear as a judge of fidelity.*

(General references A3, A9.)

(i) The target of high fidelity

High fidelity reproduction is essentially reproduction such that the most critical person can listen intently to it without any apparent distortion and without any appreciable fatigue, other than any effects due to the single point source of sound. With the present limited state of knowledge we have no definite measurable limits which can be set for each form of distortion, to ensure high fidelity.

However, it is generally agreed that a high fidelity amplifier should comply with the following specification. Values in brackets are rather extreme.

Frequency range : 40-15 000 (30-20 000) c/s.

Variation in output $\pm$ 1 db ($\pm$ 0.5 db) at three levels.

Total harmonic distortion not more than 1% (0.7%).

Intermodulation distortion not more than 3% (2%), measured at 40 c/s and, say, 7000 c/s—r.m.s. sum.

Power output sufficient to ensure that overloading does not occur.

Phase angle as small as practicable.

The desired degree of damping on loudspeaker at bass resonant frequency.

Hum inaudible.

Noise* to give specified dynamic range 70 (80) db :

Total noise† 52 (62) db below max. r.m.s. peak signal.

Noise* to be inaudible in low noise room :

For room volume 3000 cubic feet—

Loudspeaker efficiency	3%	10%	30%	45%
Total noise	−30	−35	−40	−42 dbm

Conditions of test—Including microphone (or other source) and loudspeaker.

This specification does not include some very important loudspeaker characteristics, such as frequency-modulation distortion, time delay for specified increase in sound pressure, transient decay characteristics, and sub-harmonics (see Chapter 20 Sect. 7). These have been omitted owing to difficulties in testing and the lack of standardized procedure.

(ii) Practicable high fidelity

There is no great difficulty in meeting the specification given in (i) above so far as a microphone and amplifier are concerned. However, when a loudspeaker is added, to say nothing of the additional distortion contributed by transmission and reception, or reproduction from disc records, it is impossible to meet this specification.

*See Sect. 7(v).

†Based on total noise 18 db above masking level of noise at 1000 c/s. For measurement of noise see Chapter 19 Sect. 6.

This raises the question whether such high standards are essential. The principal characteristics in question are variation in output, distortion, and frequency range.

Variation in output over the frequency range

A tolerance of the order of $\pm$ 5% appears to be generally acceptable for the overall electro-acoustical performance of a sound system.

Distortion and frequency range

The permissible non-linear distortion in a high fidelity amplifier is dependent on the frequency range, the type of programme, the critical character of the listener and the sound level. It has been observed that the sensitivity of the ear to distortion in music appears to be a maximum for sound levels in the vicinity of 70 to 80 db (Ref. A51). Hence for sound levels of the order of 90 db (sound level meter indication), somewhat higher values of distortion than those specified in (i) above would be permissible—possibly of the order of 2% total harmonic distortion with a cut-off frequency of 10 000 c/s, and 2.5% with a cut-off frequency of 7500 c/s.

Most direct radiator loudspeakers exceed 2% total harmonic distortion over certain frequency bands when operated to give a sound level (meter reading) of 90 db in a fairly large living room. However, the problem is complicated by the fact that many types of music have maximum intensity levels in the 200 to 500 c/s frequency band where loudspeakers often have distortion values well below their maximum values.

When the distortion in any individual case is distressing to a listener, the high cut-off frequency should be reduced until he is relieved from the discomfort. It is better to have freedom from discomfort than to have a wide frequency response, particularly when listening for a sustained period. This leads to the conclusion that " the true measure of the quality of an electro-acoustical system is the maximum bandwidth which the public finds acceptable " (Ref. C7). Thus all high fidelity amplifiers should have a choice of high roll-off* frequencies.

In addition, it is necessary to provide the listener with tone controls on treble and bass, with the choice of boosting or attenuation on each (see Chapter 15). He can then please himself on the choice of tonal balance and frequency range.

In the writer's opinion, nearly distortionless reproduction of bass frequencies up to say 400 c/s is far more important than reproduction of frequencies above 10 000 c/s.

The importance of **improved bass response** in reproduced music is gradually gaining recognition. Various methods are being adopted to produce acceptable extreme bass response without requiring an excessively large loudspeaker. Fortunately the extension of the bass range, with or without bass boosting, does not directly result in any increase in the subjective effects of the distortion such as occurs with extension of the treble range.

High fidelity reproduction with a level frequency response, when compared with a distorting amplifier, seems to lack bass. If a fair comparison is to be made between them, when operating at the same sound levels, the high fidelity amplifier will require bass boosting.

A direct radiator loudspeaker, or a conventional horn type, cannot give high fidelity reproduction at frequencies lower than the bass resonance, because the distortion rises very rapidly. Thus the minimum frequency for high fidelity depends upon the loudspeaker and any extension to lower frequencies will require another, and usually larger and more expensive, loudspeaker.

The output transformer should, for high fidelity, have low distortion at the lowest frequency which the loudspeaker can handle. A transformer to handle high power at very low frequencies and low distortion is both heavy and expensive, but is not otherwise difficult either in design or manufacture.

If the output transformer and loudspeaker are incapable of handling frequencies below a critical value without serious distortion, it is wise to filter out these lower frequencies and so rid the amplifier of their bad effects. It is better to have a clean bass limited to say 80 or 100 c/s, than to have a distorted condition resulting in intermodulation effects over a wide frequency range.

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*Sharp cut-off characteristics are undesirable for fidelity. However a roll-off and sharp cut-off may be combined to give an acceptable result.

(iii) The ear as a judge of fidelity

It is common practice to regard the ear as the final judge of fidelity, but this can only give a true judgment when the listener has acute hearing, a keen ear for distortion, and is not in the habit of listening to distorted music. A listener with a keen ear for distortion can only cultivate this faculty by making frequent direct comparisons with the original music in the concert hall.

Non-linear distortion in any good quality **amplifier** should be so low as to be inaudible to the most critical listener. This distortion can therefore only be checked by measurements.

The ear is the only final judge of fidelity with **loudspeakers**, although it should be supplemented by measurements of harmonic distortion, frequency response, frequency-modulation distortion, damping of bass resonance, time delay for 60 db increase in sound pressure, transient decay characteristics (Shorter's method), and subharmonics—see Chapter 20 Sect. 7.

The ear is the only judge of **tonal balance**.

In any acoustical test, the **sound level** should be that of normal loudspeaker operation.

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