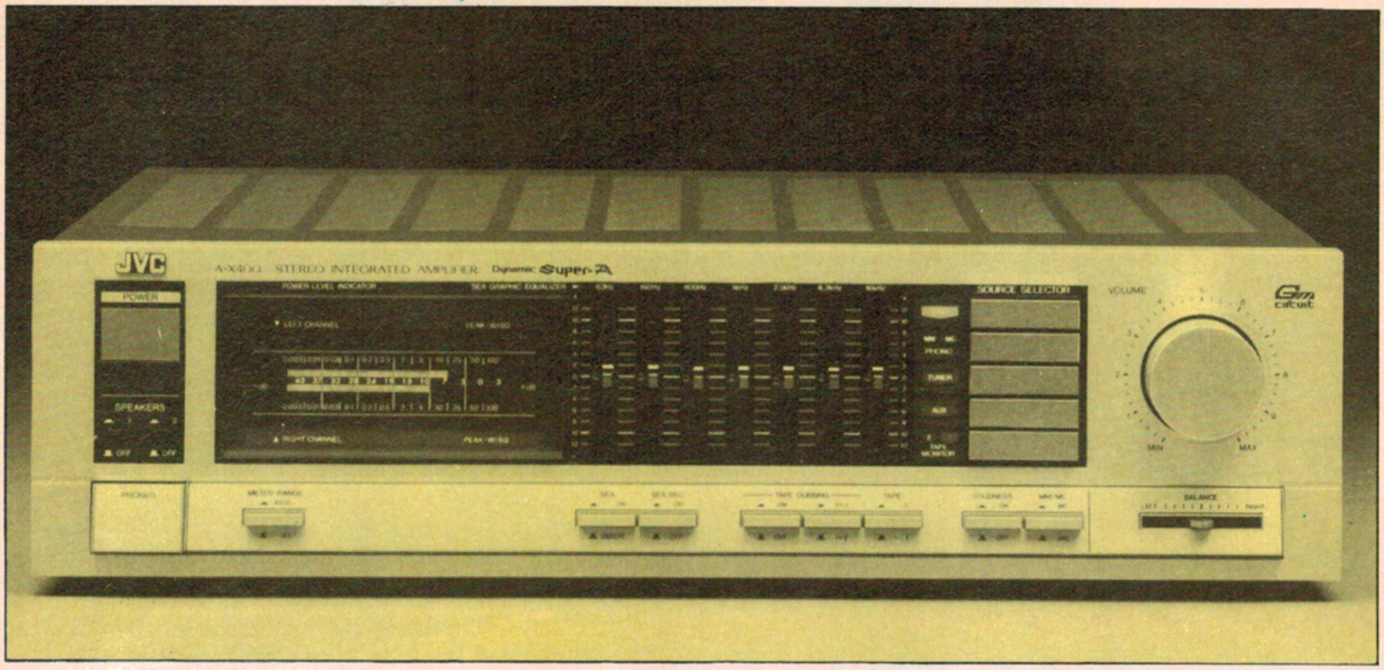


An introduction to hifi Pt 3

HIFI facts & figures

decibels, distortion, dynamic range &c.

In this chapter we have a look at some of the main parameters used to describe hifi equipment: frequency response, harmonic and intermodulation distortion, dynamic range and power output. We introduce decibels too, which are most important for an understanding of hifi performance figures.

by NEVILLE WILLIAMS

Beginning with frequency response, the components of a high fidelity sound system, and the system as a whole, should logically exhibit a substantially "flat" frequency response for all frequencies over the potentially audible range. This is commonly rounded out in hifi literature, these days, to "flat from 20Hz to 20kHz".

We say that hifi components should "logically" exhibit this flat response but not all hifi components do. For example, most FM tuners can only be expected to be flat from 20Hz to 15kHz and even wideband AM tuners are usually not even as good as this. We shall discuss the reasons for this in detail later in this series.

For the moment though, let us agree that "flat from 20Hz to 20kHz" is the normal standard we expect from high fidelity components. Anything less than this might be regarded as "medium hifi" or even "adequate hifi" but not "genuine hifi". (See Fig.1).

Graphs & decibels

The notion of frequency response being "flat" (or otherwise) can be depicted in a graph in the general form of Fig.2, showing the relative output of an audio stage, component or system along the vertical axis, plotted against frequency on a logarithmic scale along the horizontal axis.

While the vertical (output) scale appears at first glance to be linear, the units (decibels) are logarithmic, preferred because they relate more naturally to the logarithmic response of the human ear. (See Chapter 1).

Whether in an acoustic context, as in the earlier chapter, or electrical as here, a power ratio can be converted to decibels by determining the log (to the base 10) of that ratio, multiplying it by 10 and calling the result dB. In the case of voltage ratios, the log is multiplied by 20 instead of 10, to allow for the fact that power varies as the square of the voltage.

For those not familiar with logs, Table 1 shows power and voltage ratios, and their decibel equivalents, over the range most frequently encountered in audio graphs: from +30dB, through 0dB (reference) to −30dB.

If, under test, the output from a piece of audio equipment (for a given input) does not vary perceptibly across the entire audio spectrum, the response "curve" would be a "flat" horizontal line, (a) in Fig.2, from 20Hz to 20kHz, drawn (normally) at the 0dB reference level.

In some modern components such as CD players and power amplifiers, the frequency response may be either flat or so close to it that a graph is scarcely warranted. The information can be conveyed in the specifications in a few words: "Frequency response: flat from 20Hz to 20kHz"; or "Frequency response: 20Hz to 20kHz, +0 −1dB."

By contrast, curves (b) and (c) convey quite detailed information representing (b) the natural response of a magnetic phono cartridge playing an ordinary record and (c) the requisite response of a compensated phono preamplifier stage to provide a nominally flat overall frequency characteristic. A −18dB (approx) response from the cartridge at 30Hz is compensated by +18dB from the preamplifier, and so on across the spectrum.

Curves (b) and (c) emphasise the advantage of decibel notation: By simply noting the dB response of individual components in the record/replay chain, it is possible to predict the likely overall response pattern by simple addition and subtraction.

Subjective reactions

As previously stated, subjective tests have established that the smallest change in audio power level that listeners can detect under controlled conditions is about 1.6 times or 2dB.

On this basis, individual components in a sound reproducing system might reasonably be considered to have a subjectively flat frequency characteristic if their response does not deviate beyond a tolerance "window" about 2dB wide (eg ±1dB).

Many would dispute this, however, claiming that even a 0.5dB deviation over a portion of the frequency characteristic can make a subtle difference to the balance of the sound.

Whether or not this is making too fine a point for the average hifi enthusiast, the flattest possible characteristic is certainly desirable, if only to ensure that ostensibly small deviations in individual components don't add up to a sizeable prominence, or trough, or roll-off in the overall response, sufficient to affect the tonal balance of the system. (See Fig.3).

At the low frequency end, an overall response tapering downwards through, say, −10dB at 50Hz would cause a loss of real "weight" in orchestral or organ music. Conversely, a rising or peaked response in the 60-110Hz region would produce an unnatural "thumping" bass — perhaps not out of place with "pop" music!

At the high frequency end, a rising response in the 5-12kHz region can result in a strident or "edgy" quality

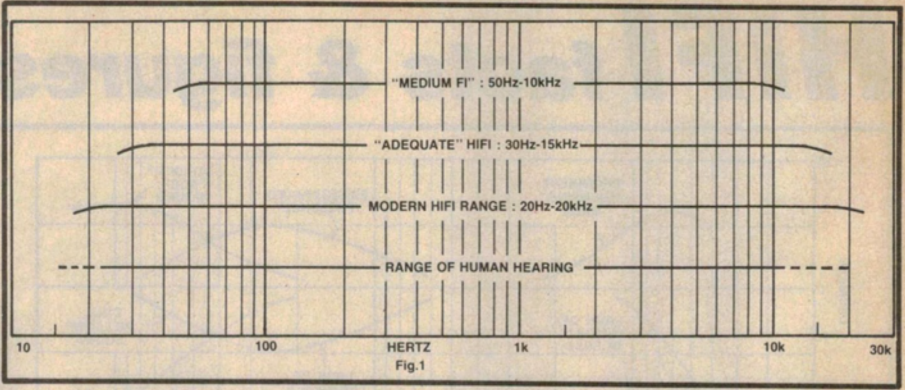


Fig.1: The desirable frequency range for modern hifi equipment is normally considered to be flat, or substantially so, from 20Hz to 20kHz. Some components are still limited to the "adequate" range.

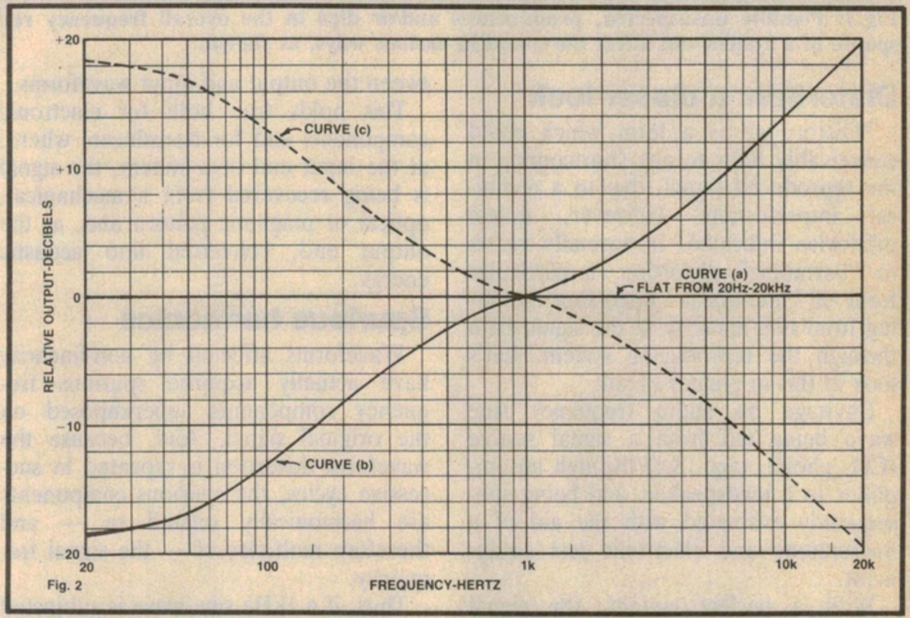


Fig.2: A typical group of frequency curves: (a) is flat from 20Hz to 20kHz; (b) is the nominal frequency characteristic of a magnetic phono cartridge, while (c) shows the required response for a phono preamplifier. Together, (b) and (c) would provide a flat response overall.

while a falling response can rob instruments of their natural harmonics producing, instead, an unmusical "mellow" uniformity in their sound.

A too-prominent middle response (2-5kHz) can impart a flattering "presence" to solo voices and instruments, and even to chamber groups, but this is at the expense of an overall balance on a full orchestra. A lack of middle response, on the other hand, can heighten the impression of dispersed, wide-range sound, but may impart a sense of remoteness, as distinct from presence, to solo voices and instruments.

There is something suspect about equipment which sounds good on one kind of program but poor on another. A good system will not tend to favour any particular class of sound or instrument at the expense of the rest!

DECIBELS (dB)	POWER RATIO	VOLTAGE RATIO	DECIBELS (dB)	POWER RATIO	VOLTAGE RATIO
+30	1000	31.62	0	1.0	1.0
+25	316	17.78	-1	.79	.89
+20	100	10.00	-2	.63	.79
+19	79.43	8.91	-3	.50	.71
+18	63.10	7.94	-4	.40	.63
+17	50.12	7.08	-5	.32	.56
+16	39.81	6.31	-6	.25	.50
+15	31.62	5.62	-7	.20	.45
+14	25.12	5.01	-8	.16	.40
+13	19.95	4.47	-9	.13	.35
+12	15.85	3.98	-10	.10	.32
+11	12.59	3.55	-11	.08	.28
+10	10.00	3.16	-12	.06	.25
+9	7.94	2.82	-13	.05	.22
+8	6.31	2.51	-14	.04	.20
+7	5.01	2.24	-15	.032	.178
+6	3.98	1.99	-16	.025	.159
+5	3.16	1.78	-17	.020	.141
+4	2.51	1.58	-18	.016	.126
+3	1.99	1.41	-19	.013	.112
+2	1.59	1.26	-20	.010	.100
+1	1.26	1.12	-25	.003	.056
0	1.00	1.00	-30	.001	.032

Table 1: Showing equivalent power and voltage ratios from +30 to −30dB, this covers the range most commonly encountered in performance curves. Check it against your own log tables.

HIFI facts & figures

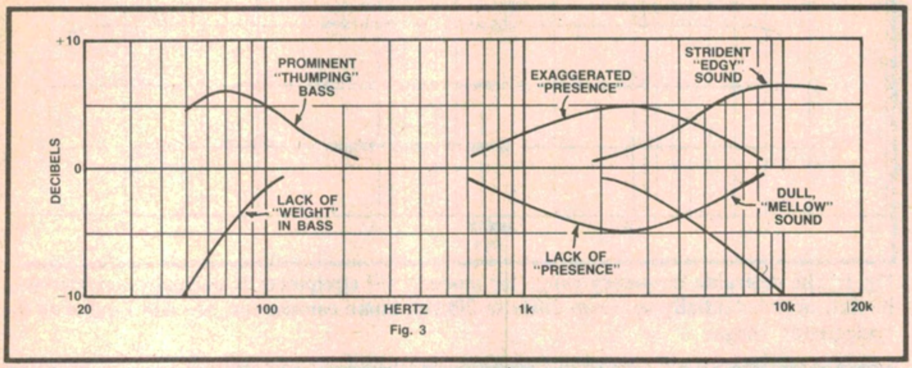


Fig.3: Possibly unsuspected, prominences and/or dips in the overall frequency response of a system can affect the sound in various ways, as shown.

Distortion: a closer look

"Distortion" is a term which could conceivably refer to any shortcoming in the reproduced signal, due to a technical imperfection. However, unless otherwise indicated, it normally refers to "harmonic" distortion, a particular form of "non-linear" distortion, resulting from non-linearity in the signal path through the reproducing system. Let's look at this in greater detail:

Envisage an audio frequency sine wave being fed from a signal source (CD, phono, tape, &c) through an amplifier to a loudspeaker, and being subsequently evaluated with the aid of a microphone and electronic test equipment.

With a perfect system, the signal would be a pure sine wave at all stages, with no deviation from its mathematically based contour and therefore no distortion in terms of wave shape.

However, in practical, as distinct from perfect equipment, the input/output relationship may not be completely linear, resulting in some discrepancy between the output and input waveforms.

This holds true both for electronic components and for transducers where, at the input end of a system, the signal is being recovered from a mechanical, optical or magnetic pattern and, at the output end, converted into acoustic energy.

Spurious harmonics

Waveforms affected by non-linearity have actually acquired spurious frequency components superimposed on the original signal. And, because the waveform distortion is repeated in successive cycles, the spurious components are harmonically related to — and therefore multiples of — the signal frequency.

Thus, if a 1kHz sine wave is subjected to non-linear processing, it will acquire spurious harmonics, for example: a second harmonic at 2kHz, a third harmonic at 3kHz, a fourth at 4kHz and so on.

Fig.4 should serve to illustrate the effect of non-linear processing on a signal passing through an audio component. In each example, an input sine wave is plotted vertically, while the resultant output waveform is to the right of the input/output transfer curve.

With a linear transfer characteristic (shown dotted) the output would be a sine wave, free from harmonic distortion.

However, in (a) the transfer characteristic deviates from linear at one end while, in (b) the characteristic is S-shaped. Both are exaggerated in the drawing for purposes of illustration but both conditions are encountered to some degree in practical electronic amplifiers and transducers.

In (a), one excursion of the output wave is compressed and analysis would normally show that it has acquired a percentage of predominantly even order harmonics: second, fourth, sixth, &c. In (b), both excursions are compressed and the harmonic structure would be predominantly an odd number: third, fifth, seventh, &c.

While it is possible to measure the amplitude of individual harmonics, engineers more commonly use a "Distortion Factor Meter" to indicate the sum of all the spurious components present: actual harmonics plus any noise contributed by the equipment under test.

Described (somewhat loosely) as the total harmonic distortion (or THD), the indicated sum is expressed as a percentage of the total output. Fairly obviously, the lower the figure, the better is the linearity (and the lower the noise content) of the particular component.

Laboratory test procedures are based on a sine wave input signal as the most practical method but complex signal waveforms are affected by non-linearity in a similar manner, the difference being that every single frequency present in a program signal may acquire its own array of spurious harmonics!

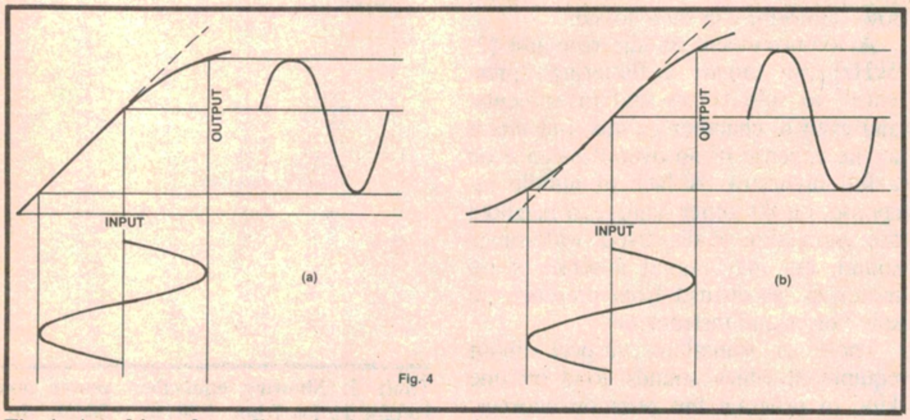


Fig.4: Any hint of non-linearity in the transfer characteristic will result in harmonic and intermodulation distortion, gross as in the illustration, or small but subtle in good quality audio equipment.

Levels of distortion

In days when the standard of sound reproduction was determined mainly by AM radio and 78rpm records, a comparatively high degree of non-linear (therefore harmonic) distortion was tolerated, even in better quality equipment — often amounting to several percent.

The level of distortion in modern, good quality equipment is very much lower, and, instead of several percent, is more likely to be around 0.1% with some components such as CD players and high quality amplifiers, exhibiting less than .01%.

It is not possible to nominate any single figure for THD which reliably represents the borderline between what is subjectively objectionable and subjectively tolerable. Much depends on the circumstances.

At the low fequency end a loudspeaker, for example, may exhibit several percent of second and/or third harmonic distortion when driven at 20-40Hz. In the guise of musical harmonics, the spurious components may significantly modify the nature of the sound, without being recognised as distortion.

At the high frequency end, phono pickups may produce mis-shapen waveforms — therfore harmonics — due to poor tracking above about 6kHz. But, because many such harmonics lie above 12kHz, they are not heard by most listeners.

Again, low order "spurious" harmonics, may be perceived as natural, harmonics of solo (notably brass) instruments, modifying their timbre without necessarily being interpreted as distortion.

Yet, in other circumstances, similar percentages of distortion could prove totally objectionable. Clearly, a high quality sound system should be able to reproduce all the complex harmonics and subtle overtones that are contained in the original signal but, ideally, should not add any of its own.

Intermodulation effects

As distinct from the generation of harmonics, non-linearity also causes signals which are being handled simultaneously to intermodulate, producing additional sum and difference frequencies. For example, signals at say, 400 and 700Hz could intermodulate to produce resultants at 7400 and 6600Hz — both totally dissonant in terms of the original tones.

Intermodulation distortion ("IMD") becomes progressively more noticeable — and objectionable — with more complex signals, and is blamed for much of the "congestion" or "muddiness" that can compromise orchestral or choral recordings. It may be less noticeable on the same equipment playing solo or chamber music.

The percentage of IMD, as measured and specified, depends on the nature of the non-linearity and on the test conditions. It may come out at about the same figure as the measured THD, or it may be 3 or 4 times higher.

Again, it is not possible to nominate any one figure representing the borderline between a tolerable level of IMD and otherwise. The objective can only be to aim for the lowest possible figure for any given type of component.

Dynamic range

In the context of hifi sound reproduction, dynamic range might be defined as the ratio, expressed in decibels, between the softest and loudest sounds which occur at a performance, or are effectively transmitted or recorded and reproduced, or are capable of being heard in a particular listening environment.

By way of example, as measured on a sound level meter in a (relatively) quiet concert hall, the SPL (sound pressure level) readings of a large symphony orchestra may range from about 30dB during the softest passages (ppp) to about 100dB at the other extreme (fff) — a dynamic range of up to 70dB.

Over the years, this has served as something of a target figure for recording and reproducing components and/or systems: the ratio between the signal level at the overload threshold and the intrinsic noise level of the component or system should be at least 70dB. (See Fig.5).

Mainly because of high surface noise, 78rpm shellac pressings were limited to 35-40dB, making it necessary for recording engineers to constantly monitor the signal to keep it within manageable levels.

With much lower surface noise, vinyl pressings offered an immediate 55dB but this figure has gradually been improved upon by better mastering technology and the development of phono cartridges and preamplifiers able to cope with higher groove modulation.

According to their jacket notes, many audiophile orchestral recordings are now made without any level adjustments during the performance, indicating a dynamic range of up to 70dB — and possibly higher in special cases.

Cassette tape equipment has undergone somewhat similar development in respect to in-home recording and replay, thanks to the emergence of improved heads, beter tape formulations and the inclusion of noise reduction systems such as Dolby-B/C, and dbx.

In the realm of broadcasting, FM-stereo can also make available programs with a similar order of dynamic range, given suitable source material, a good tuner and reasonable reception conditions.

A potential problem

Even before the arrival of compact discs and hifi VCRs, the availability of program sources with 70dB or more of dynamic range had highlighted a problem in accommodating the resulting volume levels in the average home.

It may have come as a surprise, in chapter 1, to learn that the ambient noise in a typical suburban living room has a subjective loudness level of around 40dB. If the softest sounds are set to this nominal level, so as not to be unduly masked by the noise, it follows that the loudest orchestral passages could reach a not inconsiderable 110dB at the listening position. The implications are obvious enough:

- The selected listening room must be as quiet as possible so that the ppp passages can be set to a commensurately low level.
- So as not to compound the situation, the reproducing equipment should contribute as little noise of its own as possible, yet still be capable of delivering adequate power on peaks, without perceptible overload.
- Where high peak levels are domestically unacceptable, it may be necessary to use headphones on occasions,

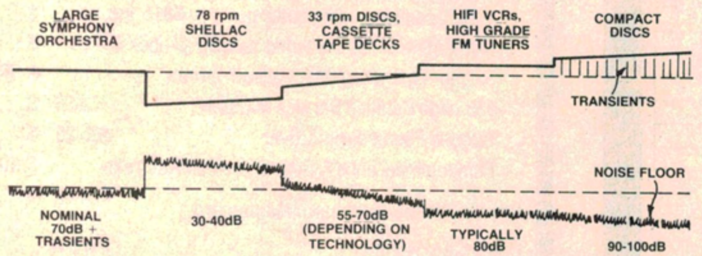


Fig. 5

Fig.5: In the past, record/replay equipment has been unable, or barely able, to cope with a realistic dynamic range. Compact discs in particular can cope with transients, as well as the main body of the sound.

or to rearrange the loudspeakers to permit closer listening.

Still higher figures?

If the dynamic range of modern "black" discs, for example, is as much as we need, or can accommodate in the home, where lies the advantage of a still higher figure from hifi VCRs and up to nearly 100dB from CD players? A good question!

In fact, the capacity for greater dynamic range does not indicate that there is any thought of encouraging orchestras to exploit the new dynamic limits — ppp and fff!

The immediate value of the new technology is that it makes it possible for even the softest sounds to be recorded well above the so-called "noise floor" of the record/replay system while, at the same time, ensuring that the loudest sounds fall well below the overload region.

The benefit becomes apparent when listening to a digitally mastered compact disc. Even as heard in the ambience of an ordinary listening room, the sensation has frequently been described as "one of listening to sound out of silence". And, in the loudest passages, there is no hint of stress, due to impending mechanical or magnetic overload.

But another important factor has emerged, to do with the transients associated with percussive musical sounds as, for example, those from an acoustic guitar.

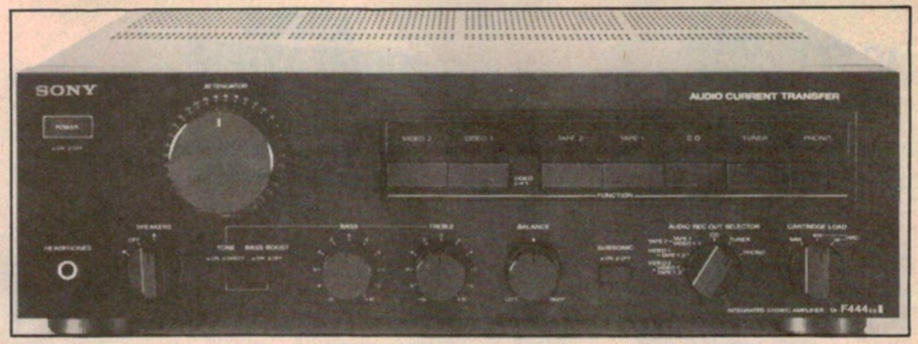
Such transients, which impart "attack" to the notes, can have a quite high amplitude but, because they are of brief duration, they do not register on a conventional sound level meter, nor do they add greatly to the subjective loudness of the sound.

In fact, they tend to be "crushed" by conventional analog tape master recorders, and audiophiles only became aware of what they were missing, with the release of "direct cut" and digitally mastered discs, having greater dynamic range and the ability to accommodate fleeting transient peaks.

And there lies the further advantage of those extra decibels of dynamic range: not just greater freedom from noise and from congestion on loud passages, but the ability to preserve transient peaks on which depends much of the attack and "vitality" of reproduced sound. At stake is not louder sound but better sound!

Power output

As we shall see later, the preservation of transient peaks on modern recordings makes extra demands on the amplifier and loudspeaker system but, even apart from this, amplifier power output is subject to a good deal of misunderstanding.



Boasting a power output of 100W RMS per channel, Sony's TA-F444ESII is typical of modern up-market stereo amplifiers. Its frequency response is 2Hz-200kHz (±0.2dB) while total harmonic distortion is just .004% at rated output.

In the laboratory, the basic test for power output involves connecting a stereo amplifier to a pair of high wattage resistors instead of the usual loudspeakers, each resistor being equal in value to the recommended load, typically 4 or 8 ohms, and rated to dissipate at least as many watts as the anticipated power output of the respective channels.

A 1kHz signal is then fed to the input of both channels and gradually increased in amplitude until the amplifiers are on the threshold of overload as displayed on an oscilloscope or, more precisely, indicated by a distortion meter showing that the THD has reached a pre-determined level.

The RMS voltage (E) across the load (R) is then measured and the mathematical average power (P) for each channel calculated on the basis that:

$$P = E^2/R$$

Assuming that both channels are symmetrical, the procedure will yield a figure which is variously described as the "RMS" (strictly a misnomer) or "continuous" or "steady tone" power output, per channel, both channels driven.

A complication arises because, with most amplifiers, a continuous tone test places a heavy load on the power supply, causing a reduction in supply voltage to the output stage(s), with a consequent reduction in their power output.

Being of a more spasmodic nature, even loud program signals do not affect the supply voltage to the same extent and the maximum power available on program material — the so-called "music power" — may typically be around 20% higher than the "average" or "continuous" rating, and therefore a more flattering figure to publish.

However, largely as the result of pressure from the Federal Trade Commission and the Institute of High Fidelity in the USA, and from their European counterparts, amplifier manufacturers are now obliged to publish a continuous rating for their equipment, with all channels driven.

They may, if appropriate, quote a music power rating based on test procedures approved by the IHF. Or they may quote a figure for "headroom" in dB, the margin of power available above the continuous rating to handle program type signals. Referring back to Table 1, a mere 1dB of headroom would represent an increase in power level of 26%

The purpose of publishing a "continuous" power rating is to ensure that buyers will be aware of the minimum power available from a stereo amplifier under the most onerous program drive conditions — eg a concert organ at full volume with sustained pedal bass.

If this minimum figure is reckoned to be adequate, extra output by way of music power becomes a bonus. (We will have more to say about appropriate hifi power levels in a later chapter on amplifiers.)

But just a note of warning: Some manufacturers of inexpensive, non-hifi equipment have jumped on the bandwagon by quoting "Peak music power" — obtaining the figure by using the peak rather than the RMS value of the output wave in the above formula, thereby doubling the answer. Not only that, but they add the output of the left and right stereo channels to give them "Total peak music power".

As a result, a humble 5+5W (continuous) amplifier, can become 6+6W music power, or 12+12W peak music power or an impressive sounding 24W total peak music power! So don't be misled.