3. The setting of $C_1$ is quite critical if the highest order of accuracy is wanted. An easily available standard frequency is the BBC transmission on 200kHz, which can be picked up in most parts of the country on a few feet of wire attached to a simple tuned circuit. For example in North Yorkshire, well over 100 miles from the transmitter, I get 150mV peak to peak on a 10ft aerial attached to the top of a tuned circuit, and this is much more than adequate to display on one input to a double trace oscilloscope, while the other trace is locked to the calibrator switched to the 10μs output. There will then be exactly two radio waves for every marker, and $C_1$ should be adjusted until the radio waves are stationary on the screen. It will be found that the adjustment of $C_1$ is then much too fierce, and a better result is obtained by splitting it into a fixed capacitor in parallel with a variable of some 10pF.

4. Finally, in Fig. 3, I would query how accurate counting could be accomplished. Whichever frequency was used to lock the timebase, the other would be travelling across the face of the tube at a rate of knots too fast to count.

*W. Winder*
*Harrogate*
*Yorkshire*

## AUDIBLE AMPLIFIER DISTORTION

IN his article on amplifiers (November 1977) Peter Baxandall has rested a naively drawn case on a narrow conception of distortion. An extreme subjective position — that there is no difference to be heard between "first class, competently designed, amplifiers" — is supported by rational criteria which, though conventional, are incomplete in themselves and utterly inadequate to the task.

It is astonishing to us that there persists — at such a late date, and in the face of even our own relatively short experience with a wide variety of internationally available commercial power amplifiers — an attitude of mind that refuses to respond to the ever increasing weight of subjective evidence from enthusiasts and experienced hi-fi equipment dealers.

We do not believe it adequate — however superficially justifiable — to attack the problem by gripes against the British hi-fi press and its reviewers' shortcomings. We do believe that "first class, competently designed" power amplifiers sound different, and that the differences matter and can be rationally accounted for, and a prescription for universal good quality laid down.

In the first place we do think total harmonic distortion in the classical sense — with the harmonics weighted in Olson's manner — to be relevant. At the same time we know that pre-amplifiers and power amplifiers do sound different even though their "on paper" specifications are far superior to the programme material, from tape or disc, used in their evaluation.

If the Quad diagnostic set up (Fig 1, original article) is to be used as the ultimate test of amplifier quality why then do the Quad 303 and 405 sound different? This is not a trick question — in that the 303 has an output capacitor and the 405 does not — but what does happen if we put, say, a 2000μF capacitor between an amplifier and a loudspeaker? The sound becomes "warmer" and "muddier." Yet this intrusion would not appear in an analysis of the Quad frequency-response and phase-balancing network. (To us the Quad network — representing a passive amplifier — appears to have 12dB/octave slopes and thus to be on the threshold of instability.)

In his AES paper of 1973 Otala[1] describes a diagnostic circuit which he treats as a constant delay with one h.f. roll-off pole included to compensate for one dominant h.f. pole used passively at the input of the amplifier he describes. The reason would appear to be that the ideal amplifier will delay but not destroy the sound.

However, a 1kHz toneburst with d.c. offset (representing speech, for example) into a circuit such as the Quad diagnostic network will distort — the toneburst will tilt. But because the amplifier cancels this tilt the effect of the network is not observed. Thus a dramatic silence — suggesting no distortion. Into a loudspeaker there would be an audible change when compared with a d.c. amplifier or one with a cut off at about 3Hz or less.

Experience in the last ten years suggests that amplifiers (valve and transistor) start to sound alike when the bandwidth is extended nearly a decade on each side of the audio band (giving 3Hz—150kHz,—3dB) *at full power*; and when the distortion is about the same from 20Hz to 20kHz; and when the damping factor at the point where feedback is sampled is relatively constant over the whole audio band (implying a wide open-loop response); and when total phase change is less than 10° from 20Hz to 20kHz.

There are other subtle factors that affect the final quality. But differences in sound are not easy to express in words. Nor is it possible always to say which is right and which is wrong. But if a difference exists one must attempt always to verify, to measure and to explain.

*Tim de Paravicini and John Greenbank*
*Moonlight Electronics Ltd*
*Cambridge*

**Reference**

1. "An Audio Power Amplifier for Ultimate Quality Requirements." Jan Lohstroh and Matti Otala, Audio Engineering Society 44th Convention 20-22.2.1973, Rotterdam.

MR BAXANDALL raises several spurious arguments in an apparent attempt to prove that audio amplifier design reached its pinnacle in the mid-sixties and that further work is therefore pointless (November 1977 issue).

No serious worker in this field would doubt that extreme care and attention to detail are necessary whenever any comparative testing is undertaken. It is an established requirement that all documented experiments be prefaced by a description of "methodology". Indeed it is quite common to find that far more time and effort is expended in establishing an experimental regime and in the elimination or quantification of potential errors than in the performance of the comparative experiment itself. A further necessity is the use of "control" experiments to establish a median and to prevent "cheating" and the influence of emotional prejudice. It is regrettable that some reviewers omit this part of the scientific procedure.

Such knowledge of valid experimental technique is not unique to the BBC or to Mr Baxandall. It has been applied by anyone who has been to university.

Despite the doubts of Mr Baxandall and the apparent desperations of Mr Williamson (letters, October 1977), the most careful experimental auditioning does reveal audible differences between many audio amplifier systems. There is no magic about this or requirement for "golden ears". Nor is there any need for Mr Williamson to get on to his engineering high-horse to make blanket condemnations. The whole point has been missed. It is not seriously suggested that amplifier differences can only be heard and not measured. A great many of the "subjective" differences can now be tracked down and accounted for in engineering terms. However, not all the necessary experimental techniques have been published for obvious commercial reasons.

The Quad nulling experiment is well known but has significant limitations. A considerably more exact and elegant technique is now used by AEA in the USA and other workers in the UK. This is the technique of quantisation of the input and output signals for analysis by a digital computer. This technique enables a "real-time" comparison to be made throughout the course of a piece of music and with a great degree of accuracy; it has permitted some interesting correlations between measured errors and audible deficiencies.

I cannot believe that Mr Baxandall takes the subject seriously if he never listens to his amplifiers as part of their development programme. Apart from anything else a carefully planned series of listening tests can check an amplifier's compatibility with various loudspeakers and cartridges and identify problem areas for investigative laboratory action. Before writing this letter I was able to contact the designers of six different UK makes of high-quality audio amplifiers. In each case the designers (all qualified and experienced engineers) considered it necessary to perform listening tests in the course of their development programme. Obviously either they or Mr Baxandall are wrong.

I perceive, however, that the old men of the industry are set in their ways and are unlikely to change. No doubt Messrs. Baxandall and Williamson do not expect Quad to bring out replacements for the 303 or the 33. Personally I have more respect for Quad. And, no doubt, Mr Baxandall will not find it necessary to publish any new amplifier circuits. I find it sad that perfection has already been reached because so much sounds so imperfect.

*Stan Curtis*
*Mission Electronics Ltd*
*London, SW6*

MAY I add my support to Peter Baxandall's criticism of reviewers who describe in great detail gross differences in the performance of many of the amplifiers and loudspeakers in the top quality class when careful comparison indicates that there are no such audible differences. Moreover they claim to hear these gross differences when commercial gramophone records are the source of the test programme.

Now the distortions in any recording and replay system using commercial gramophone records are between one hundred and one thousand times greater than in any of the top quality amplifiers, while the loudspeakers used to judge the amplifier performance have distortions about one hundred times greater than the amplifiers. Not only are the distortions in a

gramophone record system vastly greater than in a good amplifier, but a high proportion of the distortion is of the frequency modulation type and significantly more annoying per unit of distortion than are the harmonic and intermodulation distortions that occur in an amplifier. Perhaps one of the reviewers can provide an explanation of just how it is possible to detect the trivial distortions in a good amplifier in the presence of programme source distortions that are about one thousand times higher.

My laboratory is continuously involved in assessing the sound quality from a wide range of equipment and the most troublesome problem that we encounter is that of obtaining programme material of the high quality that is essential if valid comparisons are to be made on amplifiers and loudspeakers in the top class. We rejected commercial gramophone records as a source at least ten years ago and we confine ourselves to using first or second generation copies of 15in/s tapes played on a professional tape machine in the £2000 class. This sets a high standard and leads us to reject 80% of the studio tapes we obtain because they are significantly inferior in quality to the remaining 20%. When gramophone records must be used we employ direct cut discs.

With such high class programme material at our disposal we cannot find any trace of the gross distortions so vividly described by a small group of reviewers having facilities no more extensive than many hi-fi enthusiasts and undertaking the reviewing in their spare time. Adjectival extravagance appears to be considered an acceptable alternative to technical accuracy, a substitution that can only lead to the rejection by the industry and by the public of those magazines that indulge in these fantasies.

I would comment on an important aspect of these comparisons that is rarely appreciated. There are generally only small differences in the performances of components all in the same price class and the issue is rarely one that unit "A" is clearly better than unit "B". In practice "A" has some distortions, using the word in its widest sense, that "B" does not have, and vice-versa. The judges have to decide which of two different combinations of distortion they find least objectionable. If one comes to a decision when listening to radio station or record No. 1, it is common to find that the opposite decision is reached on station or record No. 2. Differences in the quality of the programme sources are at least as important as the differences in the performance of equipment in the top class.

I would comment on another of Peter Baxandall's points, the use of listening panels in assessing sound quality. Listening panels appear at first thought to be an excellent way of obtaining a broadly based opinion of the sound quality of a system, but actual experience leads us to doubt that view. If more than a few judges are involved in a single listening session they cannot all occupy reasonable seats, nor can they make the changeover between units being compared just at the instant when the music is appropriate for checking some specific difference in performance that they have noted. We are gradually moving away from the use of such panels unless we are specially requested to institute panel tests by a client. The procedure we now prefer is to have three or four experienced listeners compare the receivers individually, operating the changeover push-button etc., themselves while listening to high quality programme material. Each writes up his own notes and after the last man has done so, he reads the previous notes, checks for differences of opinion and when advisable re-checks any point of differences. Each listener is free to make a changeover just when he wishes to check some specific difference between the two systems being compared and he is free to continue his comparison for just as long as it takes to arrive at a soundly based opinion. We find this procedure leaves the listener much more confident in his decision than when taking part in a panel listening test. Combined with the results of measurements on the objective aspects of the two systems and an appropriate statistical analysis of the data, we believe that we obtain a more accurate indication of the performance than is obtained from the current assessment techniques.

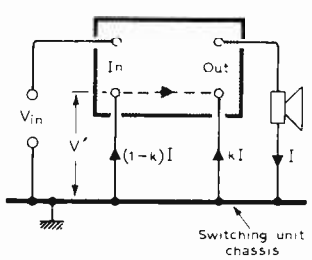
*James Moir*
*James Moir & Associates*
*Chipperfield*
*Herts*

*Mr Baxandall replies:*

I was interested to hear about the great care taken by James Moir to obtain programme input sources of the highest available quality, and I agree with his preference for conducting the tests with one listener at a time, this person being allowed to operate the changeover switch. The identity of the equipment tested, in relation to the switch positions, should not be known to the listener. I note that experience has been that, when all due precautions are carefully taken, first-class amplifiers are found to be absolutely indistinguishable.

Though Tim de Paravicini and John Greenbank say I have a narrow conception of distortion, they do not state how their conception differs from mine. I would say simply that an amplifier has perceptible distortion if it causes a perceptible quality change when introduced into a very high grade audio chain, due care being taken to match levels. Surely this is the fundamental meaning of the word? If my article is carefully read, it will be found that no other conception of the meaning is implied.

It is suggested that I refuse to respond to the ever-increasing weight of subjective evidence relating to audible differences between first-class amplifiers, and Stan Curtis says this is because, like my good friend Reg Williamson, I am "an old man of the industry, set in my ways and unlikely to change." We've had a good laugh over this – but I do, nevertheless, accept that I'm set in my ways and unlikely to change, if this is taken to mean that I view all new evidence with the initially suspicious attitude that is a proper accompaniment of a truly scientific outlook. Thus I do, indeed, refuse to respond *too easily* to evidence which is not the outcome of proper scientific procedures, which does not tie up logically with other established results, and which disagrees with my own direct experience. However, if, on further careful investigation, I find my earlier notions are proved to be wrong, then I will certainly, and gladly, change my views.




But I have found no trace of reliable evidence to support extreme notions such as that a power amplifier should be able to produce full power output from 3Hz to 150kHz, nor that its phase shift should be less than 10° at all audio frequencies. The fact that a university department somewhere or other may have concluded that something of the sort is desirable does not seem to me in itself to carry much weight.

Messrs de Paravicini and Greenbank say explicitly that they do believe that first-class competenently designed power amplifiers sound different, and it may be relevant to mention, in this context, that since writing the article, my attention has been drawn to an interesting contribution "Six amplifiers – how did they sound?" by I. G. Masters, in the *Audio Scene Canada* magazine. This says, to summarise it very briefly, that six good amplifiers were carefully compared on an A - B basis, using various loudspeakers, and "Lo and behold! – we heard some very striking differences." Some showed up badly with difficult loads, some didn't – but the tests were done at quite low power levels and no overloading was allowed to occur. It was then discovered that some amplifiers measured the same when tested separately as when tested in the comparator set-up, whereas others did not, and this led to a careful investigation of earthing arrangements in the comparator. When the unwanted earth-loop effects had been understood and cured, "we heard . . . no difference. None." A "straight wire" test was also done – "The amplifiers not only sounded the same as each other, they sounded the same as our 'straight wire'." Though no diagram is given, it seems that the essence of the situation can be represented as here shown. The various amplifiers initially had their inputs and outputs switched on the live sides only, the earthy sides being taken to the switching unit chassis "as with most (possibly all) switchers that would be found in hi-fi stores." Thus, with one amplifier switched in, the loudspeaker current, $I$, must return to the earthy output terminal of the amplifier, and if the amplifier has the earthy sides of its input and output joined together internally, as is often the case, this current can return via two paths, as shown. The portion $(1 - k)I$ thus flows in the signal input earthy connection, producing a voltage drop, $V'$, as shown, and this is injected in series with the signal source. Since $I$ may be several amps, a small fraction of an ohm of lead impedance will be enough to produce a significant value of $V'$, and this value is clearly dependent on the variation of loudspeaker impedance with frequency. Some amplifiers have no low-resistance internal connection between the earthy sides of their inputs and outputs, and in such cases all the loudspeaker current must return directly to the earthy output terminal, i.e. $k = 1$. No peculiar effects then occur.

Even when double-pole switching is adopted, similar effects to those described above can occur if the wiring is not suitably arranged.

Messrs Paravicini and Greenbank ask why the Quad 303 and 405 sound different. I suggest they should carefully re-read Peter

Walker's contribution on page 135 of *Hi-Fi News* for July 1977. Provided the comparison is completely fairly done, as stated, including arranging that the overall system frequency response is not significantly different in the two cases, Quad are prepared to stake their reputation on the 303 and 405 sounding exactly alike. Differences in frequency response of the amplifiers are negligible provided the programme material is free from significant unwanted components at subaudio frequencies.

The comment about 12dB/octave slopes in the Quad nulling test set-up indicating a system on the threshold of instability is not justified, for the elements in question are not within a feedback loop, either in the amplifier or in the separate network.

Returning to Stan Curtis's letter, he says the Quad nulling experiment is well known "but has significant limitations." Unfortunately he does not state what he regards these limitations as being. It seems to me that when properly used, in the *various ways* mentioned in my article, it is by far the most satisfactory technique for directly investigating subjective distortion in such a way that the "margins of safety" may be estimated.* I have read about the digital technique being used by Analog Engineering Associates, but whether this should be regarded as more elegant depends, I think, on one's point of view. It is certainly far more complex and expensive, and, because of this, may be said to lack the elegance of simplicity! In common with the Quad nulling technique, it operates with programme as the signal, and whereas it clearly can be made to yield vast quantities of information, not all useful, on effects going on within amplifiers, I do not see that it is a preferable technique for investigating the subjective quality of a given amplifier.

Stan Curtis finds it difficult to believe that I take the subject seriously, since I do not normally listen to amplifiers as part of the development programme. Though he may find this difficult to believe, it is nevertheless true! With regard to the compatibility of amplifiers with loudspeakers and pickup cartridges, I cannot for the life of me see why listening tests should be required, for the problem is a straightforward one involving impedances, phase angles, signal levels, protective-circuit operation etc. It does not surprise me to learn, however, that many designers do feel it necessary to resort to listening tests. Mr Curtis says "obviously either they or Mr Baxandall are wrong." But is it not, perhaps, truer to say simply that different people do things in different ways? It is a fact that a design I did for a commercial firm was not listened to at all until the circuit design was quite completed, but subsequently came top in an independent subjective assessment of many competitive designs from various countries. Quad too assure me that they adopt the attitude that if you understand what you are doing thoroughly enough, there is no need for listening tests during the design and development of amplifiers, and that they do not normally carry out such tests. Moreover, their pioneering work on electrostatic loudspeakers has shown that even loudspeaker development can with advantage be done largely on a basis of "theoretical designability," with the bare minimum of subjective testing.

Lastly, Stan Curtis finds it sad that I should believe that perfection has been reached, for, as he says "so much sounds so imperfect." I can assure him, most sincerely, that I couldn't more fully agree with this observation as far as the end product of most hi-fi systems most of the time is concerned. If it were not so, we could more frequently enjoy artistic subtleties and differences without the intrusion of technology. I also agree with him that there are many amplifiers around that fall short of the ideal performance, as judged subjectively. But I must end by repeating that I am in no doubt at all that the best amplifiers, unlike some other links in the overall chain, easily meet the requirements for subjectively perfect sound reproduction. Nevertheless, designers, including myself, will continue to bring out new designs, for there are so many reasons for doing this other than basic sound quality – power ratings, reliability, production economy, versatility of functions, etc.

*Peter J. Baxandall*
*Malvern*
*Worcs*

*As some readers will have spotted, the editor inadvertently left out two resistors, one in each input to the monitoring system.

## LOGIC DESIGN

THERE is an important principle that was not brought up in the fourth article of the "Logic design" series by Holdsworth and Zissos (May 1977 issue).

The realization of the circuit for the alarm bell output in Fig. 14 (f) is more complex than need be. Two of the cells in the merged state diagram Fig. 14 (d) indicate unstable states in which the circuit cannot remain. Therefore the outputs in these two states do not matter and the $b$ output can be high. This simplifies the circuit from:

$$b = \bar{A}f\bar{a} + A\bar{f}\bar{a}$$

to

$$b = \bar{A}f + A\bar{f}$$

In this example there is not a great saving in hardware; two 2-input Nand gates are used instead of two 3-input Nand gates, but in more complex problems the savings could be significant.

One must take care in the use of this simplification as there is a delay in the transition from the unstable to the stable state. This results in an output spike of short duration which could affect a following circuit. This spike is far too short to operate the alarm bell in the illustrated problem.

*A. R. Harris*
*Biltondene Developments Ltd*
*London SW8*

*Professor Zissos and Mr Holdsworth reply:*
We agree with Mr Harris that a further reduction of the bell equation is possible by using the circuit conditions, $A = 0$, $f = 1$ and $a = 1$ and $A = 1$, $f = 0$ and $a = 1$ for simplification purposes. The bell equation then reduced to

$$b = \bar{A}f + A\bar{f}$$

However, in this circuit a spike will not occur as a consequence of using this simplification and it is essential for the bell to ring particularly when a fault occurs to draw the attention of the operator to its occurrence.

When the transition $S_{01}$ to $S_{23}$ is made (Fig 14(d) the input signals required are $f = 1$ and $a = 1$. By virtue of the design specification these signals must occur in the sequence $f = 1$ followed by $a = 1$. Initially the circuit will take up the condition $A = 0$, $f = 1$ and $a = 0$ and the bell rings as required. The transition then takes place when $a$ becomes 1. During the transition from $S_{01}$ to $S_{23}$ $f = 1$ and $a = 1$ and $b = 0$. When the transition has been completed $A = 1$, $f = 1$ and $a = 1$ and again $b = 0$ as required.

Similar conclusions may be drawn regarding the transition from $S_{23}$ to $S_{01}$.

Perhaps it should be noted that, due to an authors' error, state $S_{23}$ has been marked incorrectly as $S_{02}$ and the bell signal in this state should be $f\bar{a}$.

*B. Holdsworth and L. Zissos*

*Editor's note: The following remarks were unfortunately omitted from the authors' reply to Mr R. M. Hutton's letter on minimisation in logic design in the December 1977 issue. Apologies to the correspondents.*

We are not at all sure what is debatable about Example L, nor can we agree with your statement that in this example we have demonstrated the vulnerability of our method. We are aware that a change of state assignment will lead to a different solution. All other known methods of logic design are vulnerable in precisely the same way and it is up to the logic designer to examine all possible solutions if he wishes to find the simplest solution. This is perfectly easy to do in the case of a four-state state diagram but becomes increasingly more difficult as the number of state variables increases. If minimal solutions are not vital it is probably more economically sound to reduce the design time.

With respect to the relative advantages of mapping techniques in comparison with algebraic methods this is really a question of which method the designer is familiar with. Certainly students we have taught do not find algebraic methods any more difficult to use than mapping techniques and vice versa. If you refer back to article 1 on Boolean algebra you will find that there are a very limited number of rules to remember. We would not press a claim either way with respect to this point and would suggest that the designer should use the method he is most familiar with.

*B. Holdsworth and L. Zissos*

## THE DECATRON

READING T. R. Thompson's letter (November 1977 issue) about the 3NF valve "integrated circuit." brought to mind the old "Decatron" tubes, which are still available (if you know where to look). These, of course, are the equivalent of a decade counter-decoder-driver and display all in one! They haven't even done that in semiconductor i.cs to my knowledge.

*R. E. Williams*
*Tilsworth*
*Beds*

*Letters commenting on Eric F. Taylor's articles "Distortion in low-noise amplifiers" (August and September 1977) will be published in a later issue.*