A new distortion measurement

Better subjective-objective correlation than given by t.h.d.

by R. A. Belcher, B. Sc., Ph.D., M.I.E.E., BBC Research Department

This article describes a new technique for measuring non-linearity distortion which gives much better correlation with subjective assessment of sound quality than does the conventional total harmonic distortion measurement. Known as the double comb-filter method, it uses pseudo-random binary test signals and largely digital processing, so the cost of instrumenting it can be expected to fall with the increasing availability of l.s.i. circuits. Ultimately, the hardware may cost less than that used at present for t.h.d. measurement. Ways are suggested for using the new technique to measure cross-over distortion, transient intermodulation distortion and other parameters such as "wow" and "flutter" and linear distortions. The BBC is now testing it on sound-signal transmission circuits and studio equipment.

RECENTLY THE HI-FI PRESS has been particularly interested in discussions about the best way of relating objective measurements to the subjective assessment of non-linearity in audio systems. Several articles have dealt with the relative merits of total harmonic distortion, two-tone intermodulation, or band-limited noise tests. The main conclusion seems to be that there is a lack of hard facts about the degree of correlation which can be expected between the measurement of distortion and the subjective effect.

Over the past decade broadcasting organisations have become increasingly interested in this problem. In particular, the BBC has been studying the use of noise-like test signals. As a result of five years' research work, a test method has been developed which gives distortion figures that correlate much better than total harmonic distortion measurements with subjective estimates of sound programme quality. The new test method uses pseudo-random noise as the test signal and comb-filter techniques to separate the distortion products from the test signal. Equipment for this method should be fairly cheap to produce as it makes use of digital signal processing.

To assess more fully the possible applications of the new method, the BBC is conducting field trials with experimental equipment, and the results of these trials should be available during 1978.

Traditionally, the non-linear distortion of a sound signal circuit has been measured objectively by using a sine-wave test signal; the amount of non-linearity is conveniently expressed as the ratio, measured at the output of the. test circuit, of the power level of the harmonic distortion products to the power level of the fundamental plus harmonics. This ratio is known as the total harmonic distortion and is sometimes quoted as a percentage; in this article it will be given as a decibel ratio.

It has long been recognised that total harmonic distortion measurements generally do not give a good indication of the subjective impairment due to non-linearity, and in 1950, in an effort to provide improved subjective agreement, Shorter1 proposed that the measurement of distortion should be weighted to take more account of high-order harmonics. In 1960 Wigan[2] proposed an improved weighting criterion to be applied to the harmonics of a 1kHz test signal. His subjective data was obtained using a pulsed tone as the programme signal, and the non-linear distortion was produced by an arrangement which ensured that the amount of total harmonic distortion was substantially independent of the applied signal level.

Wigan's findings were of limited application as no way was suggested by which his weighting criterion could predict the unpleasantness of distorted programme (other than pulsed-tone) signals. Further, for experimental convenience he had also excluded circuits in which the amount of distortion was a function of signal frequency and applied signal level.

When a complex programme signal suffers non-linear distortion, unwanted signals arise not only by the generation of harmonics but also by intermodulation between the spectral components of the programme signal. In 1945, Brockbank and Wass published a mathematical analysis[3] of the intermodulation spectrum generated by a multi-frequency programme signal. Their objective was to enable the distortion spectrum of a multi-frequency input signal, e.g. a programme signal, to be predicted from knowledge of the distortion spectrum generated by a single-frequency input signal.

There are two accepted methods of measuring the amount of intermodulation distortion of an audio frequency circuit. That adopted by the Society of Motion Picture and Television Engineers (SMPTE) was proposed by Hilliard[4] in 1946, and the method adopted by the International Telephonic Consultative Committee (CCIF) was proposed by Scott5 in 1945. Both use two simultaneously applied sine-waves as a test signal, but they have different frequency spacings. The Hilliard method uses 60Hz and 3kHz, and the Scott method 3kHz and 3.05kHz.

To obtain a more complete knowledge of the variation of non-linear distortion with frequency it would be necessary to make measurements of intermodulation distortion over the whole frequency range of interest. This type of intermodulation test has been found to be of use by Harwood[6] in research on improving the performance of loudspeaker units.

Unfortunately, no method has yet been devised to enable intermodulation tests to predict the unpleasantness of distorted programme signals.

One interesting result given by Brockbank and Wass is that if the programme signal is assumed to be represented by n tones, each of equal power, and if n is greater than 30, then the contribution to the total distortion power due to harmonic products is at least two orders of magnitude less than that due to intermodulation products. For complex signals such as those produced by speech and music, n is generally large enough for the distortion power level contributed by harmonic products to be negligible. They also derive an equation for the total distortion power, assuming that n is large enough for harmonic products to be ignored, which shows that the power of the distortion signal resides mainly in products generated by higher order terms in the power series. This emphasis on higher order distortion products is effectively what Wigan and Shorter recommend for improved subjective agreement.

Random noise tests

The work of Brockbank and Wass, and Shorter and Wigan suggests that a noise signal should be a good test signal for sound-signal non-linearity measurements, since n would then be very large, and the power level of the distortion signal would automatically be weighted in favour of higher order terms.

Noise-like test signals have been applied to the routine non-linearity testing of various low-quality sound-signal transmission systems, e.g. multi-channel communication systems[7] and p.c.m. codes for speech signals[8]. The popularity of this type of test stems from the fact that the statistical properties of a speech signal have been treated as Gaussian, and therefore the unwanted noise power spectrum generated by a noise-like test signal was considered to be a good estimate of that produced by speech signals.

In 1963, Danes[9] reported the use of a one-third octave band of noise centred on 8kHz for intermodulation testing in an f.m. sound broadcasting system, and he found some degree of correlation between the measured low-frequency noise power generated from the test signal and the subjective impairment of normal programme signals.

In 1970, a standard test signal was proposed by the CCIR for study[10] by various member broadcasting organisations; it was to be used in measuring crosstalk and non-linearity in high-quality sound-signal transmission circuits. This test signal was to be produced by shaping the spectrum of a random noise signal, to make it representative of an average programme signal.

In 1971, Nikaio and Nitatori [11], of the NHK reported their study of non-linearity measurements on sound broadcasting circuits using random noise as a test signal. Their test signal was similar to that proposed by the CCIR, but it had a 3/4 octave wide spectral gap which could be moved in 1/2 octave steps to cover most of the audio-frequency range. Distortion products were selected in this gap by a 1/4 octave-wide filter. The result of each test was a plot of distortion versus centre-frequency. Their work did not include a study of the agreement between the measured distortion and its subjective effect.

In the BBC, tests for non-linearity are routinely made by measuring the total harmonic distortion of a 1kHz tone. This signal is applied to a test circuit at a level 2dB higher than the nominal quasi-peak value of the programme signal. With the advent of f.d.m. transmission circuits routine tests using this high level signal were reported to cause undesirable levels of crosstalk interference. As the use of f.d.m. circuits was likely to increase, the BBC undertook to assess the standard test signal proposed by the CCIR in the hope that crosstalk problems would be less severe (this was confirmed by later work), and also that routine tests for non-linearity using this test signal would give at least as much information as was provided by the total harmonic distortion method.

In the form proposed by the CCIR, the standard test signal had no spectral gap in which to detect non-linearity products and was therefore only suited to crosstalk measurements. The BBC therefore proposed a variant of the test signal, having a spectral gap between 3.5kHz and the upper limit of the audio band, and conducted informal tests to study its use in routine tests for non-linearity. The results showed that the degree of correlation between r.m.s. distortion measurements using this test signal and subjective assessments of the threshold of impairment of a programme item was at least as good as was obtained using a routine total harmonic distortion test, and was probably better.

Pseudo-random noise tests

Later measurements of this type were helped by replacing the random (thermal) noise source with a digitally-generated pseudo-random binary sequence (p.r.b.s.) noise source. This type of source provided a power output which did not vary with temperature; moreover, as the amplitude of distortion products fluctuated regularly at the repetition rate of the pseudo-random noise, a meter with a short integration time (a BBC peak-programme meter [12]) could be employed. This enabled accurate measurements to be made more quickly and allowed the use of the meter which is routinely used within the BBC for circuit testing.

Then various band-splitting arrangements were considered with the aim of increasing the effective bandwidth of the test signal. First, a standard set of 1/3 octave filters was used but was later rejected because their rate of cut-off was insufficient to provide the required measurement resolution. It was then decided to construct a two-band arrangement using specially designed filters to provide high resolutions (approximately 80dB of noise separation) with noise bands of 150Hz to 700Hz, and 1kHz to 3.5kHz approximately.

Concurrently a formal subjective investigation provided data relating to a wider range of programme impairment and more critical listening material than had been used previously. These data related the operating points of four amplifiers to subjective impairment and could therefore be used at a later date to study the subjective agreement obtained with any objective method of non-linearity measurement, simply by making objective measurements on those amplifiers.

Next came the task of optimising various parameter values for the two-band method, using the new subjective data. This work is described more fully in a BBC Research Department report [13]. Work was completed in mid-1973 and it was decided that the two-band method could give estimates of subjective impairment that were much more accurate than those given by the routine total harmonic distortion method. The two-band method was, however, of limited use as it was not designed for testing circuits where non-linearity was pronounced at high signal frequencies.

In 1972, Nikaido[14] reported his use of a sound-programme signal in making measurements of non-linearity. (Since a spectral gap had to be provided, the signal was not suitable for simultaneous broadcasting.) A disadvantage of this approach is that the measurements have to be integrated over a long time interval, as the results are influenced by the changing spectrum of the programme signals. The alternative would be to standardise on particular sound-programme excerpts as test signals but this has the disadvantage that the resolution of the test apparatus would be limited by the distortion in the sound recording system. The practical requirement is for a method which uses a test signal which is accurately reproducible, which simulates a sound programme signal, and in addition is inexpensive to produce. For these reasons, the practice of using the sound-programme signal as a test signal was, not considered by the BBC to be appropriate for routine tests.

Meanwhile methods by which band-splitting might be improved were being considered. Multi-band techniques were investigated, but were not developed because of the difficulties of making high resolution comb-filters to provide a comb-like test signal spectrum with many "teeth", and a measurement filter with a complementary comb-like response. Furthermore, this approach would be prohibitively expensive.

Comb-filter methods

In 1974, I proposed an alternative approach to the problem. This new proposal recognised that the pseudo-random noise signal had itself a comb-like spectrum (see Fig. l(a)). If this noise signal were to be frequency-shifted, its components would be anharmonically related, and most of the distortion products would then fall in the gaps between them as shown at (c). Non-linearity products would be measured by applying an equal and opposite frequency shift (d), followed by a comb-response filter (e) to remove the comb spectrum of the noise signal. The output of the comb-filter (f) would then be registered on a p.p.m.

Apparatus was constructed to test this proposal[15]. The work was completed by mid-1975, and a comparison of the accuracies of this comb-filter method (the "single comb-filter method"), the two-band method, and a total harmonic distortion method revealed that both noise-separation methods were more accurate in their estimates of subjective impairment than the total harmonic distortion method. But the single comb-filter method was to be preferred since it could be used in testing circuits with frequency-dependent non-linearity.

To improve the method a double-comb-filter technique was proposed in which two p.r.b.s. signals were combined to provide a test signal with anharmonic components and two cascaded comb-filters were used to reject the test signal and accept the distortion signal for measurement. This arrangement[16] is attractive as it should be less expensive to instrument and, furthermore, it has been found to be as accurate as the single comb-filter method using the subjective data given in reference 13. This accuracy also applies to frequency-dependent non-Iinearities studied more recently.

Test signal generator

The test signal used with the double comb-filter method is produced by the addition of two signals, each having a harmonic spectrum but with different fundamental frequencies. The harmonic structure of each spectrum may be likened to a "comb" of frequencies. Intermodulation products generated from this test signal may be conveniently divided into two groups: those which arise by intermodulation within each comb structure and those which arise by intermodulation between comb structures. The first group of products will ocupy the same frequency positions as the original comb structure and will therefore be indistinguishable from it. The second group of products will fall in the spectral gaps of the test signal and can therefore be measured. Fortunately, these measurable products represent a significant proportion of the total distortion signal. Two comb-filters having attenuation "teeth" aligned with the "teeth" of the test signal can be used to remove the original components and enable the remaining distortion products to be measured.

The test signal can conveniently be generated by two maximal length pseudo-random binary sequence (p.r.b.s.) generators (m-sequences). Such binary signals can be generated by digital shift-registers with feedback. If an n-bit shift register is used to generate the p.r.b.s. signal then the sequence length is *2n-1*. The spectrum of a p.r.b.s. signal consists of harmonics of the sequence repetition frequency and the harmonics are all substantially of equal power, as illustrated in Fig. 2(a) and (b), provided that the required test signal bandwidth is less than one tenth of the shift-register clock frequency. The repetition frequency of an m-sequence is given by the clock frequency divided by the sequence length.

When the two p.r.b.s. signals are added, the test signal spectrum illustrated in Fig. 2(c) is obtained. In practice this test signal is used at the same "peak" level measured with a p.p.m. as the "peak" level of the programme signal which it replaces. Fig. 2(d) shows the modified spectrum when distortion products are generated. A simple comb-filter response as in Fig. 2(e) is used to remove one p.r.b.s. signal, while a second filter (f) in tandem removes the other p.r.b.s. signal, leaving distortion products as illustrated in (g).

Fig. 3 is a simplified block diagram of the test-signal generator. G_1 and G_2 are m-sequence generators with repetition frequencies of 152.6Hz and I09.8Hz. These two frequencies were chosen by experiment. The first (152.6Hz) could be generated using a convenient shift-register and oscillator combination. The second frequency could be changed in steps of approximately 1Hz and within this limitation the second frequency was near optimum, i.e. that which gave best subjective/objective correlation. it should be noted that the more important parameter is the ratio of the two frequencies rather than their absolute values; altering this ratio alters the weighting of odd and even order products, and therefore also the subjective / objective correlation.

The scaling amplifier adds the two binary signals in a given ratio, the optimum value for the ratio being that which gives a test signal spectrum with all components of equal power.

In later tests, a spectrum shaping of the test signal was found to be important as it helped to reduce the spread in subjective impairment attributable to a given distortion figure measured by the new method, taking into account both flat and frequency-dependent nonlinear effects. A suitable spectrum shaping characteristic is shown in Fig. 4. It can be produced by connecting a CCIR average programme weighting network[18] in series with a 50μs de-emphasis network. The band-pass characteristic of the shaping filter converts the p.r.b.s. digital signal into one with a multi-level noise-like waveform. If a shaping filter of this type were not used then a low-pass filter would be necessary.

The above filter characteristic agrees closely with that proposed by the UK Post Office[19] as a result of recent measurements of the power loading and average spectra of sound programme signals for broadcasting. Their weighting characteristic is intended to be used to shape the spectrum of a white-noise signal in order to simulate that of an average sound-programme signal. Such a weighted noise signal could be of use in testing, for example, the' power handling ability of loudspeakers or audio power amplifiers. It therefore seems likely that the test signal required by the double comb-filter method could also be of use in these areas.

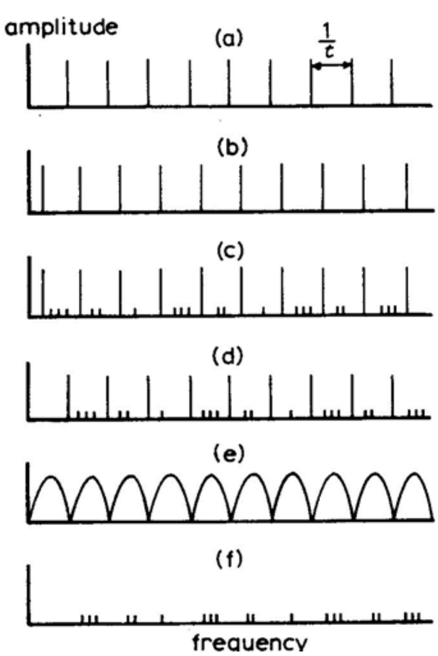


Fig. 1. Spectra illustrating principle of the single comb-filter distortion measurement method: (a) spectrum of pseudo-random binary sequence; (b) the same p.r.b.s. frequency shifted; (c) signal from system under test, showing distortion products in gaps; (d) signal re-shifted; (e) *response* of *comb-filter for removing comb spectrum of p.r.b.s. test signal; and (f) comb-filter output. is repetition period* of *p.r.b.s.)*

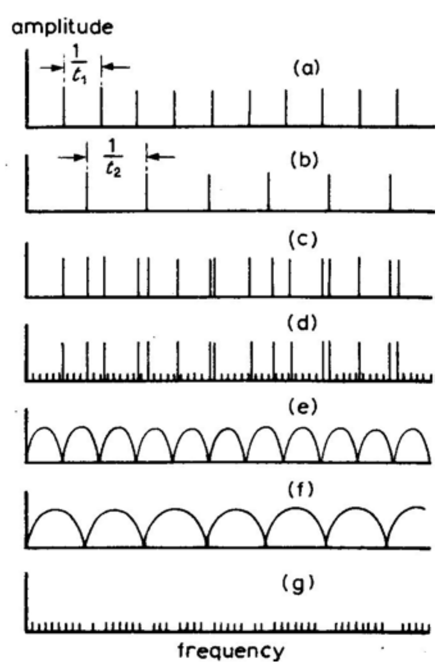


Fig. 2. Spectra illustrating principle of the double comb-filter method (t_1 and t_2 are repetition periods of m-sequences G_1 and G_2 respectively): (a) m-sequence G_1; (b) m-sequence G_2; (c) *test signal; (d) signal from system under test; (e) first comb-filter response; (f) second comb-filter response; and (g) output from comb-filters.*

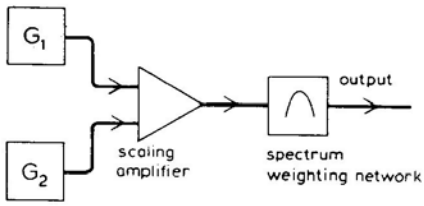


Fig. 3. *Essentials* of *test signal* generator for *double comb-filter method.*

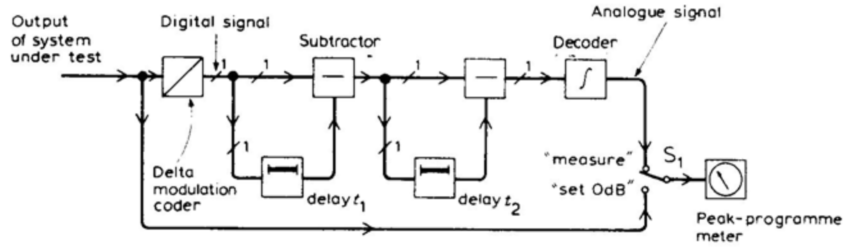


Fig. 5 *(above). Analyser for output signal from system under test in double comb-filter method. Levels* of *test signal and distortion producta are measured on* a *p.p.m.*

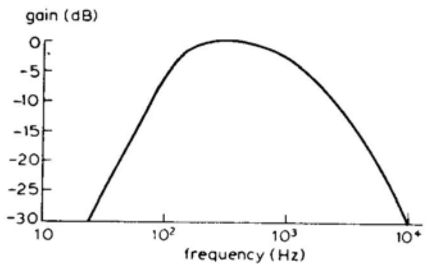


Fig. 4. *Test signal frequency shaping characteristic.*

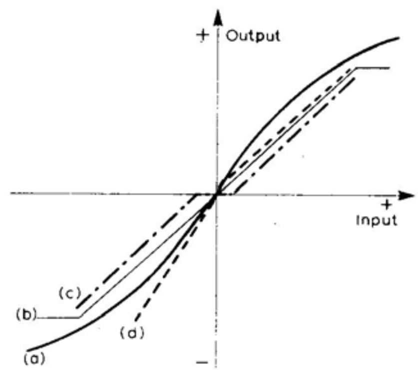


Fig. 6 *(right) Typical transfer characteristics* of *audio* circuits: *(a) S-type characteristic (b) op-amp type of characterist (c) circuit with cross-over distortion; and (d) t.h.d. constant with applied signal level.*

In the experiments with the double comb-filter method it was found to be advantageous to modify the quasi-peak to mean ratio of the test signal by including a dispersive network (four 700Hz single section all-pass circuits in tandem) after the weighting network. This modification allowed the optimum value for the mean test signal power loading to be reduced by 1dB and is of use in testing circuits, e.g. f.d.m. carrier circuits, where it is necessary to keep the mean test power loading to a minimum during routine tests.

Test signal analyser

Fig. 5 is a simplified block diagram of the test signal analyser. Digital signal processing is used because of its stability and relative cheapness in providing the required performance. For further economy in analogue-to-digital and digital-to-analogue conversion, linear delta-modulation coding[20] is used in preference to linear pulse-code modulation (p.c.m.). The quantisation noise of the system limits the measurable separation to 54dB: this is later shown to be 17dB below the threshold of impairment.

The input signal to the analyser is applied to the delta-modulation coder which provides a 1-bit output signal at a rate of 4Mbit/s. This signal is then processed by a digital filter arrangement comprising two subtractors and two shift-register delays t_1 and t_2. This filter is effectively two simple comb-filters in tandem: a comb-filter operates by subtracting from an input signal a delayed version of itself. In Fig. 5 t_1 and t_2 are produced by approximately 27kbit and 37kbit shift registers respectively; time delays t_1 and t_2 are equal to the inverses of the test signal repetition frequencies. The delta-modulation signal is then decoded by analogue integration of the binary output signal. Emitter-coupled logic circuits were used in the delta-modulation coder and decoder in order to produce pulse waveforms with well-matched rise and fall times. This matching is essential in order to maintain low harmonic distortion within the measuring equipment.

A crystal oscillator is used to ensure that the comb-filter responses are maintained in accurate alignment with the "teeth" of the test signal, but for clarity, clock signal paths are omitted from Fig. 5. The gain of the direct path is set so that, at a signal frequency at which minimum loss is produced by the comb-filters, the output level from the "measure" path is equal to that from the "set 0dB" path.

The levels of test signal and distortion products are measured using a BBC peak programme meter (p.p.m.), which is a quasi-peak indicating instrument. It should be noted that the results reported later for the double comb-filter method may not be directly applicable if other meters are used. However, a simple additional circuit could be provided to simulate the action of a p.p.m. and to enable a noise-separation indication to be independent of the ballistics of the measuring instrument.

Using the arrangement in Fig. 5, the switch S_1 is set to the "set 0dB" position and the output test-signal level from the system under test is adjusted until a reading of 0dB is indicated by the p.p.m. Noise-separation in dB is then indicated by the p.p.m. when SW_1 is set to "measure". This reading actually gives a measure of (signal + distortion)/(distortion); and although this is not a true separation figure it has proved to be satisfactory in use.

Listening tests

The purpose of the tests was to obtain subjective estimates of programme quality impairment caused by two circuits (A and B) for a selection of applied signal levels. The general form of the transfer characteristics of these two circuits is illustrated by curves (a) and (b) of Fig. 6. (a) is an "S-type" characteristic in which the total harmonic distortion increases with applied level. (b) is the sort of characteristic given by operational amplifiers, and this type of distortion is sometimes known as "hard-clipping"; the distortion is negligible until clipping occurs and then the total harmonic distortion increases rapidly with applied signal level.

Frequency-dependent distortions such as those that might occur in an f.m. transmission system were also included in the study. High-frequency signals were made more susceptible to non-linearity distortion by inserting a 50μs pre-emphasis network in the input signal path. When pre-emphasis was used, a network with the complementary characteristic was connected in the output signal path to provide an overall flat amplitude-frequency response.

These tests require great care in the alignment of tape recording and signal level measurement apparatus because in order to correlate objective measurements of non-linearity distortion accurately with subjective impressions, it is essential to ensure that the applied level of test signal bears a known relationship to the previously applied level of programme signal. This is particularly important when the amount of distortion varies rapidly with the applied signal level, e.g. when a hard-clipping circuit is operated near to overload.

For convenience both programme and test signal levels are referred to the operating point of the circuit under test by a parameter termed "relative gain". A more detailed explanation of this parameter is given in Reference 16.

The listening tests explored impairment levels in the range indicated by the six point scale shown in the following table:

Six-point subjective impairment scale

Grade	Impairment
1	Imperceptible
2	Just perceptible
3	Definitely perceptible but not disturbing
4	Somewhat objectionable
5	Definitely objectionable
6	Unusable

Two programme items, "male speech" and "solo piano" were selected from a library of recorded test-excerpts as they were found to be subjectively more sensitive to distortion in the test-circuits than the other excerpts. The male speech item was used in tests in which non-linearity was not frequency-dependent and the solo piano item was used for the frequency-dependent case.

A companding system was used in the tape recording and reproduction of the test-programme so that recording noise was less likely to mask the distortion produced by the test-circuits. In addition, the programme signals went through the record-replay process only once, i.e. a "master" recording was used for the tests. No amplitude-compression was applied to the programme signals.

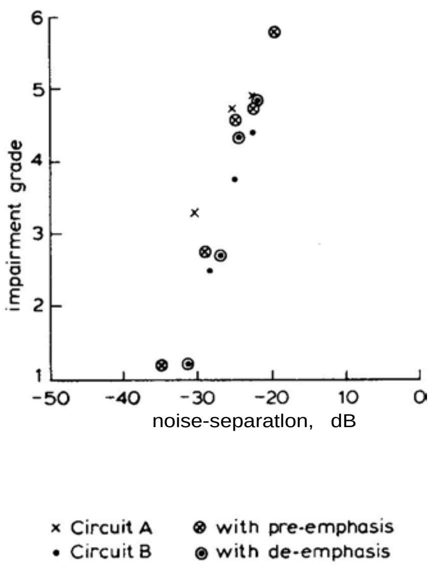
The subjective tests were carried out in a listening room having acoustics similar to those of a domestic living room.

Subjective-objective correlation

After the subjective tests had been completed, objective measurements were made using the two methods (double comb-filter method and total harmonic distortion method) whose correlations were to be compared. It was again essential to set the test levels with great care, and the "relative gain" parameter mentioned earlier was again used to ensure that the operating points of the circuits were the same as used in the subjective tests.

In order to study subjective-objective agreement plots of subjective impairment against objective distortion were drawn. In practice fairly straight but widely spaced, non-parallel lines can usually be obtained by a variety of distortion measurement methods. The best correlation is judged to be obtained by the test method which gives the closest grouping of plots along a straight line of subjective impairment against objective distortion for a variety of test circuits.

Fig. 7 *(below). Plotted points showing good correlation between subjective average impairment grade and measured noise-separation in the double comb-filter method, with two circuits A and B. Note that points are closely grouped when* compared *with those in* Fig. 8.



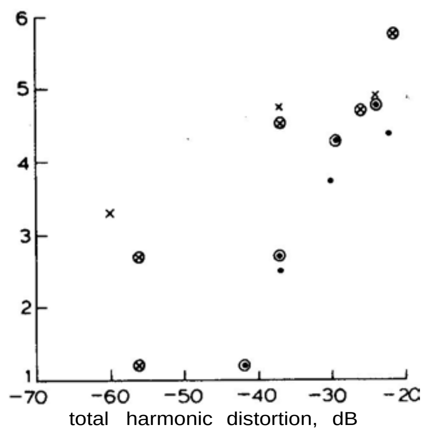


Fig. 8 *(above). Plotted points showing lower correlation between subjective average impairment grade and measured total harmonic distortion, for circuits A and B. Note the relatively wide spread of the points compared with those in* Fig. 7.

Fig. 7 shows the measured spread in noise-separation against subjective impairment for circuits A and B, both with and without 50µs pre- and de-emphasis. Similarly, Fig. 8 shows the measured spread for total harmonic distortion.

Taking grade 2.5 as a reference point for the comparison, Fig. 8 shows that the spread in total harmonic distortion is 23dB; it is apparent from the data shown that the routine total harmonic distortion method gives a very poor degree of correlation when more than one non-linear circuit is being considered.

Fig. 7. however, shows by the close grouping of plots (spread approximately 6dB at grade 2.5) that the noise-separation test gives much better correlation between subjective and objective measurement than does the total harmonic distortion method.

Possible future developments

An advantage of the total harmonic distortion method is that it employs standard laboratory items of equipment, e.g. a variable-frequency audio oscillator and a signal level meter, which may also be employed in the measurement of other sound circuit parameters such as the amplitude-frequency response.

Similar advantages apply to the double-comb filter test apparatus as it may also be used in the measurement of a wide range of sound signal distortions (both non-linear and linear), but before describing these applications in more detail, it will be useful to consider the various forms of non-linear distortion characteristics which can occur.

Fig. 6 illustrates some input-output transfer characteristics which may be exhibited by audio signal amplifiers. The type of characteristic applicable to a given amplifier can be conveniently identified by examining how the total harmonic distortion of a sine-wave signal varies with the applied signal levels. Curves (a) and (b) are the characteristics of circuits used in the listening tests described earlier. The types of non-linearity shown by curves (c) and (d) are described below, followed by a variant, type, (e). Formal listening tests have not yet been conducted for types (c) to (e), but some initial assessments have been made.

Curve (c) represents "cross-over" distortion, and in this case the total harmonic distortion increases as the applied signal level is reduced. Curve (d) represents the characteristic of a circuit where the total harmonic distortion is constant with applied signal level, and would be obtained with a circuit that gave unequal amplification to positive and negative half-cycles of a sine-wave.

Characteristics (a) to (d) have not included the possibility of the degree of non-linearity distortion being frequency-dependent as well as amplitude-dependent. This therefore is provided in type (e) which includes types (a) to (d) with, for example, pre- and de-emphasis networks to make the distortion more pronounced at high signal frequencies.

In addition, transient intermodulation distortion (t.i.d.) as described by Otala[21] can sometimes occur in audio circuits. Essentially, t.i.d. occurs when the rate-of-change of the input signal is excessive. This type of distortion is generally not measurable using a 1kHz total harmonic distortion test, and at present there is no accepted method for measuring t.i.d. and relating the objective measurement to the audible effect.

Experiments with the double comb-filter apparatus have shown that non-linearity types (c) to (e) and transient intermodulation distortion can readily be measured. The sensitivity of the method in measuring t.i.d. arises from the high slope of the test signal which can be equivalent to that of a full amplitude sine-wave signal at the upper limit of the audio band. Further more lengthy experiments would, however, be required in order to establish the subjective-objective correlation with these types of distortion. It is interesting to note that Levitt and others have determined a subjective threshhold of perceptibility for slope-overload distortion of speech signals[22], and this may be of help in studies of t.i.d.

The double comb-filter apparatus can also be used for objective measurement of amplitude-frequency response errors, and "wow" and "flutter".

An indication of the presence of amplitude / frequency response errors can be obtained by a simple experiment using the test signal part of the apparatus together with a 1kHz signal generator and an r.m.s. meter. Both signals are applied in turn at equal powers to the circuit under test. The output power of the circuit under test is measured: any difference in the power of one signal relative to the other (under linear conditions) indicates an error in the amplitude / frequency response. More detailed results can be obtained by analysing the spectrum of the p.r.b.s. test signal after passing it through the circuit under test. If a correlator is used it is then possible to measure delay and the degree of gain-phase matching of transmission circuits (as may be of interest in stereo or quadraphonic sound systems).

The technique of using a p.r.b.s. signal for testing linear systems is well known, but in the past has required expensive test apparatus (e.g. cross-correlators). With the advent of the microprocessor a relatively inexpensive equipment might well be produced for measuring linear distortions using a p.r.b.s. test signal, and such a microprocessor arrangement could easily be added to the double comb-filter apparatus since a-d and d-a converters are already provided.

When the double comb-filter method is used to measure the non-linearity distortion of sound recording and replay systems. "wow" and "flutter" produce spectral components in the gaps of the test signal. The "wow" and "flutter" components can readily be measured: only one p.r.b.s. signal and its complementary comb-filters are required. With this test signal, non-linearity products are hidden and only "wow" and "flutter" components appear in the spectral gaps. In practice it has been found that the recovered level of products generated by "wow" and "flutter" is much higher than the background noise of high-quality recorders; typical figures for "wow" and "flutter" noise-separation are 35dB at 15 in/s and 30dB at 7½ in/s. No work has been done to compare these "wow" and "flutter" figures with those measured by more conventional methods and no subjective tests have been conducted to establish tolerable levels of "wow" and "flutter" measured by this new method.

The measurement of non-linearity distortion in the presence of "wow" and "flutter" is likely to require comb-filters with wider stop-bands as the "wow" and "flutter" signals are likely to produce sidebands close to the spectral lines of the test signal. The more complex digital comb-filter required in this application could conveniently be instrumented using a microprocessor arrangement.

The author

R A. Belcher joined the BBC Research Department in 1969 after graduating from the University College of North Wales. He has worked on many aspects of analogue and digital processing of sound and video signals and has been much Involved in the use of subjective tests. In 1976 he was awarded the Gyr and Landis prize by the IEE for work on comb-filter methods of measuring audio distortion, and in 1977 he gained a Ph.D. from the University of Surrey for a thesis on this subject. Since early 1977 he has been doing research in quadraphony. Early this year Dr Belcher left the BBC to become a senior physicist at Velindre Hospital Cardiff.

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