DISTORTION

By "CATHODE RAY"

What Do We Really Mean By It?

IF there had been any doubt about there being a great many people intensely interested in what our American friends call "hi fi," that doubt was dispelled last autumn by Mr. Briggs when he sold the full capacity of the Royal Festival Hall (sitting and standing) in the first four days, on an announcement that he was going to demonstrate loudspeaker reproduction in comparison with direct musical performances. It has been necessary to arrange a second house. And I remember being mightily astonished when the Editor told me how many copies of the Williamson amplifier reprint had been sold. All this being so, there is naturally a demand for some scale of measurement for comparing one piece of sound-producing equipment with another. The advertisement copy writers' "perfect reproduction," "no trace of distortion," "impeccable fidelity," "thrilling tone," etc., cut no ice at all with *Wireless World* readers. They very rightly want some definite figures of performance.

So most of the advertisements nowadays say "distortion at 12 watts output is not more than 0.3%," or whatever it may be. That is certainly an improvement in principle, but we may be forgiven for asking some questions. Is 0.3% good, bad or indifferent? If another make of amplifier distorts 0.3% at 12 watts can its fidelity be assumed to be the same? If it were 0.1% how much better would it sound? And if it were 1%—or 5%—how much worse?

Twenty-five to thirty years ago people were already taking quite a lot of interest in this matter of fidelity of sound reproduction, but the data then consisted of a graph of output against frequency—what is usually called a frequency characteristic. If it was in an advertisement, the scales were chosen so as to make the graph look as nearly as possible like a horizontal line drawn with a ruler. The thing was then described as "distortionless." To the best of my recollection, percentages were not mentioned. "Distortion" was generally understood to mean frequency distortion—the unequal amplification of different frequencies. The reason for this was that the most obvious shortcoming of the very early gear was its frequency characteristic, which consisted of a violent peak in the middle or upper middle, and very little else.

So far as amplifiers were concerned, it was a fairly easy development to obtain their frequency characteristic curves and to improve their design so as to flatten out the peak into a nearly level plateau extending over the useful frequency range. And so began an era in which high-fidelity enthusiasts vied with one another in smoothing out the last fraction of a decibel (a unit which by then had come into vogue) often regardless of the vastly greater irregularities in the characteristics of the loudspeaker and the room in which it was heard. There *is* a good reason for aiming at a very level amplifier characteristic, but even now some enthusiasts may not realize that it is not the avoidance of frequency distortion as such (for on that count a peak of the order of one decibel is quite unimportant) but the obtaining of maximum undistorted output. If one narrow band of frequencies is amplified 1db more than others, as shown in Fig. 1, the whole level of output has to be lowered 1db (e.g., from 10 watts to 8 watts) in order to avoid overloading. In other words, moderate frequency distortion is bad, not as frequency distortion but as a potential cause of overloading or non-linearity distortion.

Non-Linearity

As time went on and gross frequency distortion was eliminated, the possibilities of appreciable improvement of sound by further levelling out of frequency characteristics dwindled. "Distortion" ceased to be frequency distortion and became non-linearity distortion (commonly but illogically called "non-linear distortion"). Now this is where we must be clear about the meanings of terms. "Non-linearity" means lack of straightness or proportionality of a characteristic, expressed as a graph. The particular characteristic understood in this connection is the input/output characteristic of any part of the equipment. Ordinary resistors are linear, because the voltage across them is directly proportional to the

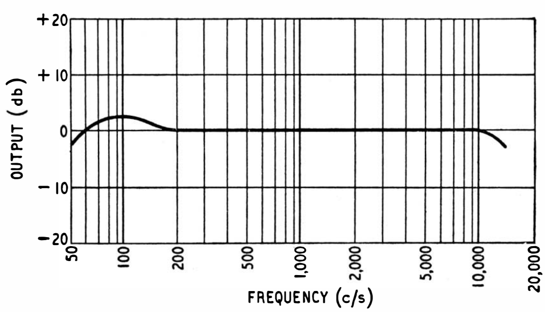


Fig. 1. Example of frequency distortion that is quite negligible as such, but should be avoided if the maximum undistorted power output is desired.

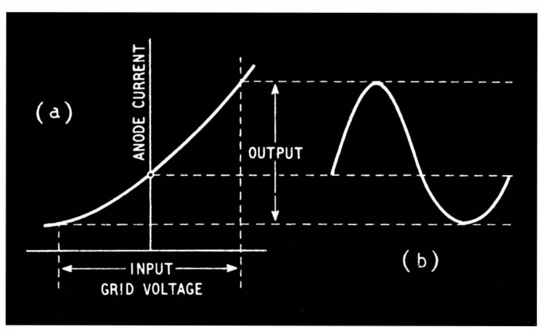


Fig. 2. Typical valve characteristic (a) with the curvature somewhat exaggerated we hope, showing the resulting distortion of a sine wave (b).

current through them; in other words, they obey Ohm's law. Valves and iron-cored coils do not. Fig. 2 (a) shows a typical sample of anode-current/grid-voltage characteristic. If the grid bias is set so that the working point is O, an input signal of sine-wave form will make the voltage swing equally on both sides of O as shown, and obviously the waveform of the output current so caused (b) is distorted, the positive half-cycle being bigger than the negative.

Harmonic Distortion

This is the effect we are now going to study. It is sometimes called "amplitude distortion," but that term has been allotted to a different effect, which may or may not happen at the same time as waveform distortion. Whereas waveform distortion is a result of non-linearity during each individual cycle, amplitude distortion means that the output level as a whole is not directly proportional to the input level. It is possible with a characteristic of the Fig. 2 (a) type, which obviously distorts the waveform, for the output to be proportionate to the input, the opposite disproportionateness of positive and negative half-cycles cancelling out and resulting in no amplitude distortion.

One of the first things we learn about non-linearity is that it creates harmonics. This has been explained so often that I needn't go into it fully. The usual line is to add together various sine waves whose frequencies are harmonically related (i.e., exact multiples of one particular frequency, the fundamental or first harmonic) and find that the results are distorted waveforms, some of which resemble those obtained by non-linearity. For example, in Fig. 3 a double-frequency or second harmonic (b) is added to a fundamental (a) and the result (c) is very like the output of Fig. 2. That is the synthetic method. Then there is the analytic method of breaking down a distorted wave (graphically or by experiment) into a fundamental and harmonics. It is then explained that the characteristic tone of each musical instrument depends on the amounts of the various harmonics it emits, relative to the fundamental, and that if these proportions are altered, either by frequency distortion or by adding harmonics by non-linearity, the characteristic tone is distorted.

True enough. But by now we are supposed to have got rid of frequency distortion that could drastically alter the proportions of harmonics; such frequency distortion, for example, as poor high-frequency response, which would tend to suppress them. And while such distortion might make a clarinet sound like a flute, it couldn't (even if it took place) account for the appalling sounds that result from severe overloading. The fact that the sounds produced by musical instruments listened to with pleasure contain a generous series of harmonics is evidence of that. An amplifier advertised to give 10% harmonic distortion would hardly find favour with "hi-fi" connoisseurs, yet what is 10% compared with the 50% or more generated by well-regarded pianos? If the only effect of non-linearity were to create harmonics, we should be at a loss to explain how such unpleasant reproduction comes with quite moderate harmonic distortion percentages.

It is now generally agreed that it is *not* the harmonics that are responsible for the worst of the unpleasantness. In *Wireless World* for May 19th, 1938,* I described a simple experiment for demonstrating this. On the assumption that copies of that issue may not be lying around to hand, I will briefly recap. A receiver is arranged with two separately-adjustable sine-wave input signals and an output meter. The low frequency, say 100 c/s, is adjusted to be 10 times the voltage (and therefore 100 times the power) of the other signal, say 533 c/s. In spite of this the 533-c/s note sounds about as loud as the 100 c/s, because the ear is so much more sensitive at the higher frequency. At first each signal can be heard as a clear pure note, as it was when alone. But at a certain setting of the main volume control a roughness of tone becomes noticeable; and at a still higher setting the higher note becomes indistinguishable, the whole output degenerating into a harsh rattling kind of hum.

* "Debunking Harmonic Distortion."

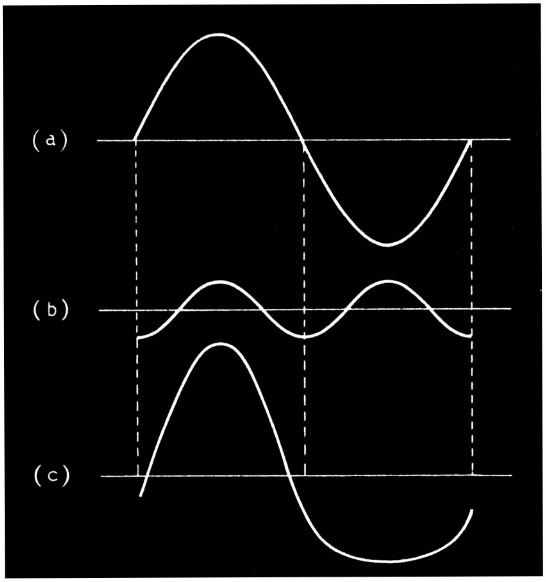


Fig. 3. Showing how the distortion in Fig 2(b) can be made synthetically by adding together a second harmonic to the original (fundamental) waveform.

If now the 100 c/s is switched off, the 533-c/s note is heard with perfect clarity. That is only to be expected, because it is weak enough to be well below the point of serious distortion. What might not be expected however is that when the 533 c/s is switched off the 100 c/s becomes quite clear and altogether different from its sound when both signals are on. This is so, notwithstanding that switching the 533 c/s off reduces the output power by only 1%, which by itself is not enough to make an appreciable difference to the amount of distortion. An increase of much more than 1% in the power of the 100 c/s alone has no such devastating effect as switching on the weak 533 c/s.

Intermodulation

The obvious conclusion is that some kind of distortion is taking place when both signals are being handled together by the amplifier which is not present with only one. Here again we come to a well-worn chapter in radio theory, of which Fig. 4 should be sufficient reminder. (a) is the undistorted two-signal input, and (b), assuming distortion of the kind shown in Fig. 2, is the distorted output. At the positive peaks of the "strong-low" signal the "weak-high" signal is amplified more than at the working point O, and at the negative peaks it is amplified less. So the weak signal is amplitude-modulated at the frequency of the strong. This can be seen more clearly if the strong signal is taken away (c). The said chapter of radio theory explains how this process introduces new frequencies, not necessarily multiples of either of the input frequencies, but "sum and difference frequencies." The Fig. 2 kind of characteristic, which creates mainly second-harmonic distortion of the low-frequency signal (f_1, say) causes the high-frequency signal (f_2) to wax and wane once per low-frequency cycle, and the frequencies created by modulation are mainly $f_1 \pm f_2$, known as the simple sum and difference or second-order intermodulation frequencies. In our experiment they would be $533 \pm 100 = 433$ and 633 c/s.

This distortion is the kind that one gets with a triode output valve, and which a push-pull circuit is used to balance out. If a pentode is used, or the push-pull system is over-driven, both positive and negative peaks tend to be affected in the same way. The result is that the third harmonic is the strongest, and third-order modulation frequencies, $f_1 \pm 2f_2$, 333 and 733 c/s in our experiment.

Generally distortion consists of a mixture of second and third, with smaller proportions of higher numbers, but most practical cases fall into one of two main classes, in which either second or third predominates.

Obvious ?

So far we have talked about the 100 c/s modulating the 533 c/s, but not the other way about. Why? Well, if one man fought another ten times as strong he might inflict something on him, but it would usually be negligible compared with what he received. In the same way we have neglected the modulation of the strong signal by the weak, though it does exist and is why the process is called *inter*modulation. When two signals going through the mill together are equally strong, each modulates as much as it is modulated.

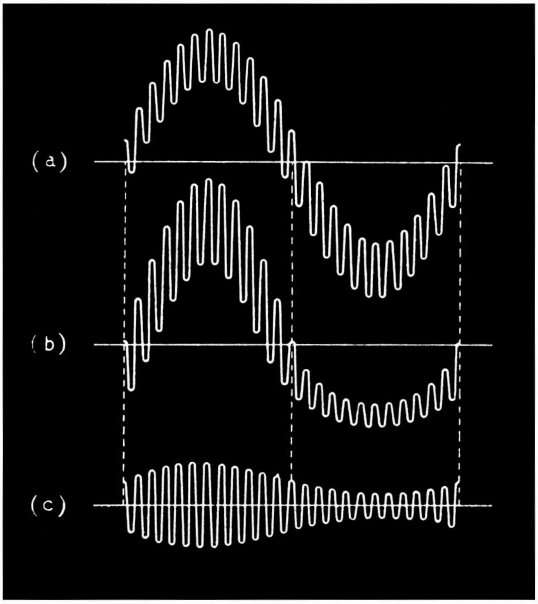


Fig. 4. When a higher-frequency but weaker sine-wave tone is added to the low-frequency signal at the input, the waveform of the combination is as at (a). After suffering distortion of the Fig. 2(a) type it comes out like (b), and by taking away the low frequency the damage to the higher frequency can be seen more clearly (c).

I said that the experiment made it obvious that intermodulation, not harmonic distortion, is responsible for nearly all the unpleasantness. That conclusion can hardly be doubted so far as the particular conditions of the experiment are concerned. But it is always risky to draw quick conclusions about the connections between physical causes and the resulting impressions on the senses. If a physical force acts on a lifeless object, the effect conforms to a simple equation covering all such events. But the impressions a human being receives as a result of physical causes often seem to bear no predictable or clear relation to them. A race of stone deaf men, though they might master the science of physical sound, could never discover what it was like to hear. Even where there does at first seem to be a clear connection, it may be misleading. For instance, it might seem definite enough that the higher the frequency of a sound the higher the pitch of what is heard. But even there it is not safe to assume that the two things run perfectly parallel, for it is found that the pitch of a note of constant frequency varies slightly with its intensity.

Still less safe is it to draw hard and fast conclusions about the relationship between unpleasantness of sound and the distortion that causes it. Our particular "obvious" conclusion—that intermodulation accounts for nearly all the unpleasantness caused by non-linearity distortion—when I expressed it in 1938 was immediately challenged. And it certainly is unwise to draw such a sweeping conclusion on the basis of one simple experiment. Does it hold for all different combinations of frequencies? And does it hold for typical programmes?

One typical programme is speaking. But speech is an extremely difficult type of sound to study for unpleasantness. Music is much easier, so we shall assume music is our staple diet of listening (whether as the food of love or not is unimportant just now). There do seem to be some clear-cut rules about combinations of musical sounds. One of them is this: that the smaller the whole numbers in which the ratio of the frequencies of two sounds can be expressed, the more harmonious the combination appears to the listener. To take one extreme, the ratio with the smallest possible numbers is 1:1, which means that both sounds have the same frequency, so are heard as one sound, without any disharmony or indeed any distinction at all between them (assuming, of course, that they are coming from the same source). The next simplest ratio is 2:1, which means that the frequency of one note is twice that of the other. Musicians say that it is an octave higher. Although of course the two notes are easily distinguishable when heard separately, they blend so smoothly together that most untrained listeners are unaware that more than one note is being played. People are said to be singing in unison even though the women are singing all their notes twice the frequency of the men. This being so, it should be pretty safe to say that even 100% second-harmonic distortion, *if it consisted only of the creation of second-harmonic or octave-higher frequencies,* could not cause harshness in the sound. It would certainly make the music sound "brighter" and as this would be different from the original it would have to be classed as "distorted," though to some ears it might be considered an improvement. The effect on a single sustained note can easily be tried if one has two a.f. signal generators that can be synchronized an octave apart and the higher one brought up from zero level. The effect is identical with that obtained with a single note through an amplifier which can be made to give pure second-harmonic distortion. The same effect on real music can be produced in organs, by bringing in a coupler that adds octaves to all the notes played. This is *not* the same, however, as playing the music through the distorting amplifier, because that adds difference tones as well.

And that, of course, is the crux of the whole matter. But before going into it, let us continue a little longer with our lesson in the theory of harmony. As a non-musician I shall have to be careful; but, on the other hand, musicians themselves seem quite unable to talk our language of frequencies, etc., so fail to tell us clearly what we want to know.

The next simplest ratio might be said to be 3:1. But in music the scale starts all over again at the end of an octave, and so a note 3 times the frequency of another may be regarded as $1\frac{1}{2}$ times the note an octave higher; consequently our next ratio is really $1\frac{1}{2}$ or 3:2. And the musicians would agree, I think, that this is the next most important "interval" to the octave, by virtue of which they name it the dominant. The original (lower) note they call the tonic, by the way. And when tonic and dominant are played together, we are conscious of hearing something more complicated than a single note, or even the "brightened" note made up of the 2:1 combination; yet it is undoubtedly "in tune" and harmonious. So is a 3:1 combination, such as a fundamental and third harmonic, because the harmonic lies in this "dominant" relationship to an octave higher than the fundamental, which as we have seen (or rather heard) is almost equivalent to the fundamental.

Harmonics and Harmony

It would seem, then, that the creation of third harmonics would by itself introduce no harshness or discord, nor perhaps even unpleasantness except to the musical connoisseur who would resent unison passages for flutes being given a harmonic accompaniment. The general effect would be to make the balance of tone still "brighter" and also somewhat "richer" by the addition of the new harmonies. "Nasal" is a description that is sometimes used to refer to the double effect.

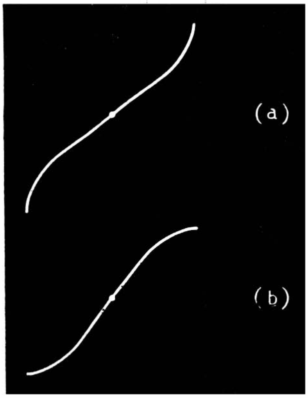


Fig. 5. Here, for comparison with the square-law characteristic of the triode, shown in Fig. 2 (a), are two varieties of the cube-law characteristic, typical of pentode valves and iron-cored coils.

Fourth harmonics are two octaves higher than the fundamental, and as regards harmony are therefore less conspicuous than third harmonics. The only serious effect would be if they were strong enough to make the music sound two octaves higher than it was supposed to be, but in practice this would hardly be so. Any distortion that produces fourth harmonic also produces much stronger second harmonic.

A similar principle holds with the odd harmonics; fifth is accompanied by much stronger third. But how do we expect the fifth to sound in relation to the fundamental? Relative to two octaves above the fundamental, its ratio is 5:4. And I think the musicians would still be with us if we declared that this is the next easiest on the ear, after octave and dominant. Sol-fa practitioners identify it as "me" above "doh." If all four notes we have now considered are played together—doh, me, soh, doh—the combination is still harmonious and pleasant. It is, in fact, the "common chord." But I suspect that a musician would consider it a bit thick, in more senses than one, if every single note of his composition were replaced by this four-note combination; which is virtually what would happen if all harmonics up to and including the fifth were added. However, although it would not be a faithful reproduction of the composer's intentions, the non-musical hi-fi expert, without being able to compare it with the original, might (I suggest) be unable to recognize it as "distortion" in his sense of the word.

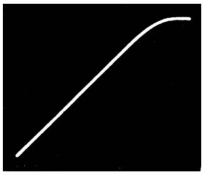


Fig. 6. When the characteristic has a sudden bend, like this, the higher harmonics are created at appreciable strength.

And so we could go on. Sixth harmonics are like thirds except for being an octave higher. But when we come to the seventh, the ratio to the next lower octave above the fundamental is 7:4. According to my untutored reckoning, this is B flat in relation to C. I don't know how it is rated by the musicians, but it sounds pretty discordant to me, even though my musical taste tends towards the modern. The eighth harmonic is three octaves above the fundamental, so may sound rather squeaky but certainly not discordant. The ninth, which after deducting the whole octaves is like sounding "doh" and "ray" together, is aggressively discordant. As we go higher up the series of *odd* harmonics the numerical ratio becomes more awkward and the musical sound more discordant. The even harmonics are not quite so, because the number can be simplified by dividing by 2, perhaps more than once, and that is musically equivalent to the interval of an octave, which harmonically hardly counts. Take the 12th harmonic; in relation to two octaves above the fundamental its ratio is 12:4, which simplifies to 3:2, and that, as we have seen, is a very easy harmony. But the 14th can only be simplified to 7:1, so it is the lowest discordant even harmonic.

What decides which harmonics are produced, and how much? As one can find out by making the same sort of comparison as Fig. 2 with Fig. 3, using different input/output (or "transfer") characteristics,

(*Continued on page* 195)

or, more elegantly, by mathematics,* it is the shape of the transfer characteristic that is responsible. The two most important are the square-law, with its smooth one-way bend shown in Fig. 2, which produces second harmonic, and the cube-law, with its S bend (but still smooth) shown in Fig. 5, which produces third harmonic. The sharper and more irregular the bends, the higher the harmonics created. The characteristics of valves worked under reasonable conditions are usually one or other of the first two (though less exaggerated) or a combination of both, and harmonics are therefore nearly all second or third or both. And we have seen that these are not in the least discordant. But if a valve runs into grid current at the signal peaks, or for any other reason has a characteristic with an abrupt corner, such as Fig. 6, the resulting harmonics are distributed well up the scale, including perhaps appreciable amounts of the discordant numbers. Incidentally, a practical way of seeing the shape of the transfer characteristic of an amplifier is to connect the input voltage across the X plates of an oscilloscope and the output voltage (phase-shifted if necessary to close the loop) across the Y plates.

It seems that unless the characteristic is so unsuitable that it brings in at least the seventh among the odd harmonics and the 14th in the even series, there should at any rate be no harshness, if harmonics were all that happened. However, there are intermodulation products to be reckoned with. And I am afraid that if we started to reckon with them at all seriously just now it would take up too much space. We shall have to put it off until next month.

* See "Relationships between Amplitudes of Harmonics and Intermodulation Frequencies," by M. V. Callendar and S. Matthews, in *Electronic Engineering*, June, 1951, p. 230, where the results are conveniently tabulated.

Output Transformer Design

For Amplifiers Employing Negative Feedback

By R. F. GIBSON*

IT is relatively easy to design a feedback amplifier with a flat response and good inherent stability to cover a range of 9 octaves. It becomes increasingly difficult, however, as the range is extended another one or two octaves, largely owing to instability troubles caused by the output transformer.

The basic requirements for a.f. transformers for use with negative feedback amplifiers, providing low-distortion power outputs, are well known but may be briefly recapitulated as follows:—

High primary inductance.
Low primary/secondary leakage inductance.
High-frequency resonance at a frequency where the loop gain of the feedback section of amplifier is less than unity.

Some additional considerations of practical importance are:—

Economical design.
Adequate electrical insulation.
Suitable choice of core material.
Moderate I^2R losses.

Consideration of these requirements will show that the design features must effect as good a compromise as possible between several conflicting requirements, e.g., high primary inductance means a large number of primary turns which necessitates a large I^2R loss or a large winding space. A large winding space requires a highly sectionalized winding to keep down leakage inductance. This precludes economical design and increases the difficulty of maintaining adequate electrical insulation.

One way of reducing primary turns is to use a high permeability core material, but this solution is often ruled out on the score of cost.

The ordinary grades of silicon iron have a relatively low distortion coefficient but suffer from the disadvantage of very low permeability at low flux densities. This has a serious disadvantage when considered in relation to feedback amplifiers. Briefly, the very low primary inductance at zero signal level necessitates the amplifier designer using otherwise unnecessarily long time constants in his l.f. couplings to keep away from the 180° phase shift associated with a 12-db slope which would result in low-frequency instability. No doubt many readers will have had painful experience of this trouble.

Instability

One major cause of h.f. instability is resonance "inside" the range of significant loop gain, resulting in a reversal of feedback polarity within the pass band of the amplifier. This is usually produced by the increased leakage inductance associated with a large number of turns in conjunction with high inter-winding capacitances.

The foregoing remarks may appear to give a somewhat gloomy picture of the performance of an output transformer in a high-quality feedback amplifier. Fortunately, it is possible, by careful and adequate design, to obtain a performance which, in fact, leaves little to be desired, and some of the basic requirements of such a design will now be discussed.

1. *Core material:* There appears to be no better material at present available than silicon steel. There are, however, several varieties of this material the relative merits of which will be discussed later.

2. *Winding space to core cross-section ratio:* without going into the mathematics of this problem it may be stated that economic considerations inevitably lead

* R. F. Gibson, Ltd.

to the choice of a small window to core ratio; this choice also helps considerably in easing the problem of obtaining a high resonant frequency and low leakage inductance.

3. *Efficiency:* Once a small window space has been decided upon it will be found that the weight of copper which can be fitted into it is small and therefore it becomes fairly safe to assume that the I^2R losses will not be unreasonable, providing the primary wire gauge is large enough to handle the r.m.s. value of standing d.c. plus audio-frequency current without overheating and the I^2R loss ratio between primary and secondary is reasonably near 1 : 1.

4. *Primary Inductance:* An empirical formula which has been found useful in determining the primary inductance is—

$$L = \frac{R_l \times V\beta}{2 \times 10^3} \quad \ldots\ldots\ldots\ldots\ldots\ldots \quad (1)$$

where L is in henries, R_l = anode-to-anode load (ohms) and $V\beta$ is the feedback voltage ratio. In the case of push-pull EL84's with 26 db feedback, this works out at $\frac{8000 \times 20}{2 \times 10^3}$

5. *Flux density:* Again a simplified equation—

$$N = \frac{10^3\sqrt{WR_l}}{KfA} \quad \ldots\ldots\ldots\ldots\ldots\ldots \quad (2)$$

where N = number of primary turns, W = V.A input to primary, R_l = anode-to-anode load in ohms, f = frequency of bottom distortion limit, A = cross sectional area of core (sq. in.) and

K = 1.6 for intermediate grade ⎫
1.7 for high grade ⎬ laminations
3.3 for "Unidi" material† ⎭
3.5 for "C" cores

This formula gives a practical answer for ratings up to 25W if the core area is in the region of $\frac{\sqrt{W} \times 30.}{(0.5 + K) \times f}$

Empirical data plus a consideration of general requirements will then enable a suitable core to be selected.

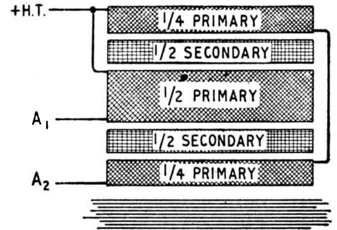
Going back to a choice of a suitable core material, we have available, intermediate grade silicon steel, high-grade silicon steel and oriented-grain silicon steel, the last mentioned being available in the form of either laminations or "C" cores. "C" cores are expensive and show only a small advantage over "Unidi" laminations both as regards the coefficient in equation (2) and the primary inductance to AN^2 ratio. Oriented grain material does however show a very marked advantage over the other grades of silicon steel and in the case of laminations is reasonably economical provided that it is obtained in the form of "no waste" E and I laminations.

It now remains to select a core size which can be made to satisfy the requirements of equations (1) and (2) and the clause concerning temperature rise. In the case of a 12-watt transformer using push-pull EL84's, the "no waste" size 4, having a 1in. wide core and a $1\frac{1}{2}$in. × $\frac{1}{2}$in. window, fits the requirements when built into a square stack.

The simplest winding arrangement which will provide a level response up to 30 kc/s is as shown in Fig. 1. and this provides a d.c. resistance balanced with respect to A_1 to h.t. and A_2 to h.t. The inter-

† Geo. L. Scott and Co., Ltd.

Fig. 1. Simple method of sectionalizing which gives adequate coupling



winding capacities are unbalanced but the overall coupling factor is good enough to take care of this. It should be noted that the winding layout shown is not suitable for a transformer having primary taps for the so-called "ultra-linear" circuit. One way of providing correct screen couplings is to transpose the primary and secondary windings.

Tests carried out on a transformer designed according to the foregoing data show that the expected results are well maintained in practice. The actual readings obtained were:—

Primary d.c. resistance 340 Ω; secondary 0.98 Ω; leakage inductance 24mH; initial inductance of primary, better than 130 H.

The measured performance is as follows:—

± 1 db from 25 c/s to 42 kc/s and the distortion limit on a sine wave trace is 28 c/s to 35 kc/s at 12 watts output from secondary, these figures being slightly over 1 octave better than can be obtained on the same size of core with intermediate grade laminations.

BOOKS RECEIVED

Television, by V. K. Zworykin, E.E., Ph.D., and G. A. Morton, Ph.D. Revised second edition covering fundamental physical principles, complete systems for monochrome and colour and details of camera and display tubes. Pp. 1037+xv; Figs. 698. Price 40s. Chapman and Hall, 37, Essex Street, London, W.C.2.

Radio and Television Engineers' Reference Book. Edited by E. Molloy and W. E. Pannett, A.M.I.E.E. Compendium of descriptive information, data and servicing hints in all branches of radio communication, contributed by 36 specialists. Includes chapters on sound reproduction and distribution, disc and magnetic tape recording. Pp. 1542+xx; Figs. 1117. Price 70s. George Newnes, Ltd., Tower House, Southampton Street, London, W.C.2.

Ibbetson's Electric Wiring. Edited by C. R. Urwin, A.C.G.I., A.M.I.E.E.; W. F. Parker, M.I.E.E., and F. G. Thompson, M.Sc. (Eng.), A.M.I.E.E. Ninth edition of this textbook of theory and practice for practical wiremen and students. Pp. 296+viii; Figs. 119. Price 11s 6d E. and F. N. Spon, Ltd., 15, Bedford Street, London, W.C.2.

Fundamentals of Transistors, by Leonard M. Krugman. Summary of design procedure and formulæ for the principal transistor circuit configurations, with an introductory chapter on basic semi-conductor physics. Pp.140; Figs. 110. Price 21s. Chapman and Hall, 37 Essex Street, London, W.C.2.

Radar Pocket Book by R. S. H. Boulding, B.Sc., M.I.E.E. Basic information on radar systems, components and circuits for the use of operators, installation and maintenance engineers. Pp. 176+vii; Figs. 156. Price 15s. George Newnes, Ltd., Southampton Street, London, W.C.2