Do your ears deceive you?

- **Dr. TOM FARRIMOND**
  reveals some startling facts on audio accuracy, distortion and measurement (1980).

Introduction

Here I intend to examine a possible role played by harmonic distortion in determining, what's generally described as 'amplifier accuracy', and suggest a method of establishing a relationship between the objective characteristics of an amplifier and its ability to reproduce sound resembling the original, that is, its accuracy when using a live standard for comparison.
I'll also be suggesting - rather contentiously - that an amplifier with zero harmonic distortion may not necessarily be as effective in conveying accurate detail to the ear as an amplifier which contains a certain amount of distortion of a specific type.

Some evidence in support of this suggestion is provided. The ear is apparently distortion-deaf to certain harmonic series, but although it may not hear a particular type of distortion, it nevertheless derives a benefit from it in another way. The distortion may be instrumental in providing better acceleration and deceleration of the basilar membrane in the ear, so enabling the small variations in harmonic content of music to be more accurately revealed.
It is argued that the distortion-pattern generated by the amplifier should match, or he congruent with, that produced by the ear. A congruent distortion pattern appears to be a more natural by-product of the valve than the transistor, which perhaps may account for the reported preferences in favour of the former, particularly with regard to musical detailing and depth.

Amplifier specifications

Amplifier designers over the past decade have in general eschewed the lowly valve in favour of solid-state devices, yet it is transistorised amplifiers which have recently been criticised as reproducing the sound of music less realistically than their thermionic brethren. There is little to be gained from simply stating that one type of amplifier is better than another. What is required is that the differences be accurately measured and their causes unambiguously determined. It has become clear that in some respects, objective tests of amplifier performance carried out without regard to the subjective effect upon the listener, may be of limited value. Just as it is equally true, that casually conducted listening tests can be misleading.
There is little point, except perhaps for advertising purposes, in specifying lower and lower amounts of distortions of type X, if there are also present factors Y and Z which have the greater effect upon the sound. The assumption is often made, that if a meter reacts more favourably to amplifier A than to amplifier B, then so should the ear. However this may not necessarily follow.

Recently, greater attention has been paid to factors in addition to absolute measures of distortion, which may have produced some of the audible differences between amplifiers which have been reported. These include for example: measurements of slew rate; transient intermodulation distortion and feedback (or the lack of it); peak output available before the protection circuitry is activated; ability to drive loads at different phase angles (rather than a simple load resistor); and latterly, the amount, type and relative levels of the distortion produced.

The problem in amplifier design, has been to determine which variables have the greatest influence on sound quality. There is no direct or objective way of knowing this; it must be established by noting the responses of listeners under controlled conditions. Otherwise it is possible to perpetuate the situation in which objective improvements are made which contribute no similar subjective improvement. The degree of accuracy with reference to the live sound is the only safe criterion which can be used as a measure of success.
Side-by-side comparisons of amplifiers without reference to some standard such as the original sound are

also fraught with difficulties and are only of value when the listener is familiar with the sound of live music and so has a fairly reliable built-in standard, in the form of auditory memory, for making a judgement. Otherwise an inexperienced listener may select amplifier A as preferable, because it lacks bass control, which may add a further sound dimension.
In comparison. Amplifier B may be thought less desirable since it apparently lacks this quality of bass emphasis. Under such conditions preferences are not necessarily a guide to accuracy. It is probable that test using preferences alone, without reference to some standard, have brought about the current criticisms of listening tests.

Live versus reproduced

The most satisfactory type of test would be one which employs an on-the-spot criterion to which reference can be made. For example, a live standard could be used like a musical instrument, which if activated mechanically, would provide an output which is easily repeatable. A paper-roll piano is one example which might well serve as an excellent standard - at least for that particular type of music!

A possible layout for amplifier-testing using a live standard could make use of a small revolving stage divided diametrically by a partition into two semicircles ‘A’ and ‘B’. The piano would be situated on one half ‘A’; the amplifier and speaker system on half ‘B’. The audience would be seated in an auditorium with good acoustics. Half the circular stage at any one time would project into the auditorium. The other half would be situated behind a sound-attenuating dividing-wall, so that no sound from behind the dividing wall would be heard (Fig.1).

In practice. complete sound insulation would not be necessary as long as there was sufficient attenuation to prevent interference with the sound in the auditorium. The acoustic treatment of the room at the rear should be such as to prevent room characteristics from being picked up by the microphone, since only the direct acoustic output from the piano is required. In the live condition ‘A’, the half of the stage with the piano would be in the auditorium. When listening tests have been carried out, the stage is then rotated through 180 degrees to carry the piano behind the dividing wall. Simultaneously, of course. the amplifier and speaker system which are mounted on the ‘B’ side, would now be located in the auditorium. The sound from the piano is picked up by a microphone heard via the test amplifier and speakers. The listeners’ task is to compare the sound under conditions ‘A’ and ‘B’. A group of experienced listeners would first hear the ‘live’ music, which could be selected as far as possible to cover wide dynamic and frequency ranges. Also, by means of a closed loop paper-roll, the musical samples could be kept reasonably short. The group would then hear the same music, but this time reproduced through the microphone, amplifier and speaker system.

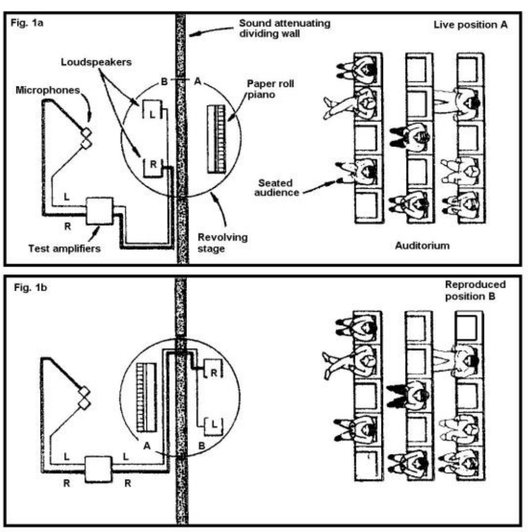

Fig.1 Live vs. reproduced test-auditorium

Accuracy rating

After some initial practice with the range of amplifiers to be tested, they would then be able to give an accuracy rating to each amplifier which could be by means of a scale ranging from one to ten, where ten is the value reserved for absolute accuracy, that desirable state in which it is not possible, under the test conditions, to perceive any difference between the sound of the live piano and the sound when reproduced via an amplifier. This would be subsequently confirmed by blind auditions where, again after practice to set the sound of the standard clearly in the minds of the listeners, the reproduced and live sounds would be presented in random sequence.

Inability of the listeners to discriminate between the two conditions would be the desirable goal. The most 'accurate' amplifier would retained as a secondary standard and used when testing a new crop. The number of variations which can be made to such a system is legion, for example the effect of using different speaker systems could be investigated. This may alter the rating for a particular amplifier, but at least the results would be meaningful and applicable to a definable set of conditions. It may be found that two different amplifiers are chosen as equally good in the subjective sense, even though their objective parameters are different. This would imply that, currently amplifier design is a process of balancing the various 'audio aberrations' in much the same way as the design and construction of a photographic lens.

Data analysis

The next phase would be to measure the objective variables of each amplifier and to correlate them with the subjective estimates of accuracy. Analysis of the data would enable a multiple regression or 'predictive' equation[1] to be constructed. This could then be used to predict subjective auditoria accuracy in terms of the objective parameters of the amplifiers, such as distortion, slew-rate and so on. Most of the terms in the equation would necessarily be reserved for objective tests of performance such as distortion and slew rate, but other variables related to the listener could be included, for example age, number of years exposure to live music and music preferences.

A possible equation is as follows:

$$\mathbf{D = 0.3X + 0.2T + 0.25S + 0.1H_2 + 0.3H_3 + 0.4H_5 - A + K}$$

- *Where:*

D is the subjective rating of quality or accuracy of the amplifier,
X is the percentage of crossover distortion,
T is the percentage amount of transient intermodulation distortion,
S is the slew rate in volts per microsecond,
$\mathbf{H_2, H_3, H_5}$ etc. are the percentage amounts of harmonic distortion at a variety of frequencies,
A is the age of the listener and
K is a constant for that particular equation.

To arrive at such a predictive equation, a large number of amplifier measurement and listening tests must be carried out. An equation is then constructed to give the most accurate prediction of the subjective result using only the measured parameters. The process of carrying out the tests required in order to formulate a predictive equation for amplifiers, although complex initially, would make subsequent evaluation of auditory accuracy much easier. (Where as stated, auditory accuracy is the ability of items of audio equipment to produce a sound which resembles the original.)

The method described above is about as objective as it seems possible to get in the area of the subjective evaluation of performance. As in most evaluations of audio equipment, the listening should be carried out over a long time period so that judgements are reliable. Objectionable qualities tend to become fatiguing with time and differences not immediately apparent become obvious later. It goes without saying that the listener should also care about music!

Amplifier differences

This leads to a consideration of the reported differences in performance between transistor and valve amplifiers. It seems surprising that a link in an audio chain with such low distortion content as an amplifier should assume any significance, since it is combined with the other components in the system in which distortion levels are more than ten times greater. But if one amplifier does produce a difference in sound quality as compared with another, under the same conditions, then the reasons for the differences must be examined. It was suggested recently (Farrimond, 1978)[2] that an amplifier which generated predominantly closely-spaced even harmonics may, because of masking effects on less palatable harmonics, sound better than an amplifier with a lower distortion content. Gordon King (1978)[3] in a series of tests found that there was a strong preference, when using recorded material, for an amplifier after its level of distortion had been increased from .03 per cent to 0.3 percent by the injection of even-order harmonics. One reason seems to be because objectionable odd- numbered harmonics in the signal from the record are masked, so producing a more pleasing effect.

Distortion deafness Apart from masking effects. there appear to be other benefits for the ear from the presence of residual distortion. It seems possible that an amplifier with a specific type of distortion, may provide more accurate subjective reproduction than another amplifier having a smaller distortion content but of a different type*, or even one with no harmonic distortion at all*. This possibility arises since the ear appears to be able to suppress the sound of a range of harmonics if they conform to a specific pattern: it responds only to the fundamental associated with that harmonic range. There is evidence in support of this theory. For example, the fundamental of a low note played very softly on a piano may no longer be audible because the ear is relatively insensitive to low frequencies. But even though the fundamental has disappeared, the sound remains the same. The ear apparently uses the remaining harmonic skeleton to reconstruct the sound of the missing fundamental. Experiments by Patterson (1969)[4] also show that the ear may construct a note, which is not actually present, from a series of different note, but appears to be due to the nerves in the ear. Patterson combined signals with the following frequencies: 1kHz, 1.2kHz, 1.4kHz, 1.6kHz, 1.8kHz. 2kHz and 2.2kHz. The listeners reported hearing a tone of 200Hz, which is not one of the original frequencies. There is additional evidence that a range of harmonics with specific relative sizes and frequencies, may not be heard as separate signals but only as a pure note, Békésy (1960)[5]. For example, if a pure-tone of 200 Hz is presented to the ear at a sound pressure level of 94 dB, harmonics are produced within the ear at 400Hz, 600Hz, 800Hz, 1kHz and 1.2kHz with magnitudes of 84dB, 76dB, 70dB, 66dB, and 60dB. This represents a roll-off of around 12dB per octave. The distortion levels are thus extremely high, corresponding to 32, 12.6, 6.3, four and two per cent respectively. (Fig.2).

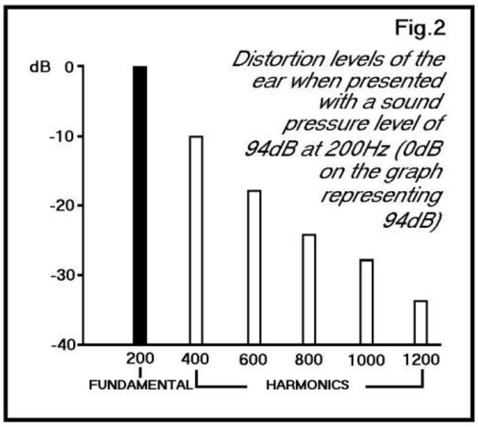

Fig.2 Distortion levels in the ear

- The surprising feature is that the brain perceives this harmonic complex as a distortion-free input. Only the pure-tone fundamental of 200 Hertz is heard. Because the ear is apparently distortion-deaf to this specific range of harmonics, it may also be deaf to a similar distortion spectrum generated by an amplifier. This distortion pattern produced by the ear is similar to that of some valve amplifiers, which may possibly explain why these amplifiers have received favourable reports of subjective accuracy, despite their objective measurements, The term coined by the writer to describe a harmonic pattern which is similar to that produced in the ear is ‘congruent’. This implies that there is a geometrical similarity of form. It would seem to follow, that any unit in an audio chain. (for instance cartridge, speaker etc.) which generates congruent patterns of harmonic distortion may receive the same treatment by the ear. If otherwise equivalent amplifiers were to be arranged in order of merit in terms of their accuracy in the subjective sense, it is quite likely that for low distortion signal-inputs, the sequence may go as follows: First place: amplifiers producing congruent harmonics. Second place: amplifiers with predominantly non-congruent even-order harmonic distortion. Third place: amplifiers with predominantly odd-numbered harmonic distortion. In the case of recorded material, which embodies relatively large amounts of non-congruent distortion, the positions of amplifiers in the second and third places may have to be exchanged. For example, an amplifier may sound more accurate in terms of the original music, if its distortion masks the non-musical distortions already present in the record. In this case, the effect is one which aphonic distortions (that is, music-compatible distortions) mask non-euphonic distortions. It would, be expected that cultural factors would dictate the positioning of amplifiers in places two and three.

Distortion paradoxes

The pattern of distribution of harmonics in an amplifier has been suggested as important in determining ‘musicality’ by Jean Hiraga (1977)[6]. An extension of this thinking to a concept of congruency in harmonics, may explain some of the paradoxes which have appeared in the literature. Gordon King (1977)[7] pointed out that a gramophone record may contain in excess of ten percent harmonic distortion. He also found in the tests reported earlier, that the addition of 0.3 per cent even-numbered harmonic distortion produced a more acceptable effect when listening to gramophone records. It is surprising that the injection of a mere 0.3 per cent distortion should produce any effect on the much larger amounts already present in the signal. The fact that it does, means that even small amounts of amplifier distortion are significant in determining sound quality! It seems that the amplifier’s congruent distortion-content, even though small, may assist in maintaining signal-integrity rather than competing with it. Just how, this may be accomplished is obviously of interest and requires elaboration. It would be expected that amplifiers in which the harmonic pattern remains congruent as the output is varied, would provide maximum auditory benefit, whereas amplifiers in which there is a change in the distribution of harmonics as output changes, would produce different auditory characteristics at each output level.

Lost detail

- Although in the original performance, orchestral music has a wide dynamic range, when recording takes place detail may be lost in several ways. The record may be unable to encompass the dynamic range so that some compromise may have to be made in the recording process involving loss of low-level detail. Noise levels also tend to increase which further decreases detail and in the process of playing back the recording, levels tend to be lower than at the original performance so that small harmonics are pushed further towards the level of in-audibility. An amplifier producing a congruent distortion pattern may tend to make possible the survival of low level harmonics so maintaining some of the characteristics of the original performance and resulting in maximum detail and sense of depth. One of the characteristics of a live performance, when the listener is close to the performers, is that there is a wealth of detail. Small amplitude harmonics are easily audible and the effect produced is one of richness and warmth. The harmonic structure associated with each sound helps it to maintain its individuality. Lack of harmonic content (as in an orchestra heard in the distance) results in lost spatial separation between instruments as well as a loss of warmth and timbre. The experience of hearing music is a temporal one in which the patterns are distributed over relatively long periods of time. In contrast the visual experience, as in looking at a painting, is more immediate and a large proportion of the information is obtained very quickly. The problem with the resolution of detail by ear and brain is that instantaneous intake of detail is restricted because of the tendency of different sounds to be amalgamated by the ear. A short burst of sound heard from a record, as when the stylus makes a fleeting contact results in a complex sound from which the resolution of different instruments by the listener is extremely difficult. At any one time signals in the ear may be in competition for the same place on the basilar membrane so that the harmonics from several instruments played in unison are difficult to disentangle. The suggested improvement in detailing, due to congruent harmonics is probably related to the physical properties of the ear. As with most mechanical systems, the various structures of the ear, from the eardrum to the basilar membrane, have mass and to bring about any change in the way in which vibration is occurring, must involve the application of a force and the expenditure of energy. Each new signal must modify the vibrations already occurring from earlier signals and this may require that the original vibrations be damped before the new mode of vibration can be established. Even though the basilar membrane is vibrating in a liquid medium, its natural damping is not instantaneous so that the application of a fresh signal may not be immediate.

Vibrations in the ear

For example, if a pure-tone of 1.1 kHz is fed into the ear immediately after the presentation of a 1.15 kHz tone the basilar membrane must have any residual vibrations at 1.15 kHz damped and a new vibration of 1.1 kHz, instituted at a slightly different, position on the membrane. The amplitude of the wave may also be required to increase or decrease depending upon the relative magnitude of the two-signals. Two changes are therefore required involving position and amplitude before the second signal can become effective. It is probable that the greater the energy level of the second signal, the more rapidly the physical state of the membrane can be changed. That is, there is an increase in the amount of detail per unit time which a particular portion of the basilar membrane can handle. If the second signal is pure (that is no harmonics) then its site of action is limited to one point on the basilar membrane depending on its frequency. Therefore the change of vibrational state is slow. If, however, the signal includes a range of congruent harmonics, then damping and restimulation of the basilar membrane would take place over a wider area, rather than at just one point, and the effect would be much more rapid since additional energy is involved in the establishment of control. Small signal fluctuations could therefore be registered by the ear much more rapidly, so providing an increase in detail. In addition, if there is a similar effect upon the loudspeaker diaphragm, then a congruent harmonic pattern is twice blest: once at the speaker and again in the ear. If this is true then the best (i.e. the most accurate) amplifier may not be a ‘straight wire with gain’ but rather ‘a bent wire with gain’, where some distortion of the right type is included! But before any firm conclusion can be drawn, the appropriate live-versus-recorded comparisons need to be carried out as stated earlier.

Valves versus transistors

Some personal observations may be appropriate concerning experiences with valve and transistor amplifiers. Recently comparisons were made between two modern amplifiers, one valve (Fig.3) the other transistor . The aim was to determine which one produced the characteristics of live music more accurately. In this case the standard involved recent memory and was provided inadvertently by that excellent body The Salvation Army. They were chosen (in addition to other live sources of sound) since they are to be found performing in the open air, thus ensuring that the acoustics of the ‘auditorium’ are fairly constant: also they can be approached closely so that there is no loss of detail. The characteristics of wind instruments heard this way en masse are of roundness, warmth, detail and depth. There is no trace of hardness and the sound is easy to listen to without fatigue for extended periods. High loudness levels do not produce any sense of strain.

Fig.3 Michaelson & Austin TVA-1

The valve amplifier used for comparison purposes was the **Michaelson and Austin TVA-1** produced by Soundlease Services and reviewed in Practical Hi-Fi by Dave Berriman (1978)[8]. The transistor amplifier was a highly-rated American model with an advanced specification. Power outputs of the two were reasonably similar, but point for point, the objective data were superior for the solid state device. Specifically. slew-rate was three times faster, signal to noise ratio was better by 15dB, total harmonic distortion was roughly fifty times better (that is 0.005 per cent as compared with 0.2 per cent). In spite of this, the valved TVA-1 produced a subjective effect which was judged to be more like that of the real thing in terms of the qualities described earlier. The evaluation in favour of this particular valve amplifier, into which the writer incorporated some of the component variations suggested by Chris Rogers (1977)[9], persisted for direct-cut discs as well as conventially-recorded material. Piano tone in the Dave Grusin 'Discovered Again!' recording by Sheffield Lab. was adjudged to be more like the real instrument. Fine details audible via the valve amplifier were less so via the transistor amplifier. Another characteristic possibly related to the above is the capacity of 'attention getting', so that in becoming absorbed by the music one forgets the equipment and the task of criticism at hand. This was regarded as one of the major differences between the two amplifiers and is perhaps the most significant difference between them. In other respects the transistor amplifier was excellent and it required careful listening to differentiate between the two, even using analytical speakers such as stacked Quad electrostatics (Farrimond, 1975)[10]. The above-mentioned difference is difficult to define, since in contrast with qualities such as 'brightness', bass-response, etc. it appeared to be a general characteristic permeating the whole of the audio spectrum. It is a characteristic which live music has and which is less evident in the transistor amplifiers which have been heard, as compared to the TVA-1. Although the above subjective evaluation is open to the criticism which has already been made with regard to the use of a remembered criterion of accuracy, attempts were made to relate the auditions to the remembered qualities of live music derived from recent experiences. Also, the same conclusions were reached by other listeners, again using the same criteria and without knowing which amplifier they were hearing, so as to avoid prejudice. What is required ideally, to clarify the issue, is that further tests be carried out along the lines of Gordon King's investigation, but using a live objective standard to which the listener could make references and again using an amplifier to which 0.3 per cent congruent distortion could be added. Only then would it be known if the addition of 'ear-congruent' harmonics resulted in an increase in accuracy as well as producing a 'preferred' sound.

Pure tones probe

In order to probe a little deeper into the underlying reasons for the acoustic differences reported. I intend as a first step, to carry out a series of tests using pure tones obtained from a low-distortion signal generator. The tones would be recorded and heard under two conditions: the first in which nothing has been added, the second condition in which a range of ear-congruent harmonics around the 0.3 per cent level has been added to the pure tone. Listeners would be asked to select which of the two recorded tones sounded more like the reference-tone straight from the signal generator. It would certain be interesting if the opinion was that the distorted signal sounded 'purer'! Perhaps this may not apply in the case of steady pure tones in which there is no harmonic variation as found in complex signals. It may be necessary to use rapid frequency modulation of the sine wave, or alternatively, complex material such as music, to determine whether congruent harmonics confer any beneficial effect or not, in terms of increased resolution of detail.

Summing up

In summary, it seems at least possible from experimental evidence, that perhaps totally eliminating harmonic-content from an amplifier may be counter-productive in terms of creating or preserving the auditory illusion of live music. The presence of congruent harmonics, as defined, may assist the ear to differentiate more readily between small signals in complex material. A viewer in an art gallery would not be asked to view under monochromatic lighting on the grounds that a single wavelength is pure relative to white light, which is a mixture of many different wave-lengths. What happens to the colours well combined, in terms of what the eye perceives, may be similar to the combination of certain harmonics in the ear. Under some circumstances, the harmonic pattern may produce a different and unpredictable, subjective effect - an effect which could not be forecast by meter measurements of the input.

The interaction of sound with the listener's 'receiving apparatus' and the requirement of audio equipment to duplicate such interaction in the cause of accuracy, suggests that amplifiers of the future may be engineered with specific qualities rather than as neutral or passive components whose very state of perfection, in the physical sense, may hinder experimentation. Possibly a good valve amplifier has natural characteristics which make it relatively 'ear-compatible' as compared with, other types into which compatibility must be engineered. The answer to many of these questions must await further experimentation.

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